Performance Studies of VoIP over Ethernet LANs

Di Wu

A dissertation submitted to

Auckland University of Technology

in partial fulfillment of the requirements for the degree of

Master of Computer and Information Sciences

2008

School of Computing and Mathematical Sciences

Primary supervisor: Nurul I. Sarkar

Table of Contents

Attestation of Authorship

I hereby declare that this submission is my own work and that, to the best of my knowledge and belief, it contains no material previously published or written by another person (except where explicitly defined in the acknowledgements), nor material which to a substantial extent has been submitted for the award of any other degree or diploma of a university or other institution of higher learning

Acknowledgements

I would like to express my gratitude to my supervisor Mr. Nurul Sarkar, for his support and encouragement throughout. Special thanks to Ms. Krassie Petrova for her valuable help. Appreciation also goes to the administrator, Ms. Ann Wu, for her support.

I would like to express my deepest gratitude to my wife Jing, for her constant support, understanding, and love.

Abstract

Voice over Internet Protocol (VoIP) is a rapidly growing technology that enables transport of voice over data networks such as Ethernet local area networks (LANs). This growth is due to the integration of voice and data traffic (telecommunication convergence) over the existing networking infrastructure, low cost, and improved network management offered by the technology.

This research investigates the performance of VoIP traffic characteristics over Ethernet LANs. In the investigation, the impact of increasing the number of VoIP clients, voice codec schemes, and traffic distribution on system performance is considered. Through various simulation experiments under realistic networking scenarios, such as small offices home offices (SOHO) and campus networks, this study provides an insight into the VoIP performance over Ethernet LANs. The simulation results indicate that all these factors can significantly affect VoIP performance over Ethernet LANs. Under both SOHO and campus network scenarios, increasing the number of VoIP clients, voice packet lengths and different traffic arrival distributions have significant impact on system performance.

List of Figures

List of Tables

Table **Page**

List of Abbreviations

ACELP	**Algebraic Code Excited Linear Prediction**
ATA	**Analog Terminal Adapter**
CSMA/CD	**Carrier Sense Multiple Access with Collision Detection**
CODEC	**Compression/Decompression**
GUI	**Graphical User Interface**
GK	**Gate Keeper**
GW	**Gateway**
IP	**Internet Protocol**
ITU	**International Telecommunication Union**
LAN	**Local Area Network**
MAC	**Medium Access Control**
MCU	**Multipoint Control Unit**
PBX	**Private Branch Exchange**
PCM	**Pulse Code Modulation**
POTS	**Plain Old Telephone System**
PSTN	**Public Switched Telephone Network**
QoS	**Quality of Service**
RP	**Real-time Protocol**
RTCP	**Real-Time Control Protocol**
SOHO	**Small Office and Home Office**

SIP	**Session Initiation Protocol**
TCP	**Transmission Control Protocol**
UPS	**Uninterruptible Power System**
VoIP	**Voice over Internet Protocol**
WLAN	**Wireless Local Area Network**
WEP	**Wired Equivalent Privacy**
WPA	**Wi-Fi Protected Access**
WPA2	**Wi-Fi Protected Access 2**

Chapter 1

Introduction

In recent years, there is a growing trend in real-time voice communication using Internet protocol (IP). Voice over Internet Protocol (VoIP) is a technology that allows users to make telephone calls over an IP data network (Internet or Intranet) instead of traditional Public Switched Telephone Network (PSTN). Therefore, VoIP provides a solution that merges both data and voice which gains benefits include cost savings, high quality and value added services. Today, VoIP is becoming one of the most widely used technologies today, more and more people and organisations are using VoIP systems worldwide. There are various VoIP communication software products are already available on the internet: Skype, Google Talk, and Windows live messenger. All of them can provide good quality, cheap, and even free phone calls [1], [2], [3].

VoIP is not only popular through the internet; it is also a rapidly growing technology through data networks such as Ethernet LANs. Ethernet is considered a good platform for VoIP [4] as Ethernet based LANs is very common in enterprises and other organizations for data networking [5]. Therefore, there is a tremendous growth of VoIP. This growth is due to the integration of voice and data over the existing networking infrastructure, low cost, and improved network management offered by the technology. In addition, wireless Ethernet networks (IEEE 802.11) allow mobile users to connect to the network from the location where network cables may not available or may not be the best choice, such as old buildings, Hospitals, and conference rooms. Therefore, WLANs are another important segments for VoIP deployments. The performance of VoIP over WLANs is also investigated in this dissertation.

1.1 Objectives of this study

Despite the potential benefits of VoIP over Ethernet LANs, one of the significant challenges faced by designer of VoIP is to provide a quality of service (QoS) to all users on the network, especially under medium-to-high traffic loads. However, IP networks were originally designed for data networking, not for voice, and additionally, an IP network is shared and ulitilised by many different devices and services. Unlike the classical applications such as file transfer or mail, VoIP is a real time service, the access competition can result in delays or packets lost which is detrimental to real-time applications. However, VoIP is an emerging technology that has many issues, how to deploy VoIP services over existing networks is still a challenge for managers, network architects, designers, planners, and engineers.

Therefore, a good understanding of VoIP traffic characteristics and network performance analysis is required to assist efficient deployment of such technologies over Ethernet LANs.

The aim of this research was to investigate the effect of the following factors on system performance:

- increasing the number of VoIP clients
- traffic arrival distributions
- voice codec schemes

1.2 Dissertation Structure

Chapter 2 introduces the background material for the dissertation. It provides an overview of VoIP technology including VoIP calls, network topologies, protocols, compression algorithms, and QoS. Chapter 3 reviews relevant literature on VoIP. Chapter 4 outlines the research methodology adopted in this dissertation. In Chapter 5, network modelling and scenarios are described. Chapter 6 presents experimental results obtained from simulation runs, and Chapter 7 concludes the dissertation.

Chapter 2

Background

In Chapter 1, the main objective of this research was outlined. In this chapter, background material relevant to the dissertation is presented to help understand the subsequent Chapters of this dissertation.

2.1 Overview

VoIP stands for voice over internet protocol. Unlike the traditional circuit-committed protocols of the public switched telephone network (PSTN), in VoIP voice signal is compressed and converted to digital voice packets, VoIP then uses the Internet Protocol (IP) for managing voice packets over IP network. Therefore, VoIP can be deployed on any IP enabled data network, such as the Internet, Ethernet, fabric or wireless network.

2.2 Types of VoIP

There are several different types of VoIP service depending on the infrastructure used for the communication: computer-to-computer based VoIP (VoIP device to another VoIP device); computer-to-Phone based VoIP (VoIP device to a PSTN device); and Phone-to-Phone based VoIP (PSTN device to another PSTN device) [40]. Each type of them has different set of requirements. This section describes the three broad categories of VoIP service.

Computer to Computer Internet telephony services via computers are totally free VoIP services. This type of VoIP services via specialised software applications (softphone software) such as Skype, AOL Instant Messenger, and MSN Messenger etc. These services require users to download their software and get them installed on PC, Caller and receiver need to use same VoIP software application (For instance, Skype to Skype, MSN to MSN etc), caller and receiver are communicated based on peer-to-peer approach through the Internet.

The requirements for computer to computer Internet telephony includes:

- softphone software,
- A sound card
- Internet access

Computer to Phone Because the Internet and conventional circuit switched telephone systems use different systems. Thus, softphone software need to routes the call through internet protocol and hands it off to a conventional telephone network. Skype, MSN, and GoogleTalk also provide services to users make phone calls from computers to typical landline phones.

Equipment requirements:

- VoIP service subscription
- Internet access
- A modem
- An Analog Terminal Adapter (ATA) that converts the analog call signal to digital signal (and vice versa).

Phone to Computer Users can make phone calls from traditional landline phones to computers with this service. A phone number will be assigned to a computer's IP address. A user can dial this number just like making normal phone calls. Therefore, wherever you are, you can receive phone calls on your computer from landline phones via the number assigned. Skype now allows users to purchase phone to computer VoIP services [1].

Phone to Phone This is the ultimate step of VoIP services. Currently, many telephone companies already use this service to handle long distance calls. In the future, telephone companies are able to use the internet to handle all the telephone calls. Therefore, VoIP services completely do not need the traditional PSTN for both call origination and termination.

2.3 VoIP System

Figure 1.1 shows a typical VoIP network topology that includes following equipments:

2.3.1. Gatekeeper: A gatekeeper or callmanager node is optional for a VoIP network. In an H.323 IP telephony environment, a gatekeeper works as a routing manager and central manager that manage all the end nodes in a zone. A gatekeeper is useful for handling VoIP call connections includes managing terminals, gateways and MCU's (multipoint control units). A VoIP gatekeeper also provides address translation, bandwidth control, access control [3]. Therefore, A VoIP gatekeeper can improve security and Quality of Service (QoS)

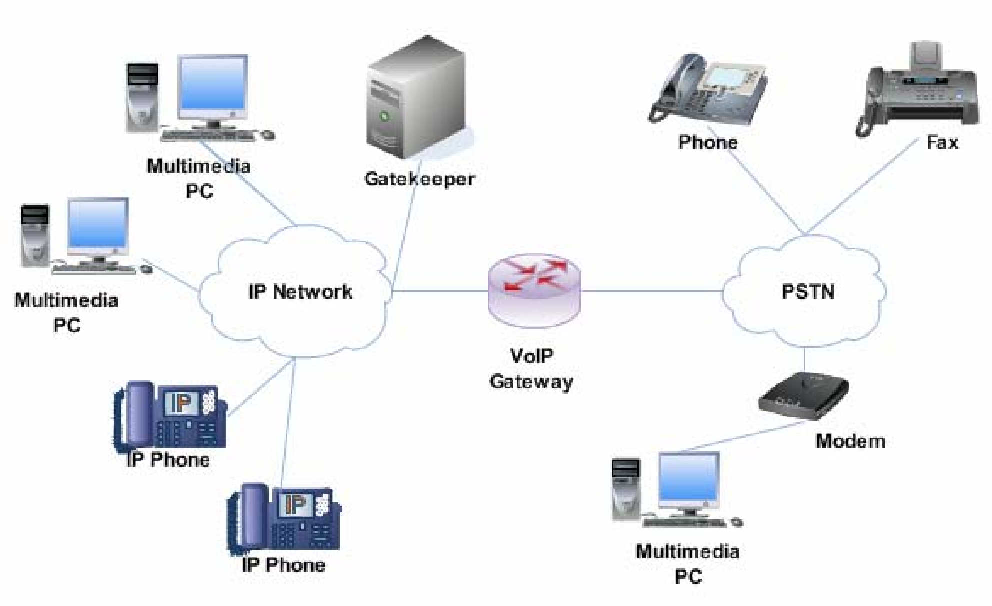


Figure 1.1: A typical VoIP Network Topology

2.3.2. VoIP Gateway: A VoIP gateway is also required to handle external calls. A VoIP gateway functions as a converter that converting VoIP calls to/from the traditional PSTN lines, it also provides connection between a traditional PBX

(Private Branch Exchange) / Phone system and an IP network.

2.3.3. VoIP Clients: Other required VoIP hardware includes a VoIP client terminal, a VoIP device could be an IP Phone, or a multimedia PC or a VoIP-enabled workstation runs VoIP software.

2.4 VoIP Protocols

2.4.1 H.323

There are two standard protocols used in VoIP network: Session Initiation Protocol (SIP) and H.323, (Skype [1] and some others use proprietary signaling and messaging protocols). H.323 [6] is ITU (International Telecommunication Union) standard based on Real-time Protocol (RP) and Real-Time Control Protocol (RTCP); H.323 is a set of protocols for sending voice, video and data over IP network to provide real-time multimedia communications. H.323 is reliable and easy to maintain technology and also is the recommendation standard by ITU for multimedia communications over LANs [8], [9]. Figure 1.2 shows the H.323 architecture.

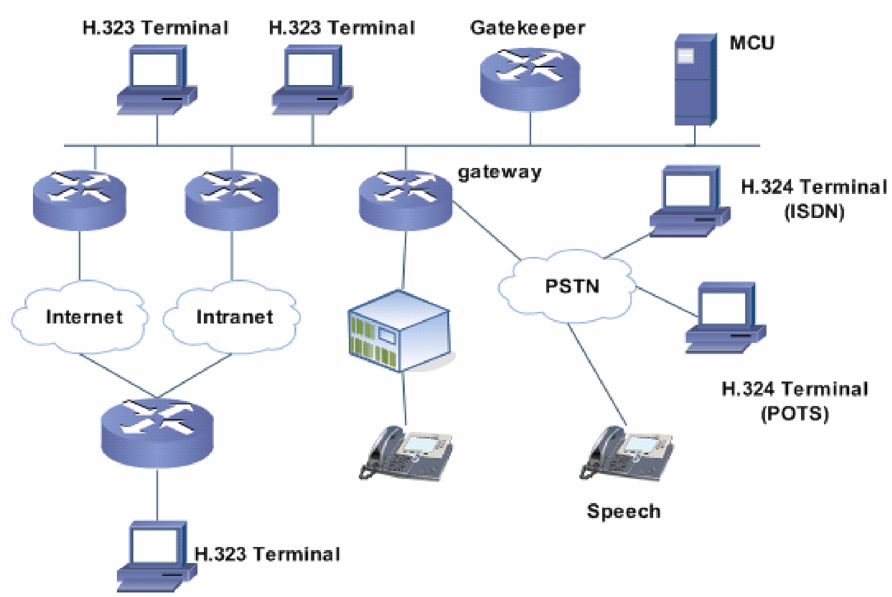


Figure1.2: H.323 Architecture

There are four basic entities in a default H.323 network [9], [10]: terminal, gateways (GW), gatekeepers (GK) and multipoint control units (MCU): H.323 terminal also called H.323 client is the end-user device. It could be IP telephone or a multimedia PC with another H.323 client. That provides real-time two-way media communication. A Gateway (GW) is an optional component that provides inter-network translation between terminals. A Gatekeeper (GK) is an optional component provides address translations and access control services. A Multipoint Control Unit (MCU) functions as a bridge or switch that enables three or more terminals and gateways in a multipoint conference.

2.4.2 SIP

H.323 has some limitations such as lack of flexibility, thus another protocol SIP is getting popular in VoIP [41]. SIP (stands for Session Initiation Protocol) was developed by the Internet Engineering Task Force (IETF) and published as RFC 3261 [12]. SIP is a signaling control protocol which is similar to http, it’s designed to initial and terminate VoIP sessions with one or more participants [11]. It is less weight and more flexible than H.323 that also can be used for multimedia sessions such as audio, video and data. Figure 1.3 shows the architecture of SIP protocol.

SIP has two components: User Agents and SIP servers. User agents are peers in a SIP. User agents could be either an agent client or an agent server. A user agent client initiates by sending a SIP request. A user agent server can accept, terminate or redirect the request as responses to this SIP request. There are three types of SIP servers include SIP proxy servers, SIP registrar servers, and SIP redirect servers. A SIP server functions as a server that handles these requests, e.g. requests transferring, security, authentication, and call routing.

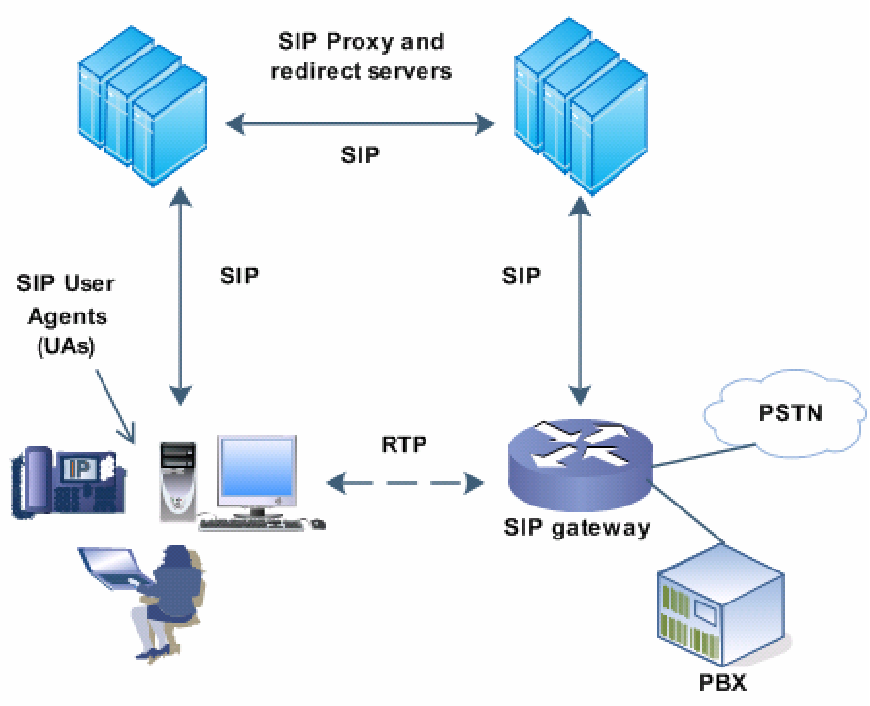


Figure 1.3: SIP Architecture

SIP is not only popular in VoIP applications but also widely used in applications include instant messaging and some other commercial applications, e.g. Microsoft MSN Messenger, Apple iChat.

2.5 VoIP Compression Algorithms

Codecs generally provide a compression capability to save network bandwidth. Currently, there are many different audio codecs available for voice applications. The simplest and most widely used codecs are G.711, G.723 and G.729 [7]. The simplest encoder scheme is G.711 (64 kb/s). G.711 is the sample based which uses Pulse Code Modulation (PCM). The acceptable packet loss factor of G.711 is up to 0.928%.

G.723 and G.729 are frame based encoder scheme with higher compression and smaller data rates (8 kb/s for G.729, 5.3 and 6.4 kb/s for G.723.1). The G.723

encoder scheme was developed for use in multimedia, and G.729 is a Conjugate-Structure Algebraic Code Excited Linear Prediction (CS-ACELP) speech compression algorithm approved by ITU-T. However, G.723 and G.729 also generate higher complexity and encoding delay with lower quality. Therefore, G.711 is considered as the default choice for this study as the worst case for bandwidth and the best in quality. In this dissertation, there is one independent simulation scenario tests G.711, G.723, and G.729 encoder schemes to investigate the performance differences. The properties of major voice codecs as shown in Table 1.1

Table 1.1 Voice codec and properties

Codec	Bit Rate	Payload	Packets Per Second (pps)	Quality	Ethernet Bandwidth	Sample period
G.711	64kbps	160Byte	50pps	Excellent	95.2 kbps	20 ms
G.729	8kbps	20Byte	50pps	Good	39.2 kbps	10 ms
G.723.1	6.3kbps	24Byte	34pps	Good	27.2 kbps	30 ms
G.723.1	5.3kbps	20Byte	34pps	Good	26.1 kbps	30 ms

(Source: http://www.cisco.com)

2.6 VoIP QoS

QoS is a very important aspect for IP-based multimedia services. Many IP services without QoS guarantees from network providers are also very successful because transport quality is sufficient to meet customer demands. However, QoS for these services cannot be guaranteed when services grow and customer demands increase. For instance, IP-based voice and video services within organisations usually do not have explicit QoS support because usually the LANs provide enough bandwidth for real-time voice and video services. However, it is very hard to assure QoS for real time multimedia services across worldwide networks. There are many factors affect voice quality, which includes the choice of codec, delay, packet loss, jitter.

Delay: High QoS should be assured by control delay so that one-way communication delay should be less than 150 ms. (ITU states that one-way, end-to-end telephony applications should have less than150 ms delay in echo-free environments to ensure user satisfaction [31]). Delay mainly comes from three components [13]: (1) delay caused by voice codec algorithms (2) delay caused by queuing algorithms of communications equipment (3) variable delay caused by various factors (i.e. network conditions, VoIP equipments, weathers etc). It is very important to minimise the voice traffic delay. Thus, a codec algorithm and queuing algorithm needs to be carefully considered. Although traditionally think the end-to-end delay of 150 ms was considered as acceptable for most applications. However, in reference [35], the authors state that a delay of up to 200ms is considered as acceptable. Moreover, a one way end-to-end delay between 150ms to 400ms is considered as acceptable for planning purposes. In this study, 200ms will be considered as the maximum acceptable one way end-to-end delay, high end-to-end delay can cause bad voice quality perceived by the end user.

Jitter: Delay variation also called Jitter. Jitter is the difference value between the delays of two queuing packets. Root causes of jitter including network conditions and packet loss; it is very difficult to deliver voice traffic at a constant rate. In order to minimise jitter a jitter buffer (also known as playout buffers) is needed. A jitter buffer is used to trade off delay and the probability of packet interruption playout. Jitter value is considered acceptable between 0ms and 50 ms and above this is considered as unacceptable [11].

Packet loss: Packet loss is also an important factor VoIP QoS. Packet loss occurs when more transmitted packets on the network then causes dropped packets. VoIP packets are very time sensitive. Therefore, packet loss can significantly affect VoIP quality. For instance, a dropped conversation, delay between communicating clients, or noise on a VoIP call. Acceptable packet loss rate is 1 % and it will be considered as unacceptable if above this ratio [26]. However, an early study shows that the tolerable packet loss rates are within 1-3% and the voice quality becomes intolerable when voice packet loss rate is more than 3% [29].

Therefore, all these factors need to be properly controlled by QoS mechanisms. When these factors are properly controlled, VoIP voice quality can be even better through lower speed connections. In the meantime, data applications in the network can be also prioritized and assured with limited and shared network resources. The quality VoIP is the key factor of VoIP service to achieve success.

2.7 Reasons for VoIP Deployment

There are two major reasons to use VoIP: lower cost than traditional landline telephone and diverse value-added services. Zeadally et al., [14] introduce how these factors influencing VoIP adoption. Each of these will be described in this section

Cost Saving: This can be achieved by reusing the devices and wiring for the existing data network as most of the organisations already have their own networks. However, the most attractive reason to adopt VoIP maybe is dramatically reduced phone call cost. Soft phones such as Skype [5] enable PC-to-PC users can bypass traditional long-distance toll calls charge as voice traffic over the Internet, they only need to pay flat monthly Internet-access fee. Soft phones also allow a PC as a VoIP phone to call a mobile phone or a home line phone at a lower rate.

Advanced multimedia applications. Cost effective is only one of the good reasons to use VoIP. VoIP also enables multimedia and multi-service applications that increase productivity and create a more flexible work environment, e.g. real time voice-enabled conferencing systems that may include white boarding, file transferring, etc. which combine both voice and data features.

2.8 Challenges of VoIP

Though VoIP is becoming more and more popular, there are still some challenging problems with VoIP:

Bandwidth: Network availability is an important concern in network. A network can be broken down into many nodes, links, and generate a large amount of traffic flows, therefore, the availability of each node and link where we only concentrate the bandwidth of the VOIP system. An in a data network, bandwidth congestion can cause QoS problems, when network congestion occurs, packets need be queued which cause latency and jitter. Thus, bandwidth must be properly reserved and allocated to ensure VOIP quality. Because data and voice share the same network bandwidth in a VOIP system, the necessary bandwidth reservation and allocation become more difficult. In a LAN environment, switches usually running at 100 Mbps (or 1000 Mbps), upgrading routers and switches can be the effective ways to address the bandwidth bottlenecks within the LAN.

Power Failure and Backup Systems: Traditional telephones operate on 48 volts and supplied by the telephone line itself without external power supply. Thus, traditional telephones can still continue to work even when a power failure occurs. However, backup power systems required with VOIP so that they can continue to operate during a power failure. An organization usually has a uninterruptible power system (UPS) for its network to overcome power failure, desktop computers and other network devices may need much of the power to continue their functions during power outages, a backup power assessment is needed to ensure that sufficient backup power is available for the VOIP system. This may increase the costs of backup power systems; costs may include electrical power charge to maintain UPS battery, maintenance costs, UPS battery etc.

Security: As VoIP becomes more and more popular, the security issues relate to VoIP network systems are also increasingly arising [37]. W. Chou [16] analysis the different aspects of VoIP security and gives some suggested strategies to these issues. In reference [17], the authors also outline the challenges of securing VoIP, and provide guidelines for adopting VoIP technology.

Softphone: Softphones are installed on computers thus should not be used where security is a concern. In today's world, worms, viruses, Trojan houses, spy wares and etc are everywhere on the internet and very difficult to defend. A computer could be attacked even if a user does not open the email attachment, or a user does nothing but only visit a compromised web site. Thus use of softphones could bring high risks for vulnerabilities.

Emergency calls: Each traditional telephone connection is tied to a physical location, thus emergency service providers can easily track caller's location to the emergency dispatch office. But unlike traditional telephone lines, VoIP technology allows a particular number could be from anywhere; this made emergency services more complicated, because emergency call centers cannot know caller's location or may not possible to dispatch emergency services to that location. Although the VoIP providers provide some solutions for emergency calls, there is still lack of industry standards in a VOIP environment.

Physical security: Physical security for VoIP networks is also an important issue. An attacker could do traffic analysis once physically access to VoIP servers and gateways, for example, determine which parties are communicating. Therefore, physical security policies and controls are needed to restrict access to VOIP network components. Otherwise, risks such as insertion of sniffer software by attackers could cause data and all voice communications being intercepted.

Wireless Security: Wireless nodes integrated in VoIP network is getting more and more common and popular [36]. Wired Equivalent Privacy (WEP) security algorithm for 802.11 wireless networks is very weak because WEP can be cracked with publicly available software. Due to the weakness of the WEP, more recent WiFi Protected Access (WPA and WPA 2) which administered by the Wi-Fi Alliance provides significant security improvements, the WPA protocol can be integrated with wireless technology in VoIP.

Chapter 3
Related Work

In Chapter 2, the background material relevant to this dissertation was presented. This chapter reviews relevant literature on VoIP. The studies relate to VoIP network performance and workload studies such as VoIP protocol analysis and traffic analysis is outlined in Section 3.1. Section 3.2 reviews literature on VoIP QoS measurements.

3.1 Network Performance Studies

In reference [30], the authors investigated nearly two thousand users and presented study from the largest and most comprehensive trace of network activity in a large, production wireless LAN. This study can help understand usage patterns in wireless local-area networks which are critical for those who develop, deploy, and manage WLAN technology, as well as those who develop systems and application software for wireless networks.

3.2 VoIP Quality Studies

Voice over IP (VoIP) has become one of the most important technologies today. With the benefits such as significant reduction of communication cost, more and more organisations are panning to adopt VoIP applications, but the quality of the performance in IP network is still not guaranteed. In Takahashi et al. [28], the authors describe how objective and subjective factors determine the perceived quality of a VoIP system. The authors also introduce a modified model based on the E-model, the authors experimental results show that E-model can be enhanced so that it better estimates users' perceptions of VoIP service.

Packet loss, delay and jitter are the most important measurements parameters for voice traffic in the network environment. In Zheng et al. [31], the authors present a study of the individual effects of various traffic parameters on the jitter behavior of packet voice stream multiplexing background traffic with different burst characteristics in the IP router. This study focuses on the voice over IP traffic going through an IP router with the bursty background traffic over network.

In Salah & Alkhoraidly [15], the authors present a VoIP deployment study base on simulation models. The authors discuss the issues relate to VoIP the deployment, such as characteristics of VoIP traffic and QoS requirements. VoIP performance also accessed over internet backbones. In Markopoulou et al. [27], the authors assessed delay and loss measurements over wide-area backbone networks. The authors find that there is significant number of Internet backbone paths lead to poor performance.

However, these VoIP network performance studies have been conducted either based on the Internet or wireless network field, none of these investigate the impact of data traffic in VoIP performance over Ethernet LAN. Standard Ethernet network has been developed use Carrier Sense Multiple Access with Collision Detection (CSMA/CD) multiple access algorithm. The main draw back of this protocol is that a broadcast channel in an Ethernet LAN interconnects all nodes, thus when only one node transmits a frame, all the nodes will receive this frame, and all nodes must wait before continuing transmission. Due to this drawback, micro-segmentation is getting more and more popular since micro-segmentation can isolate collision domain to overcome this drawback.

Table 3.1 shows the leading researchers and their contributions in VoIP performance study.This helps finding a research gap and direction for further contribution in this dissertation.

Table 3.1: Leading researchers and their contributions in VoIP performance study.

Researcher	Contribution	Year	Description/key concept
[30] D. Kotz & K. Essien	This study can help understand usage patterns in WLAN	2005	This study investigated nearly two thousand users from Large WLAN
[28] A. Takahashi et al.	This study help understand factors determine quality of a VoIP system	2004	This study introduces how objective and subjective factors determine the perceived quality of a VoIP system
[31] L. Zheng et al.	This study help understand measure important QoS factor delay and jitter	2001	This paper studies the performance behavior of delay and delay jitter
[15] K. Salah & A. Alkhoraidly	This study help understand how to deploy a VoIP system over OPNET environment	2006	This study presents a detailed VoIP deployment study base on simulation models and discuss some issues relate to the deployment.
[27] A. P. Markopoulou et al.	The authors present a study for assessing delay and loss over wide-area backbone networks	2003	This study assess the ability of Internet backbones to support voice communication

Chapter 4
Research Methodology

Simulation methodology was adopted in this dissertation for the performance modelling and analysis of VoIP over Ethernet LANs. Simulation has become a popular approach for network studies and performance modeling [18], [19], [20], [25], [39], and [40]. Some simulation-based studies for VoIP network system recently, these studies deal with performance and perceptual quality of VoIP network system [14], [15], [21], and [23]. Another important reason of using simulation is that it can be easily control the scale of the network.

This dissertation is going to investigate the performance of VoIP using OPNET based on planned and designed network scenarios. In [22], the authors present a survey study that investigated VoIP performance over wireless networks and their study shows VoIP performance of wireless networks were much worse than wireline networks. They identified the maximum number of simultaneous voice connections that could be supported for a reliable wireless voice communication, and they suggest MAC (Medium access control) protocol, queue management schemes, voice codec choice and playout buffer algorithms as effective way to improve the VoIP performance over WLANs.

Salah & Alkhoraidly [15] present detailed description of VoIP deployment using OPNET simulator. Their investigation determines the maximum number of VoIP calls that can be supported by an existing network. The paper also discuss may design and engineering issues relate to VoIP deployment includes QoS requirements, VoIP flow and call distribution etc. However, their study only based on a wireline networks.

Simulation is widely used research methodology for network performance evaluation. This is because sometimes a networks may contains a large number of network nodes and services, it will be too much time consuming and costly to establish physical networks. Thus, network planners and engineers are often use simulation before deploying real networks.

Zubairi & Zuber [19] present a simulation study that developed a model of a university campus network with OPNET and obtained Ethernet delay, traffic statistics and other interesting data. They also ran interactive voice across the network to test if the developed model for university network could handle the demanding voice applications under different traffic load conditions. Their simulation results show very good performance under typical load conditions but the delays and delay variations increase under loaded.

Capelle, et al. [20] also designed a campus network in OPNET and tested the VoIP traffic on a shared Ethernet. They investigated the network performance in case the university network offers VoIP services for each student room. Their reported results include voice end-to-end delay, delay variation for each call, and Ethernet performance parameters.

However, all these studies only considered wired network. That is all the previous studies in literatures based on either wired or wireless network. Therefore, it is necessary to do investigation on both wired and wireless network from different aspects. This dissertation investigates performance VoIP over Ethernet LANs include both wired and wireless network components. The findings of this dissertation can help organizations make decisions for adopting of VoIP system and expansion plans for VoIP services.

However, the similar methodology can be used in this study. These studies can help to deploy a VoIP network, such as “how to generate VoIP calls” and understand ” how to change of call duration distribution”. These studies also help to know what results are useful and significant.

Chapter 5

Network Modelling

In Chapter 4, a review of literature on VoIP was presented. This chapter will introduce simulation environments; the configurations of specific devices and related technologies are required to support VoIP will also be presented. This chapter describes VoIP network simulation scenarios. Section 5.1 introduces the strengths and weaknesses of OPNET simulation tool. Section 5.2 describes various simulation scenarios considered. Section 5.3 outlines VoIP traffic configuration for each scenario.

5.1 Strengths and weaknesses of OPNET

The simulation tool adopted in this dissertation is OPNET educational version 14.0. This is the fully functional version for academic institutions. OPNET is an object-orientated simulation tool for planning, modelling and performance analysis of simulation of network communication, network devices and protocols. OPNET Modeler has a number of models for network elements, and it has many different real-life network configuration capabilities. These makes real-life network environment simulations in OPNET very close to reality and provide full phases of a study.

OPNET also includes features such as comprehensive library of network protocols and models, user friendly GUI (Graphical User Interface), data collection and analysis (graphical results and statistics). OPNET network modeling usually through three modeling hierarchical steps (Network modeling, node modeling and process modeling). First, a network topology needs to be defined include scale and size of the network (e.g., enterprise, campus, office and x span y span in degrees, meters, kilometers), the technologies need to be used (e.g., Ethernet, wireless), and nodes and links (e.g., 100Base, 1000Base). The node modeling deals with interrelation of processes, protocols and subsystems in and process mode describes the behaviour define the statistical features in a simulation model.

However, the current version of OPNET can only support SIP (Session Initiation Protocol) protocol, thus VoIP equipments such as VoIP gateway and gatekeeper product models are not included in OPNET, and this means the performance of VoIP gateway and gatekeeper are not measurable. Besides VoIP gateway and gatekeeper, OPNET can simulate voice traffic for both wired and wireless nodes. The statistical and graphical results for analysing the voice traffic transmission include the jitter, end-to-end delay, delay variation, and the amount of sent/received packets etc.

5.2 Simulation Environment and Scenarios

Testbed

OPNET 14.0.A PL3 (Build 6313 32-bit)
The hardware platform: Cyclone computer in Auckland University of Technology
Computer name: WT405-60853
Operating System: Windows XP Service Pack 2
Intel Core 2 CPU 6420 @ 2.13GHz
1.99GB of RAM

Modelling Assumptions:

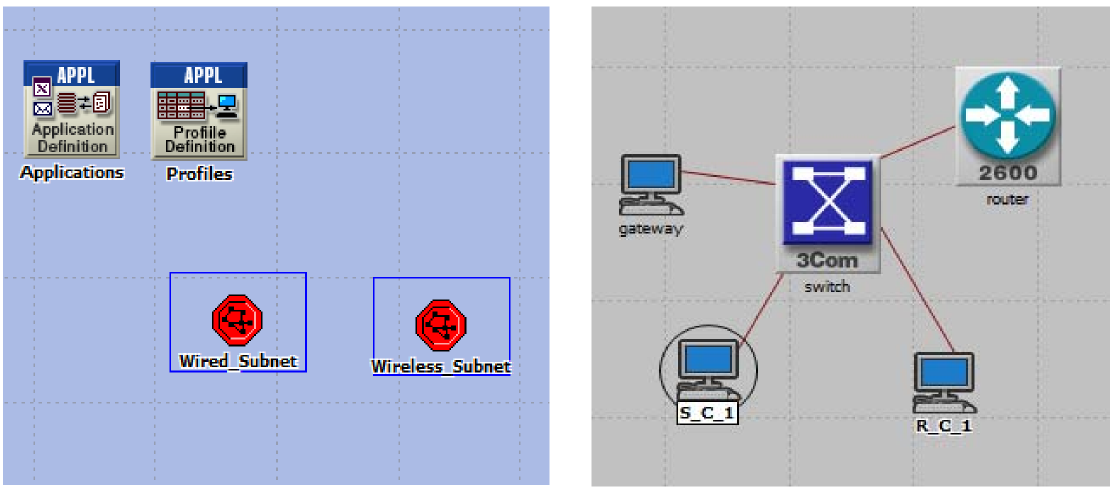
- The local area networks operate at 100Mb/s throughout the simulations.
- There is no other network traffic besides VoIP traffic in this study. Each simulation experiment considers 8 minutes of simulation time.
- This study also assumes that there are only peer-to-peer voice calls throughout the simulation, which means there is no voice conferencing.

Various scenarios were simulated in this dissertation. The simulated scenarios were going to investigate impact of increasing number of VoIP clients; impact of voice encoder schemes; and impact of traffic arrival distributions:

5.2.1 Scenario 1: Impact of Increasing Number of VoIP Clients

This scenario investigates the impact of increasing number of VoIP clients. Because when there is only one node in an Ethernet LAN, the transmission rate of the Ethernet LAN could close to the maximum rate (100 Mbps or 1000Mbps), but the effective transmission rate can be much less when the number of nodes increases. In this scenario, the simulation initially measuring a small office VoIP network that contains up to 20 workstations, 20 workstations is a reasonable number for a small office network.

The essential components are added in this scenario includes one switch, one router, and one VoIP gateway. Thus for VoIP traffic workload, the number of VoIP clients is progressively increased from 2 to 20 in the designed network. The VoIP gateway is a PC workstation. The number of VoIP clients then will be increased to 400 to see the impact it has on VoIP performance. Figure 5.1 shows the OPNET representation of network topology for Scenario 1.



(a) **(b)**

Figure 5.1 OPNET Representation of VoIP Network model

5.2.2 Scenario 2: Impact of Wireless Nodes

Many organisations are using WLANs or already have WLAN components within their networks. Therefore, it is also important to investigate the performance of VoIP over WLAN. In this scenario, a WLAN is assumed as a component of the company's local area network, and the company wants to install VoIP on the entire LAN. Then all the calls inside the company will use VoIP services. It would be interesting to investigate VoIP performance limitations over WLAN.

In this case, the local area network (LAN) consists of 20 nodes as wired component, 20 nodes is reasonable for a small company network. The next step is to add some wireless nodes. The number of wireless nodes is increased from 2 nodes to 6 nodes to see the performance differences. The wireless workstations in OPNET with built-in VoIP ability, this scenario investigates both IEEE802.11b（11Mb/s） and IEEE802.11g protocol (54Mb/s) and will try to find out which is the best wireless protocol for this VoIP network model. Wireless LAN structure is assumed as Infrastructure which means a wireless access point is needed (In this case access point for mobile stations is a wlan_ethernet_router). The wireless nodes are connected with wlan_ethernet_router in wireless subnet (see Figure 5.2), and this wlan_ethernet_router is connected to the wired network by a 100BaseT link. The wireless End-to-End delay, delay variation, packet loss and throughput are measured using OPNET Modeller. The network topology for wireless part is shown in Figure 5.2.

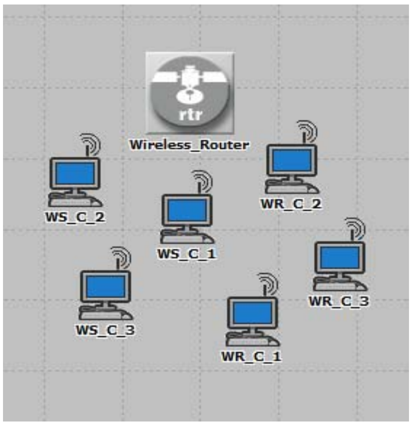


Figure 5.2 OPNET Representation of a fully connected wireless LAN

5.2.3 Scenario 3: Impact of Voice Encoder Schemes

Voice quality is crucial for designing a VoIP network system. Today, users are demanding high quality voice of VoIP network system, especial under limited condition such as low bandwidth, high packet loss rate and delay. Compression and decompression of voice signals have negative impacts of voice quality. Therefore, it is very important to select the suitable encoder for a VoIP network. In order to measure the VoIP performance and voice quality under different voice encoders, in this scenario, simulations are performed based on different codec schemes through G.711, G.723 (5.3K) and G.729.

5.2.4 Scenario 4: Impact of Traffic Arrival Distributions

This scenario measures different traffic distributions to investigate the impact of traffic arrival distributions to VoIP performance over Ethernet LAN. In previous simulations, VoIP calls are constantly generated in every 5 seconds, and duration for each call is 300 seconds. However, it is unlikely that VoIP calls will be generated in constant fixed rate or have fixed length of VoIP calls in a real network (i.e. a VoIP network may has peak hours/ off peak hours), thus this scenario assesses the VoIP performance based on some other voice arrival traffic distributions including Poisson distribution and exponential distribution.

1. **Constant arrival distribution-** VoIP calls will be generated in a constant rate and last in a constant certain time (each call duration is 300 seconds and will be added in every 5 seconds).

2. **Poisson arrival distribution** is a statistical probability distribution that expresses the probability of a number of events (or arrivals, occurrences etc) occurring in a fixed period of time. It can be used for duration between two phone calls.

3. **Exponential arrival distribution** is a probability distribution used to assess the duration of random time intervals at a constant average rate λ. The

length of VoIP calls will be set as exponential distribution (the length of the calls = 300 seconds).

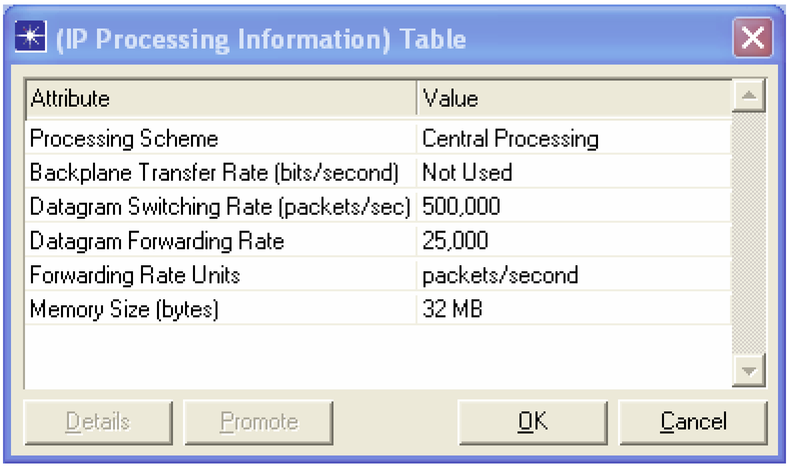
5.3 VoIP Traffic

This study assumes all the VoIP calls are point-to-point conversation therefore there is no voice conferencing.This study also ignore the signaling traffic generated by the gatekeeper because this traffic is relatively small and has limited signaling traffic for VoIP calls.

In OPNET, there are predefined component libraries that contain many vendor-specific models. However, because the simulation is for VoIP performance over Ethernet LAN and which is not based on the networks in real life, thus generic router and Ethernet switch models have been used to represent the router and the switches in the Ethernet networks. The VoIP gateway will be modelled as an Ethernet workstation in order to collect statistics inside the networks. As discussed previously, the gatekeeper signaling traffic is ignored, and not included in network model as an element. 100 Base-T links have been used to connect all network elements. Figure 5.1 (b) shows the described topology. Ethernet workstations used to model the VoIP activities. All the Ethernet workstations within the network act as parties in VoIP sessions. For example, VoIP nodes S_Client_1, and R_Client_2, S_Client_1 as a source for sending VoIP calls and R_Client_2 as a sink for receiving VoIP calls. This study only interested in the VoIP performance. Therefore, there is no other background traffic will be generated or simulated

Growth Capacity Usually, a network reserves a certain amount of network capacity to consider the future growth in users, network services, etc. Therefore, 20% -30% is a common ration of the network capacity reservation for future growth and expansion. But in this study, in order to measure the VoIP performance over Ethernet LAN, all network resources includes the router, switches, and links will be 100% utilized with no capacity reserve.

The router— In order to make this study more representative, a generic router Cisco 2612 will be adapted. The parameters are set as default value in OPNET; Cisco 2621 router has a forwarding rate of 25,000packets/second. Figure 5.3 shows the router attributes configuration.



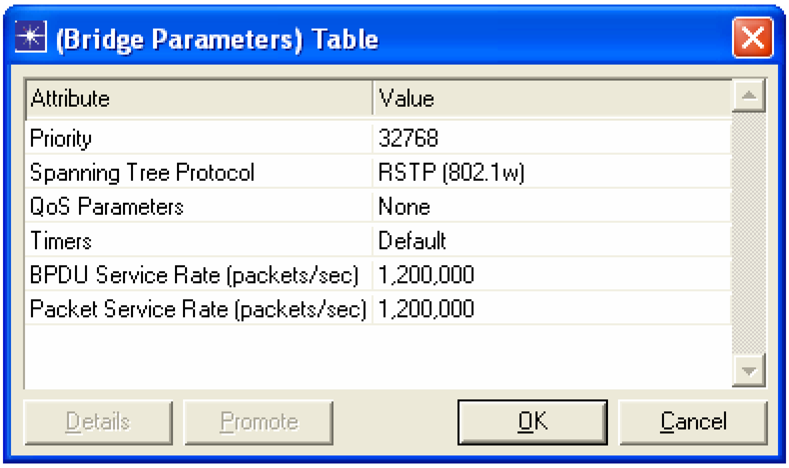
(IP Processing Information) Table

Attribute	Value
Processing Scheme	Central Processing
Backplane Transfer Rate (bits/second)	Not Used
Datagram Switching Rate (packets/sec)	500,000
Datagram Forwarding Rate	25,000
Forwarding Rate Units	packets/second
Memory Size (bytes)	32 MB

Details Promote OK Cancel

Figure 5.3 Router attributes configuration

The switches— similarly, generic 3Com switches are used. All parameters to be configured in default value, the switching speed of 3Com Superstack3 3300 is 1,200,000 packets/second. Figure 5.4 shows the switch attributes configuration.



(Bridge Parameters) Table

Attribute	Value
Priority	32768
Spanning Tree Protocol	RSTP (802.1w)
QoS Parameters	None
Timers	Default
BPDU Service Rate (packets/sec)	1,200,000
Packet Service Rate (packets/sec)	1,200,000

Details Promote OK Cancel

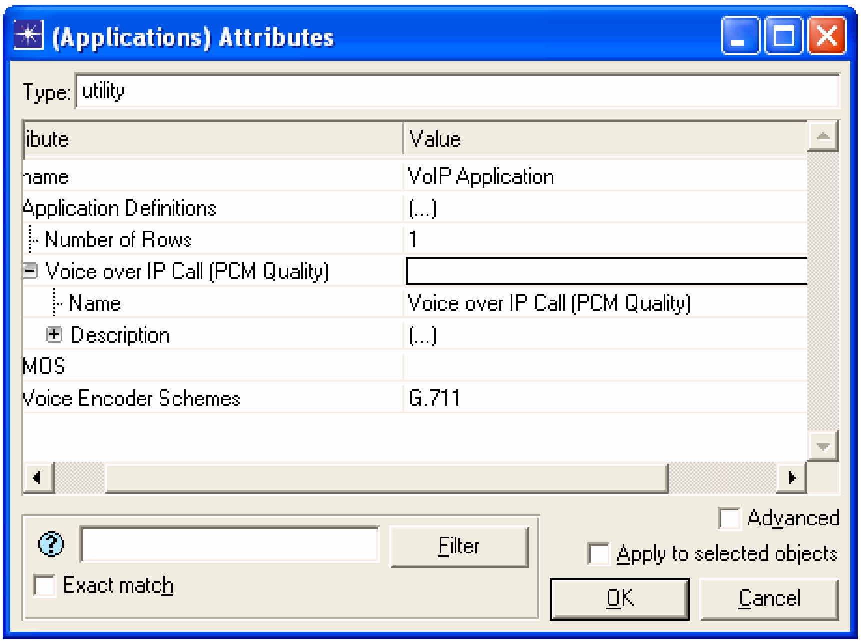
Figure 5.4 Switch attributes configuration

The links— each link in the network model will be 100% utilized.

5.4 VoIP traffic settings

5.4.1 VoIP application and profile settings

A voice application will be used to model the VoIP traffic in OPNET. An application in OPNET is a set of tasks for different phases. Each phase has different traffic behaviour that can be configured and takes place between two endpoints. Start time and duration of each task can be configured with each application. In OPNET, an "Application Definition" is used to define and configure Applications. For instance, the configurable parameters of the VoIP application are shown in Figure 5.5.



5.5 Configuration of voice application

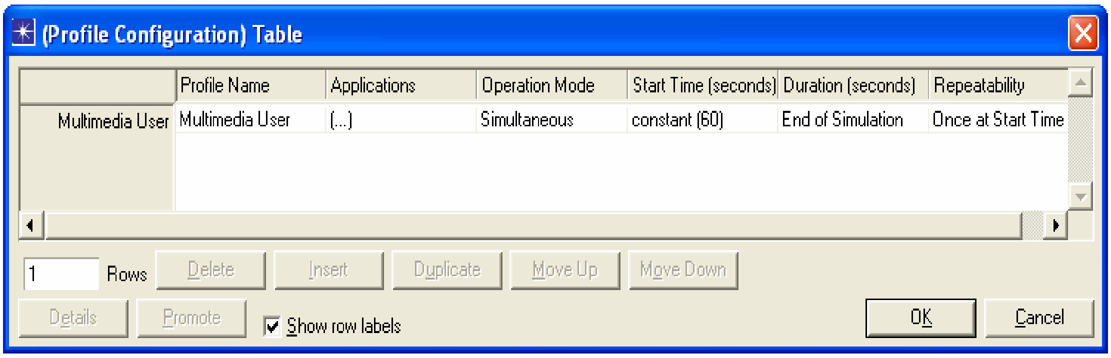
The parameter "Encoder Scheme" needs to be set to G.711, because it consumes the most bandwidth and provides the best voice quality, it is the worst-case LAN bandwidth requirement. The default value of attribute "Voice Frames per Packet" is 1. Because a voice frame in OPNET terminology is a collection of 32 voice samples and each sample size is 8 bits, thus each voice frame is 32 bytes. But the G.711 standard has a payload of 160 bytes for each VoIP packet. Thus Voice Frames per Packet attribute must be set to 5. The configuration has shown in figure 5.6 (a). Another attribute needed to be mentioned is the Symbolic Destination Name. The symbolic name is used to define the destination nodes for VoIP calls. This attribute is set to default value, which means the destination nodes of VoIP calls will be randomly chosen. Thus there are no VoIP call destination preferences; all the VoIP receiving workstations have the equal chance to receive a VoIP call.

The next step is how the workstations (nodes) will be implementing this VoIP application. A profile will be used to define the behaviour of a network workstation, profile can contains one or more applications and these applications can be configured by repeatability, start time and end times and etc. There is only one profile in the network model called VoIP_Profile. The VoIP calling clients will be designated to generate VoIP calls, thus we need to configure these workstations to support this VoIP profile. All the designated VoIP calling workstations' name will be started by letter "S", i.e. S_C_1. The same application profile is used for all nodes.

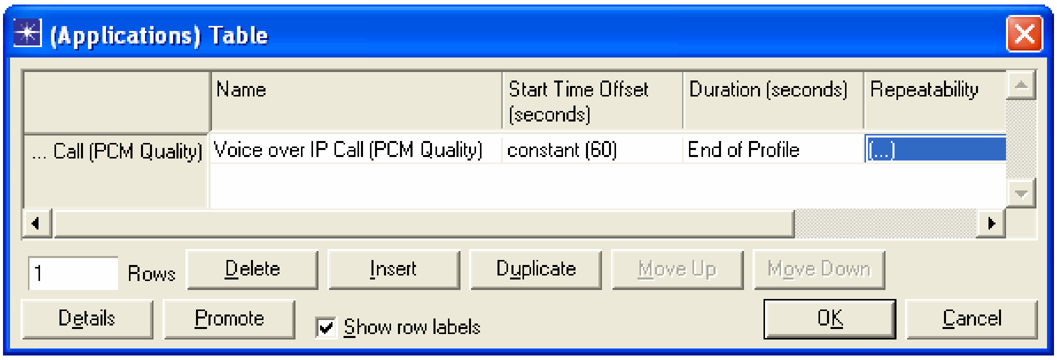
Except scenario 3, profile will be configured to generate VoIP traffic at a constant rate for all other scenarios. In this simulation, the first VoIP call will be generate 60 seconds later after the simulation starts; 60 seconds for simulation to warm up, and then add a call every 5 seconds. To do this, the VoIP application and VoIP profile need to be defined as shown in Figure 5.6 (a)

To achieve this, attribute "Start Time Offset" of the VoIP traffic is set to 60 seconds and attribute "Start Time Offset" for VoIP call to 60 seconds. The configuration of repeatability of the VoIP application is set to be "Unlimited", and the "interrepetition time" is set to 5 seconds, so it can keep generating VoIP calls every five seconds. The configurations as shown in Figure 5.6 (b), (c)

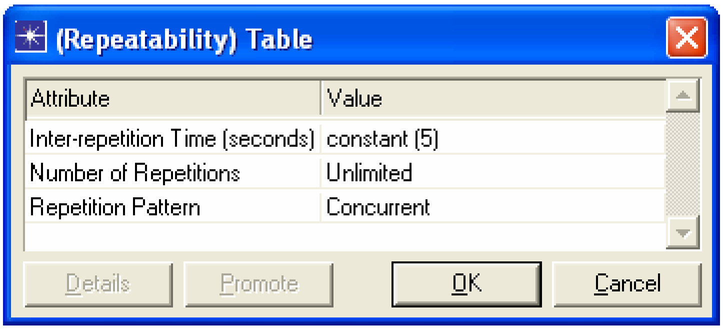
VoIP calling workstations are configured to support the VoIP_Profile by adding this profile to each workstation's supported profiles lists. Similarly, VoIP receiving workstations are configured to support the VoIP_Profile by adding this profile to each workstation's supported service lists.



(a) Profile parameters



(b) Call duration configuration

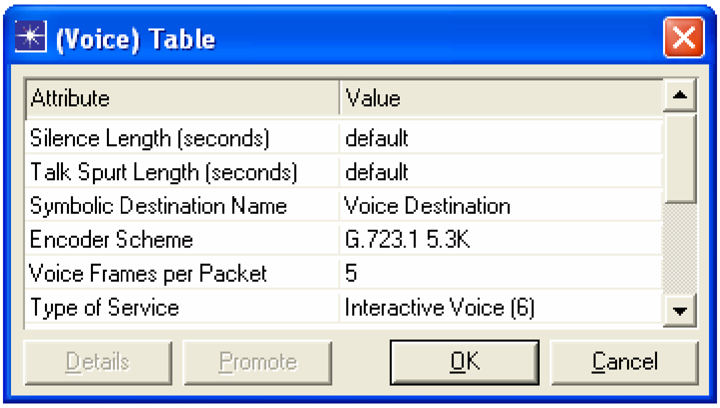


(c) Inter-arrival time

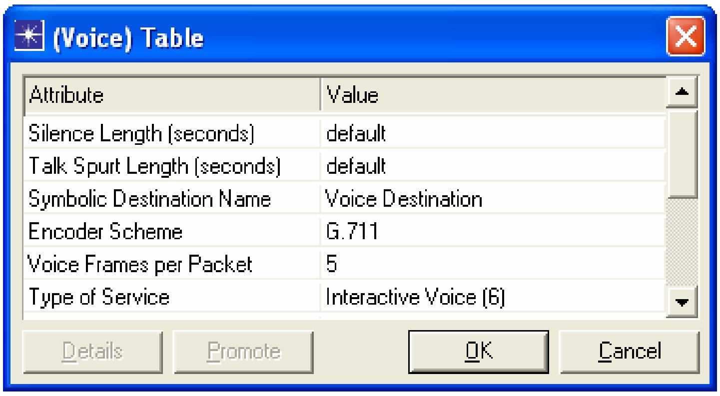
Figure 5.6: VoIP configurations (a) Profile parameters; (b) Call duration configuration; and (c) Inter-arrival time

5.4.2 VoIP traffic encoder settings

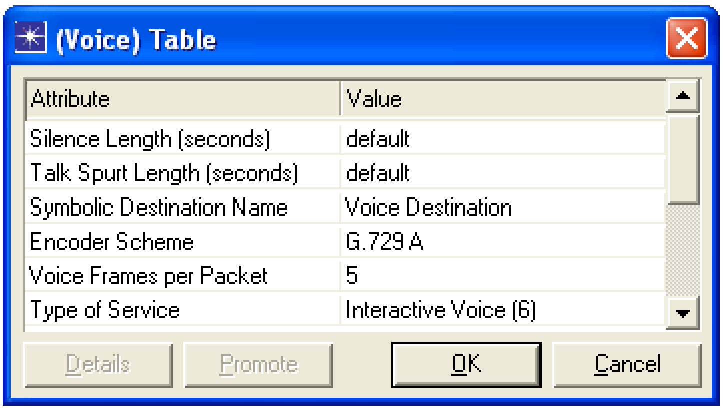
The parameter 'Encoder Scheme' is set to G.711 for scenario 1, 2, and 4. As discussed earlier in this chapter, Voice Frames per Packet attribute is set to 5. In scenario 3, G.723 (5k) and G.729 are also set to 5. The parameter details of each codec have shown in Figure 5.7 (a), (b), (c)



(a) G.711 Codec Parameters



(b) G.723 (5.3k) Codec Parameters

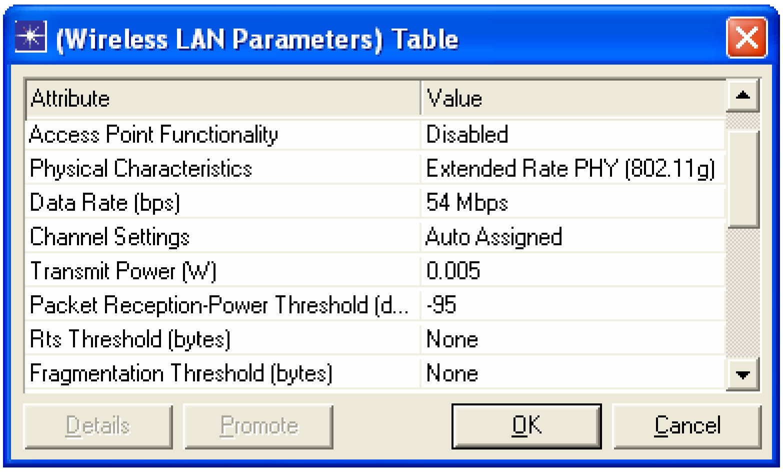


(b) G.729 (8k) Codec Parameters

Figure 5.7: Configurations of Voice Encoder Scheme parameters (a) Profile parameters; (b) Call duration configuration; and (c) Inter-arrival time

5.4.3 Wireless LAN Parameters

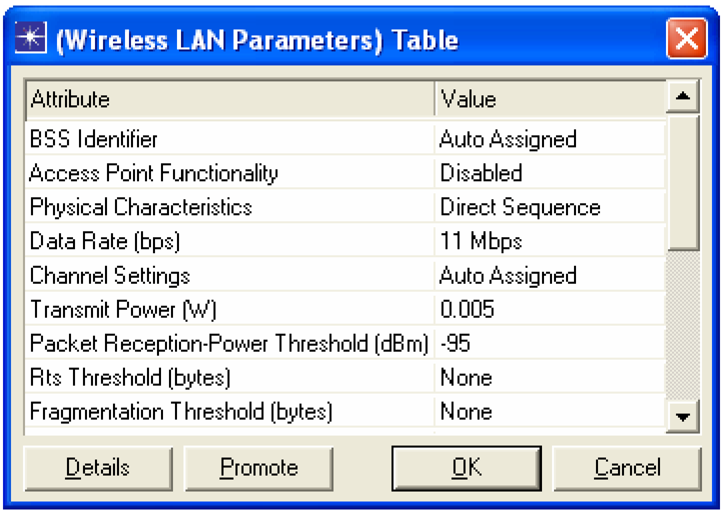
Data Rate: 54Mbps. Physical Characteristics: Extended Rate (802.11g). Buffer Size: This attribute specifies the maximum length of the higher layer data buffer. In this scenario, buffer size is set to default value which is 256,000 bits. Channel Settings: Auto assigned. The detailed configuration for each wireless workstation as shown in Figure 5.8 (a)



Attribute	Value
Access Point Functionality	Disabled
Physical Characteristics	Extended Rate PHY (802.11g)
Data Rate (bps)	54 Mbps
Channel Settings	Auto Assigned
Transmit Power (W)	0.005
Packet Reception-Power Threshold (d...	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (bytes)	None

Figure 5.8(a) Wireless LAN Parameters （802.11g）

Parameters for 802.11b protocol are very similar to 802.11g except attributes “Data Rate” and “Physical Characteristics”. Data: 11Mbps. Physical Characteristics: Direct Sequence (802.11b). The detailed configuration for each wireless workstation as shown in Figure 5.8 (b)



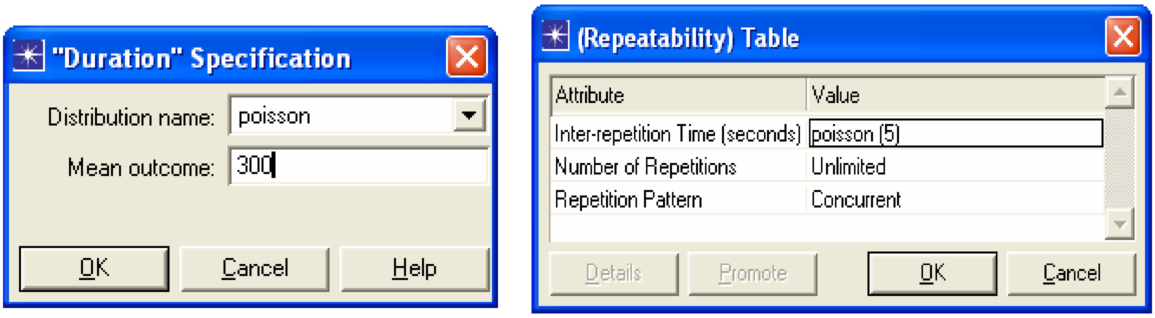
Attribute	Value
BSS Identifier	Auto Assigned
Access Point Functionality	Disabled
Physical Characteristics	Direct Sequence
Data Rate (bps)	11 Mbps
Channel Settings	Auto Assigned
Transmit Power (W)	0.005
Packet Reception-Power Threshold (dBm)	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (bytes)	None

Figure 5.8 (b) Wireless LAN Parameters （802.11b）

Figure 5.8 Wireless LAN Parameters

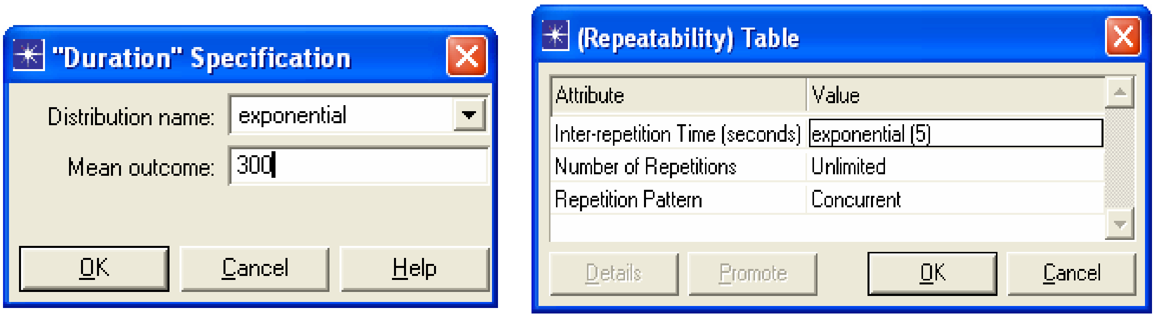
5.4.4 Traffic distribution settings

The configurations for Poisson traffic distribution and exponential traffic distribution as shown in Figure 5.9 and Figure 5.10.



(a) Call Duration Settings **(b) Call inter-arrival settings**

Figure 5.9 Poisson Traffic Distribution Settings



(a) Call Duration Settings **(b) Call inter-arrival settings**

Figure 5.10 Exponential Traffic Distribution Settings

Chapter 6

Results and Analysis

In Chapter 5, a detailed simulation network modelling were described. In order to obtain graphical results, before running simulation, a number of statistics in OPNET need to be configured for VoIP network components include VoIP traffic, switches, router, and links. This chapter presents simulation results for performance prediction of VoIP.

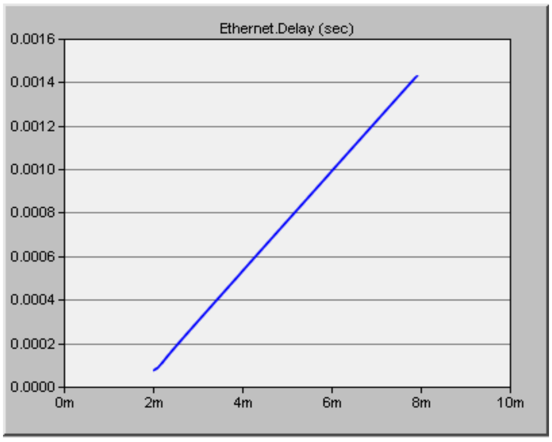
The duration of OPNET simulation was set to 8 minutes (duration time for campus network model was set to 4 minutes due to memory limitation). The VoIP traffic started at 120 seconds after the simulation is initially started. Every simulation stops at 8 minutes, the statistical and graphical results are generated by OPNET.

6.1 Scenario 1: Impact of Increasing Number of VoIP Clients

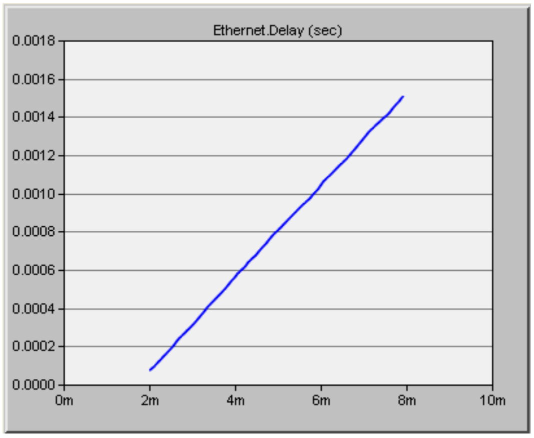
The first scenario tested the impact of increasing number of VoIP clients to network performance. The number of VoIP clients initialed from two nodes to 400 nodes.

Figure 6.1 shows network packet delay. The OPNET default reported delay configuration is the sample mean. Figure 6.1 (a) shows Ethernet delay is less than 1ms when there is only two VoIP clients on a wired Ethernet LAN. As seen in Figures 6.1 a to i, Ethernet network delay steadily increases as the number of VoIP nodes is small. The network with 20 VoIP nodes can yield about 1.6ms Ethernet delay. The Ethernet delay increases to around 9 ms for 120 nodes (Figure 6.1(i)).

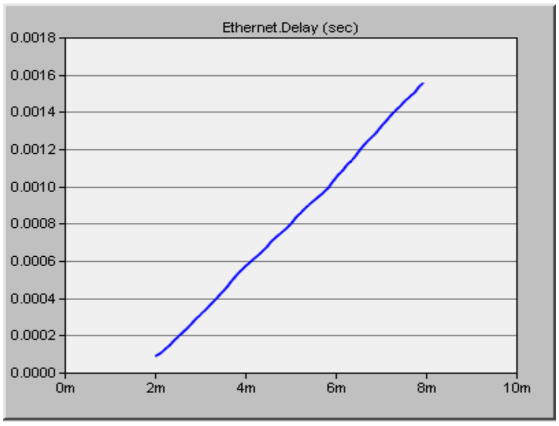
However, as seen in Figure 6.1 (j) and (k), as the number of VoIP clients increased to 200 nodes and 400 nodes, the Ethernet delays rapidly increases to more than 1 second, and the actual delay could be even much more than 1 second as the simulation time is only 4 minutes (the reason as mentioned above).



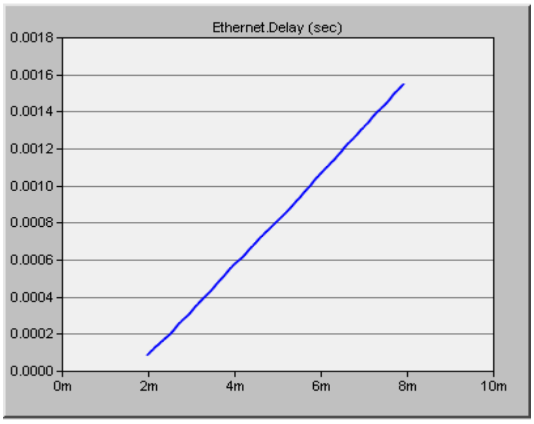
(a) Ethernet Delay (Wired Nodes = 2)



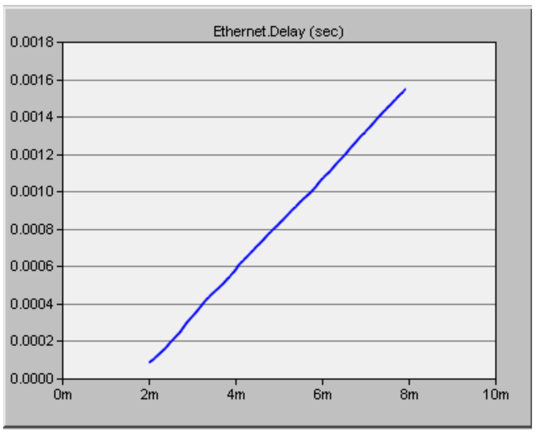
(b) Ethernet Delay (Wired Nodes = 4)



(c) Ethernet Delay (Wired Nodes = 8)

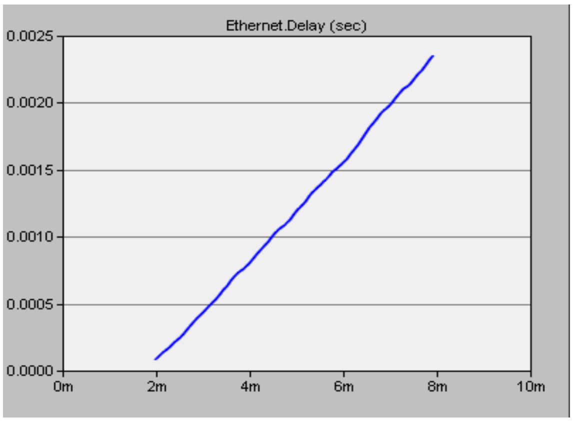


(d) Ethernet Delay (Wired Nodes = 10)

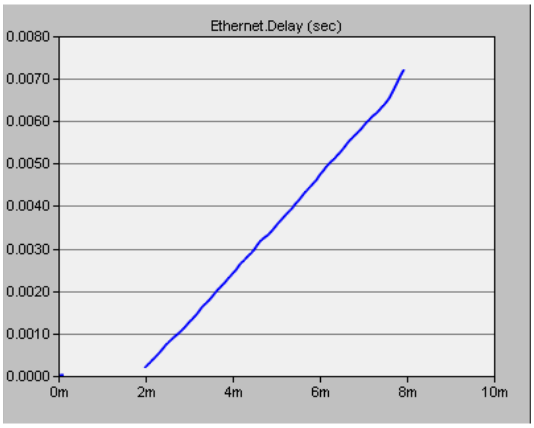


(e) Ethernet Delay (Wired Nodes = 16)

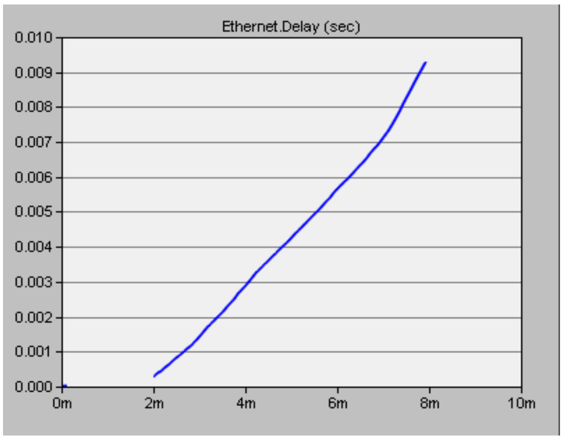
(f) Ethernet Delay (Wired Nodes = 20)



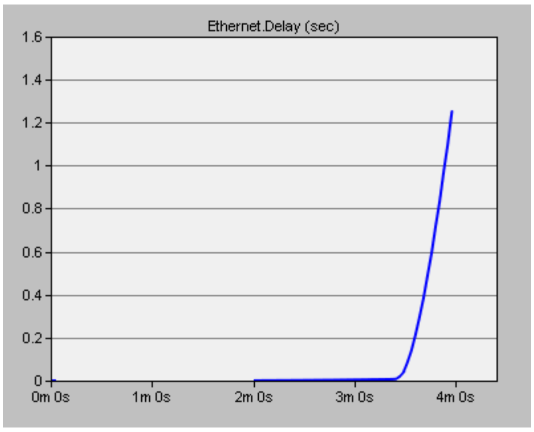
(g) Ethernet Delay (Wired Nodes = 40)



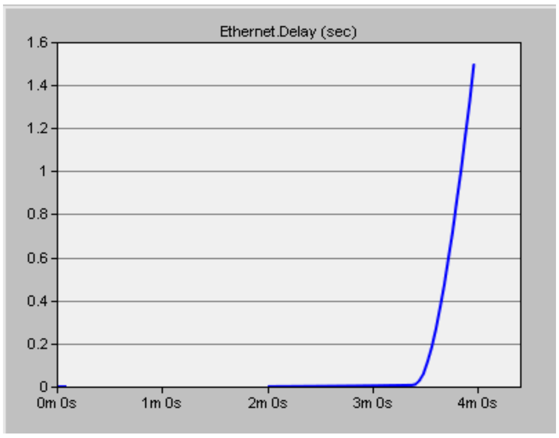
(h) Ethernet Delay (Wired Nodes = 80)



(i) Ethernet Delay (Wired Nodes = 120)

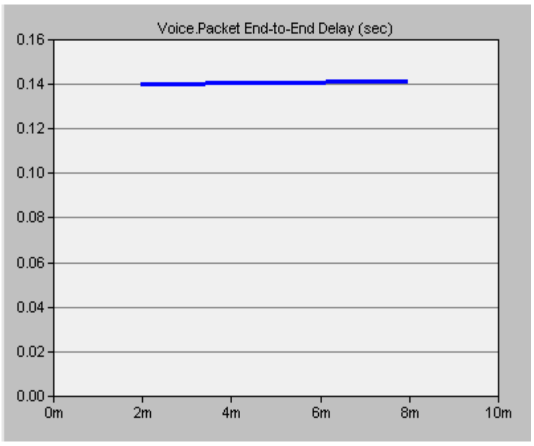


(j) Ethernet Delay (Wired Nodes = 200)

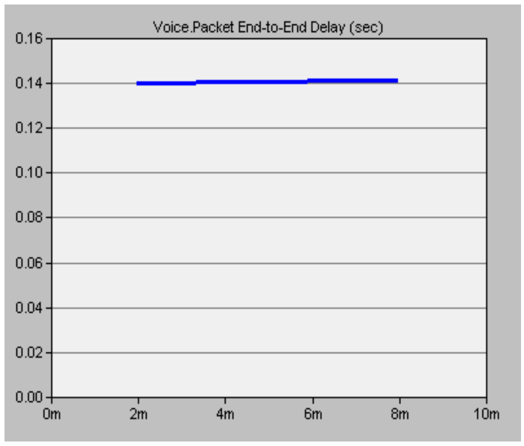


(k) Ethernet Delay (Wired Nodes = 400)

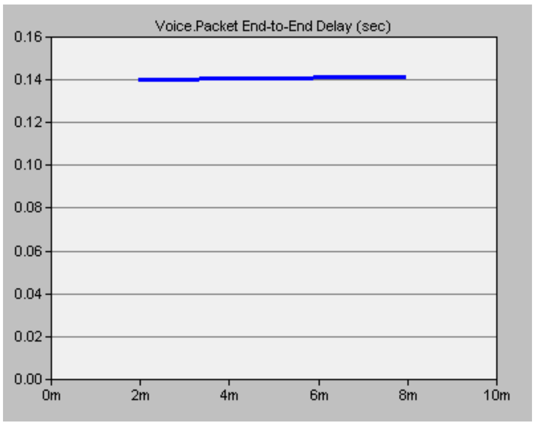
Figure 6.1 Ethernet Delay



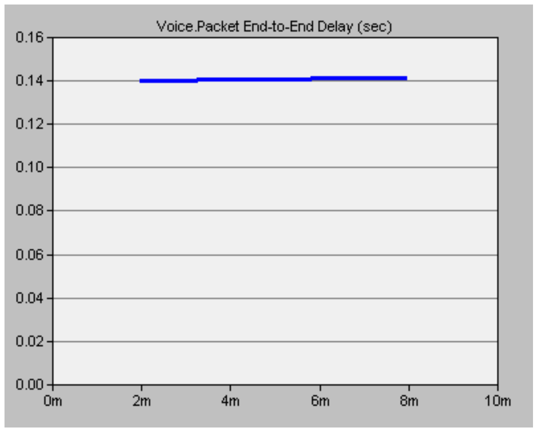
(a) End-to-End Packet Delay
(Wired Nodes = 2)



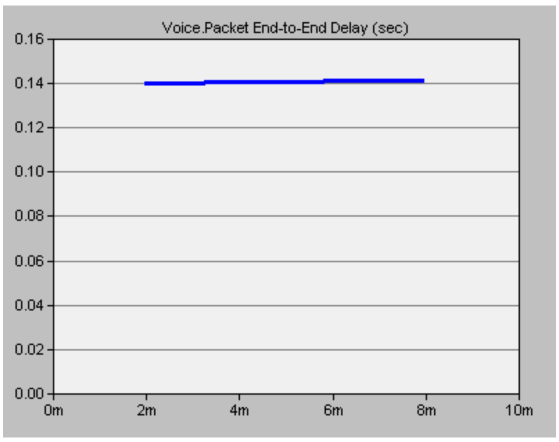
(b) End-to-End Packet Delay
(Wired Nodes = 4)



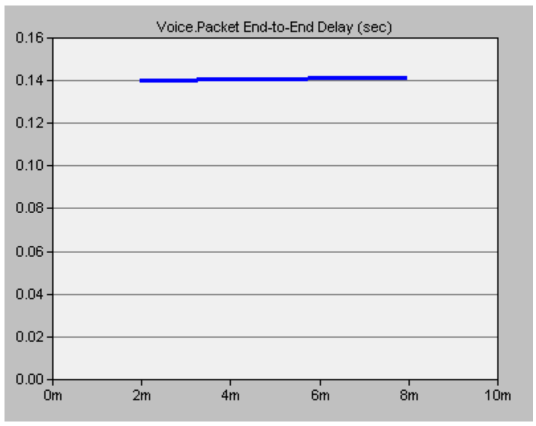
(c) End-to-End Packet Delay
(Wired Nodes = 8)



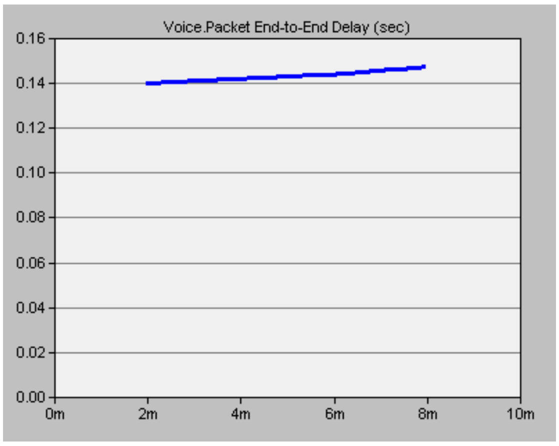
(d) End-to-End Packet Delay
(Wired Nodes = 10)



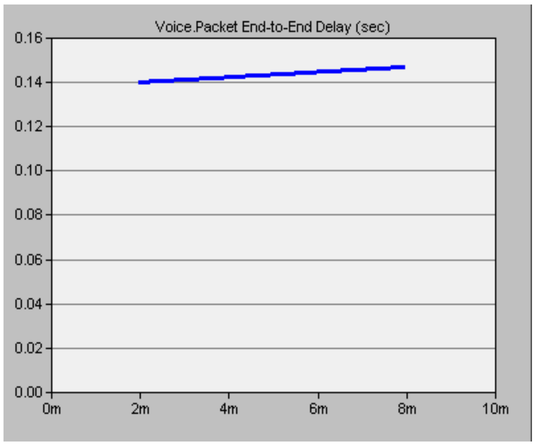
(e) End-to-End Packet Delay
(Wired Nodes = 16)



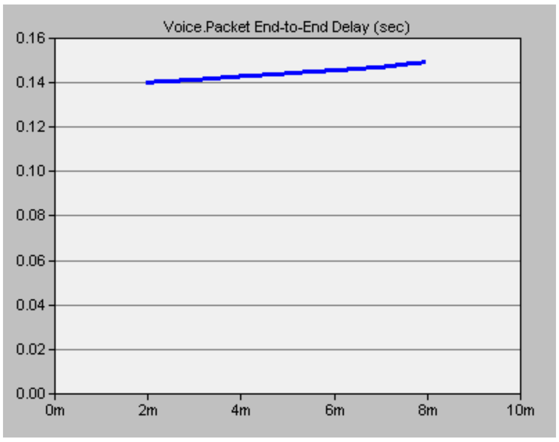
(f) End-to-End Packet Delay
(Wired Nodes = 20)



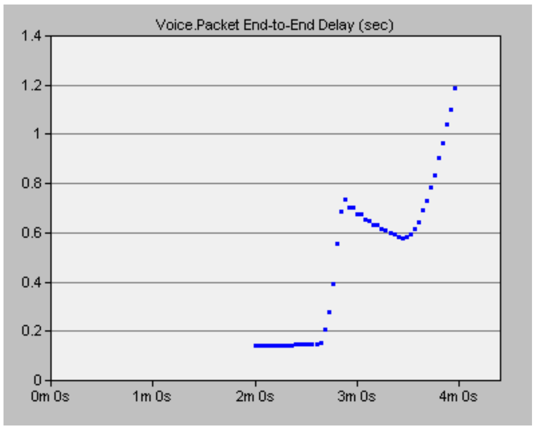
(g) End-to-End Packet Delay
(Wired Nodes = 40)



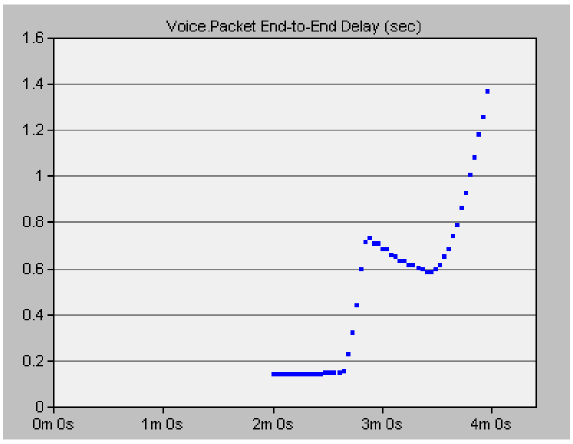
(h) End-to-End Packet Delay
(Wired Nodes = 80)



(i) End-to-End Packet Delay
(Wired Nodes = 120)



(j) End-to-End Packet Delay
(Wired Nodes = 200)



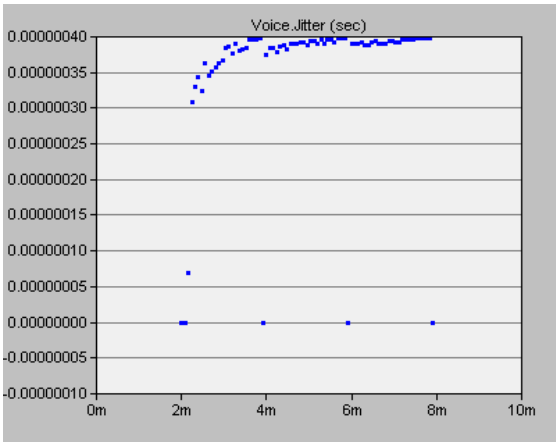
(k) End-to-End Packet Delay
(Wired Nodes = 400)

Figure 6.2 End-to-End Packet Delay

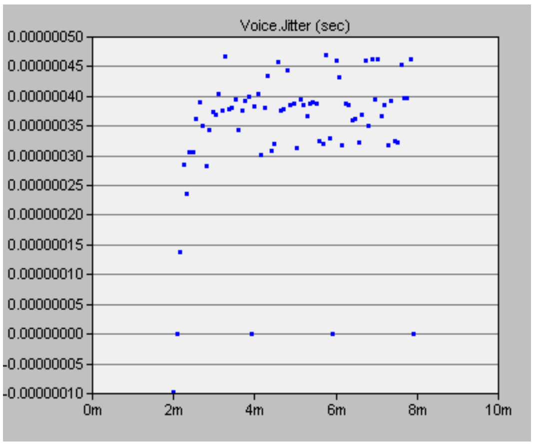
Figure 6.2 shows the corresponding VoIP end-to-end delay. As seen in (a) to (i), Voice packet end-to-end delay increases as the number of VoIP nodes increases. It is around 140ms when the number of VoIP nodes is less than 20 and somewhere between 140ms and 150ms when 120 nodes added to the network. Thus the Voice packet end-to-end delay is more or less than 150ms.

However, one important thing need to be mentioned here, the voice application end-to-end delay does not include the delays from VoIP gateway, the codecs. Therefore, delays from these sources also need to be considered (average end-to-end delay of 53ms).This gives a total delay around 200 ms, 200 ms is acceptable as discussed earlier in Chapter 2.

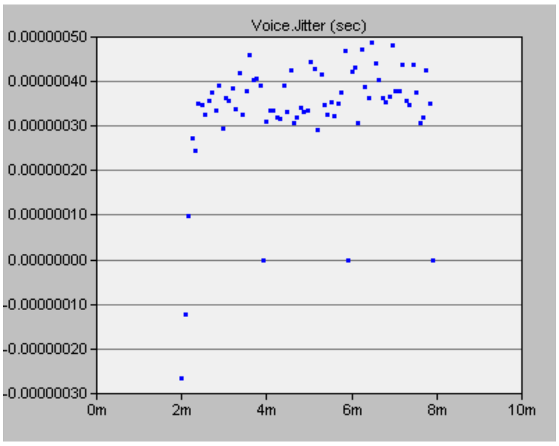
The voice packet end-to-end delay increases sharply after more VoIP nodes added to the network. The Voice packet end-to-end delay increase rapidly to 1.2 second when the number of VoIP nodes increased to 200, it increases to 1.5 second when there are 400 VoIP clients. This brings the voice delays much more than 200ms which are totally unacceptable values.



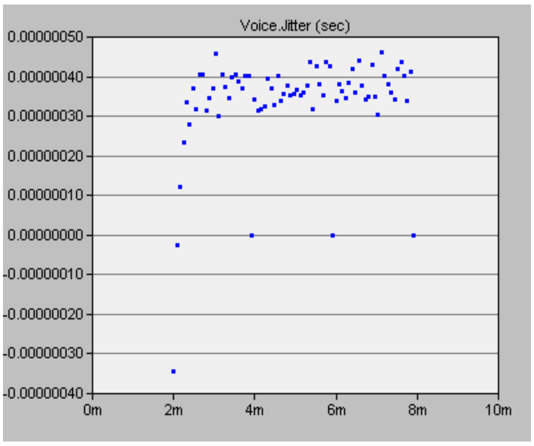
(a) Voice Jitter (Wired Node = 2)



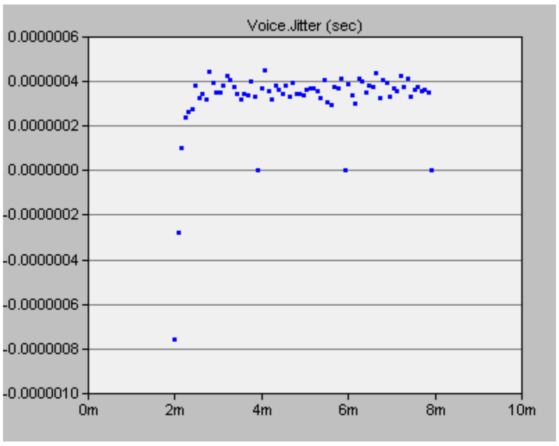
(b) Voice Jitter (Wired Node =4)



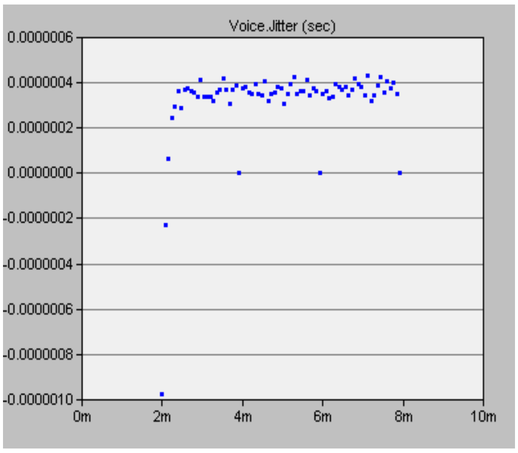
(c) Voice Jitter (Wired Node = 8)



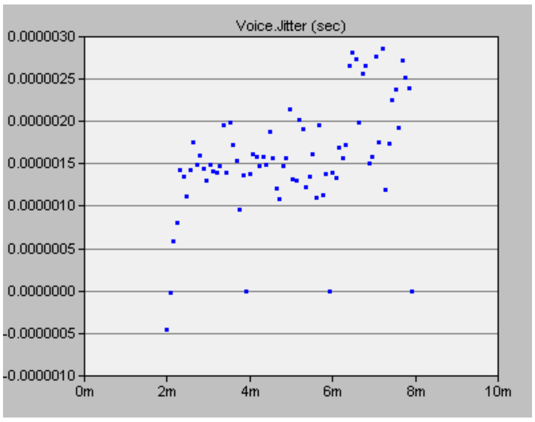
(d) Voice Jitter (Wired Node = 10)



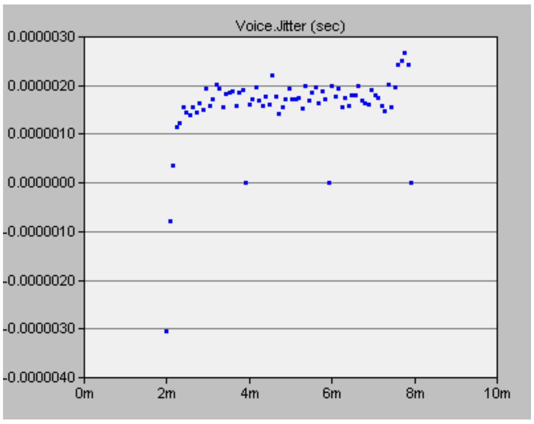
(e) Voice Jitter (Wired Node = 16)



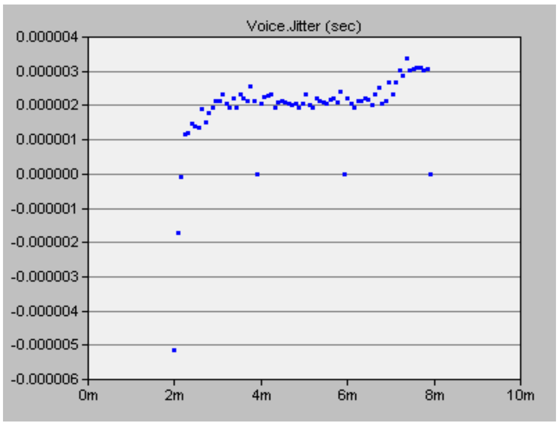
(f) Voice Jitter (Wired Node = 20)



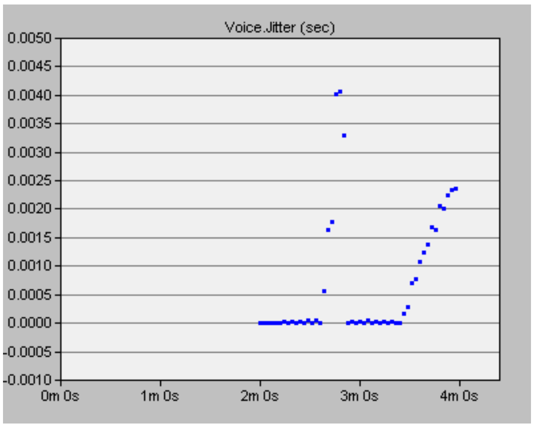
(g) Voice Jitter (Wired Node = 40)



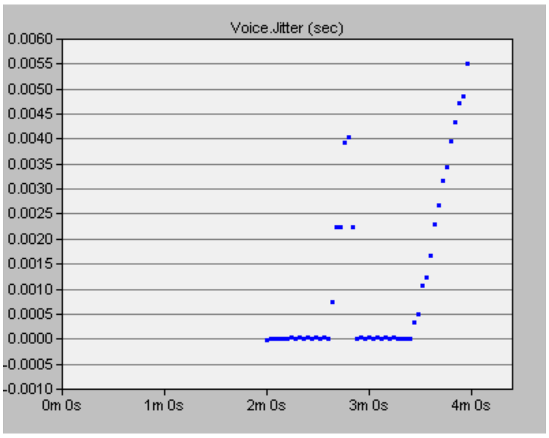
(h) Voice Jitter (Wired Node = 80)



(i) Voice Jitter (Wired Node = 120)



(j) Voice Jitter (Wired Node = 200)



(k) Voice Jitter (Wired Node = 400)

Figure 6.3 Voice Jitter

Figure 6.3 shows the voice jitter. The Voice jitter increases as the number of nodes increase but not much. When the number of node is small (2 nodes to 20 nodes), the jitter for all packets are less than 1ms (see Figure 6.2 (a.)). As the conversations last (Figure 6.2 (b.), (c.), (d.), (e.), (f.), (g.), (h.), (i.)), more packets jitter have generated.

This study configured the duration of the OPNET simulation run for 8 minutes. The The VoIP traffic started at 120 seconds at which a total of three VoIP bidirectional calls are initially added. Then, every 5 seconds 10 VoIP calls are added. The simulation stops at 8 minutes, therefore the total calls got generated. When the simulation time is close to 480 seconds, there are about 4330 concurrent VoIP calls. 10 + (480s -120s) / 5s * 60. These are the maximum concurrent calls during the simulation period. Some packets yield the highest delay (above 10ms). Note here ten calls will be added at 7 minutes and 55 seconds since the simulation stops at 8 minutes,

As seen from these graphs, the impact of increasing the number of VoIP nodes is relatively little when the number of VoIP client is small (less than 20 nodes). However, as the number of VoIP clients increases, the impact to the VoIP performance over the network becomes significant.

6.2 Scenario 2: Impact of Wireless Nodes

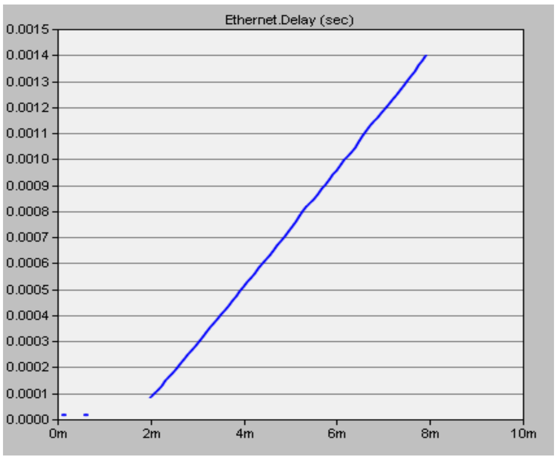
The main findings are listed in Table 6.1. In this scenario, the simulation results are shown in Figure 6.4 to 6.6, each of these figures includes a number of figures. These figures show how results changed when the number of wireless clients and the protocol changed (wireless client number from two to six, wireless protocol from 802.11b to 802.11g). Figure 6.4 (a) (b), 6.5 (a) (b), 6.6 (a) (b) show Ethernet delay. Figure 6.4 (c) (d), 6.5 (c) (d), 6.6 (c) (d) show Voice Jitter. Figure 6.4 (e) (f), 6.5 (e) (f), 6.6 (e) (f) show Voice Packed End-to-End Delay. Figure 6.4 (g) (h), 6.5 (g) (h), 6.6 (g) (h) show Wireless LAN Delay. Figure 6.4 (i) (j), 6.5 (i) (j), 6.6 (i) (j) show Wireless LAN Throughput. Figure 6.4 (k) (l), 6.5 (k) (l), 6.6 (k) (l) show Wireless LAN Data Dropped.

This network model includes both wired and wireless components. There are 20 wired VoIP clients in this network model, and the number of the wireless VoIP clients increased from 2 to 6. The simulation results show that the performance of the wireless network were quite poor when compared to the wired network. The number of wireless nodes is increased from 2 nodes to 6 nodes and the performance are tested through both 802.11b and 802.11g protocols.

As seen in Figure 6.4 (a) (b), 6.5 (a) (b), 6.6 (a) (b), the Ethernet LAN delay changes slightly and values around 1ms to 1.5ms. However, as can be seen from Figure 6.4 (g) (h), 6.5 (g) (h), 6.6 (g) (h), the simulation results show that the wireless LAN delay changes rapidly, under 802.11b protocol, 2 wireless nodes yield only around 230ms wireless LAN delay. The wireless LAN delay rapidly increased to around 550ms when the number of wireless nodes increased to 6. As discussed earlier in this study, delay is sensitive to the VoIP services as the information in a phone call cannot be lost. In addition, the high delay results indicate bad quality of the conversation and the VoIP clients can not t understand each other. 802.11g protocol has much better results than 802.11b protocol, 2 wireless nodes only yield around 26ms wireless LAN delay, and the wireless LAN delay is around 145ms when the number of wireless nodes increased to 6.

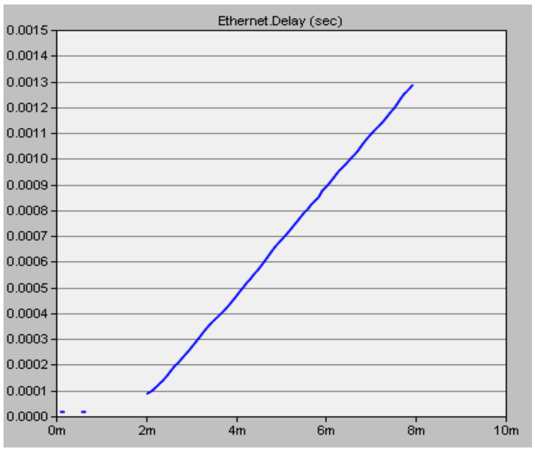
Figure 6.4 (c) (d), 6.5 (c) (d), 6.6 (c) (d) show the jitter's variance starts to increase as more calls are added, more jitters can be seen especially after 3 minutes. This happened to both 802.11b and 802.11g protocols. In general, 802.11b protocol controls jitters better than 802.11g protocol. However, the jitter did not affect the performance too much since it is in the acceptable range (less than 1ms) that can be ignored.

Figure 6.4 (e) (f), 6.5 (e) (f), 6.6 (e) (f) show the voice packet end-to-end delay. The voice packet end-to-end delay under 802.11b protocol changes from around 160ms (2 nodes) to 250ms (6 nodes). 802.11g protocol also has much better results than 802.11b protocol, the voice packet end-to-end delay changes from around 145ms when 2 wireless nodes in the network to around 160ms when the number of wireless nodes increased to 6.



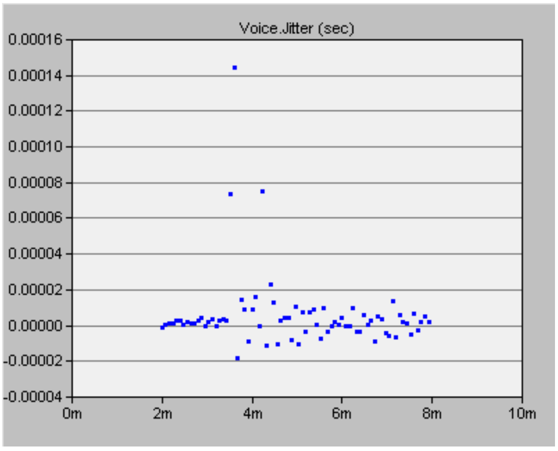
(a) Ethernet delay

Wired nodes=20, Wireless Nodes = 2, 802.11b



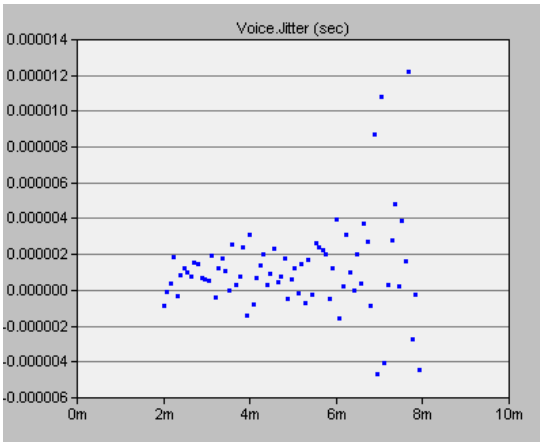
(b) Ethernet delay

Wired nodes=20, Wireless Nodes = 2, 802.11g



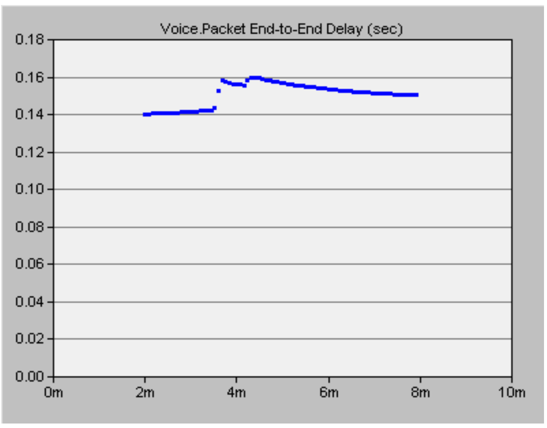
(c) Voice Jitter

Wired nodes=20, Wireless Nodes = 2, 802.11b



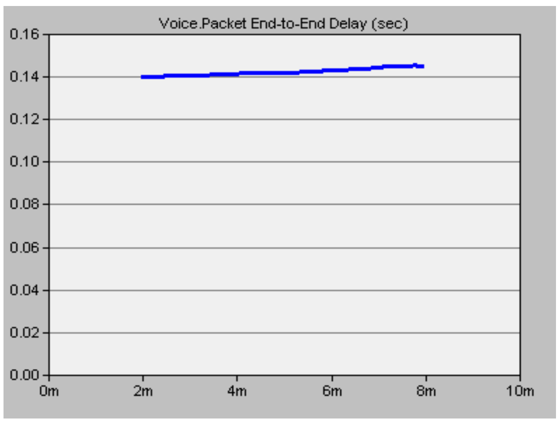
(d) Voice Jitter

Wired nodes=20, Wireless Nodes = 2, 802.11g



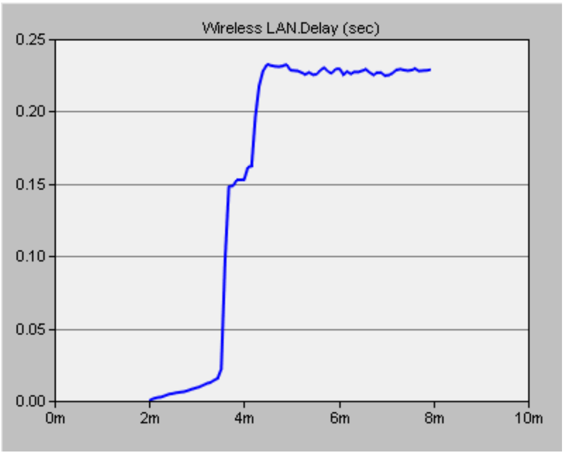
(e) Voice Packed End-to-End Delay

Wired nodes=20, Wireless Nodes = 4, 802.11b

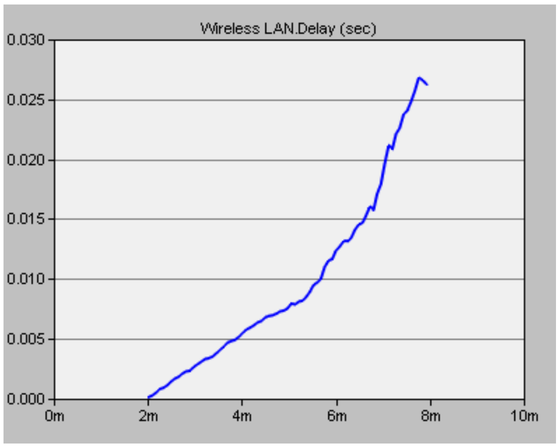


(f) Voice Packed End-to-End Delay

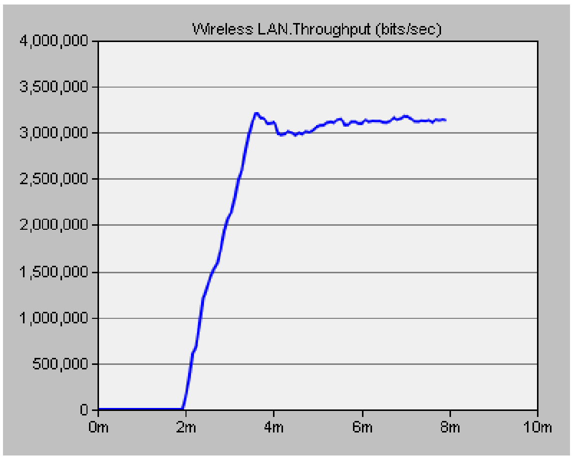
Wired nodes=20, Wireless Nodes = 4, 802.11g



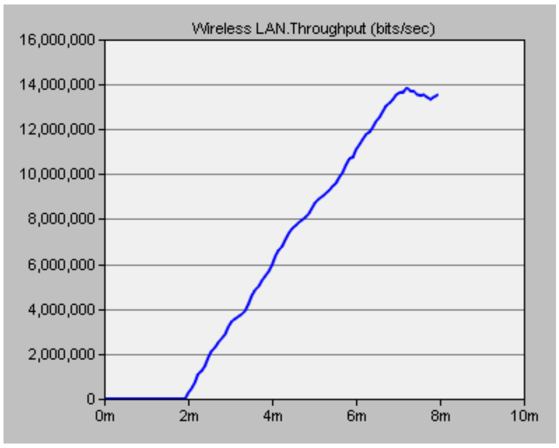
(g) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 2, 802.11b



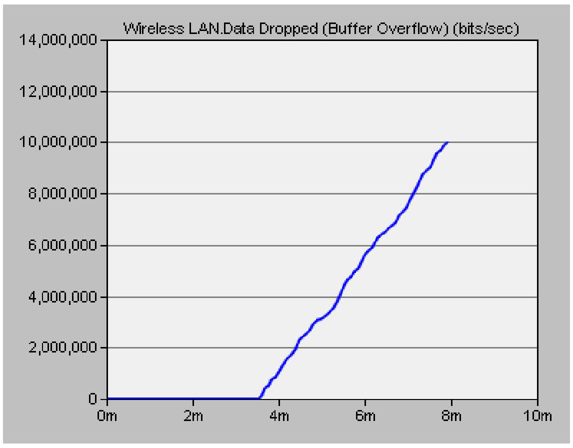
(h) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 2, 802.11g



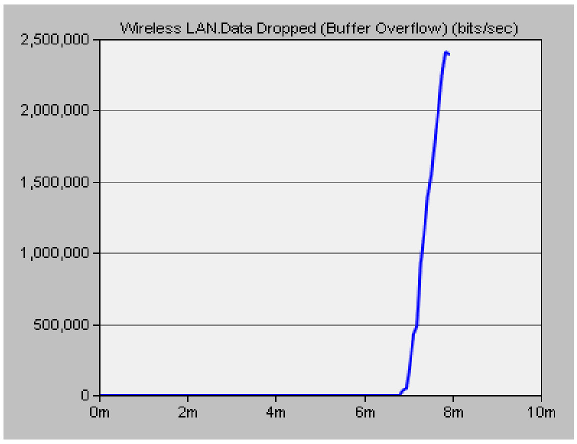
(i) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 2, 802.11b



(i) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 2, 802.11g

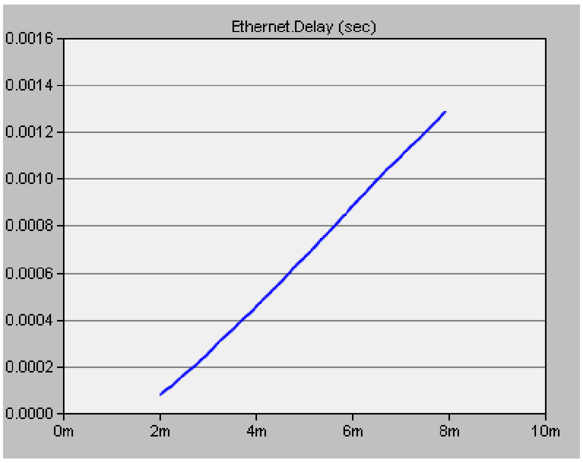


(k) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 2, 802.11b



(l) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 2, 802.11g

Figure 6.4 Wireless LAN Performance (Wireless Nodes = 2)

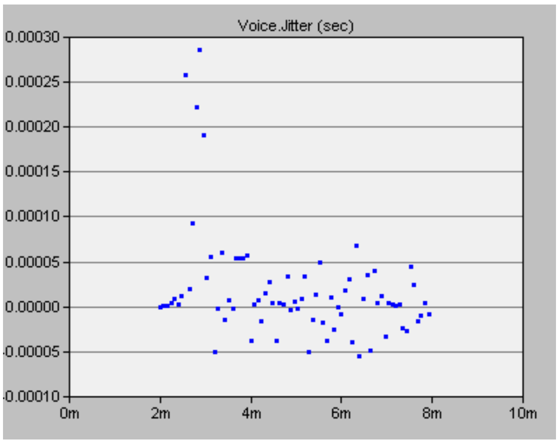


(a) Ethernet delay

Wired nodes=20, Wireless Nodes = 4, 802.11b

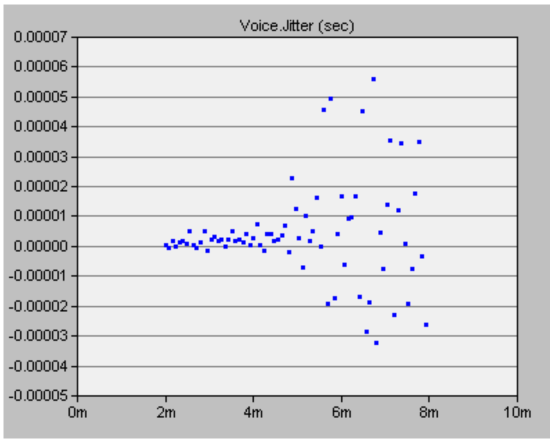
(b) Ethernet delay

Wired nodes=20, Wireless Nodes = 4, 802.11g



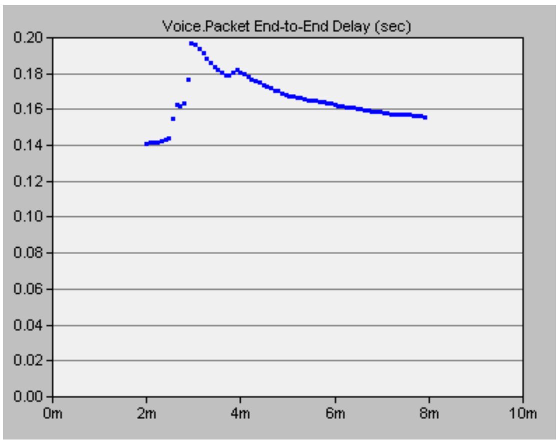
(c) Voice Jitter

Wired nodes=20, Wireless Nodes = 4, 802.11b



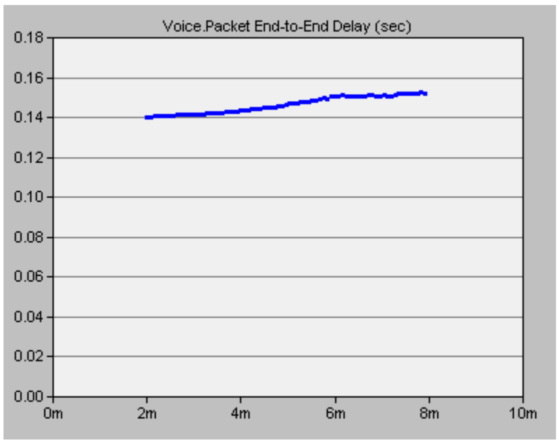
(d) Voice Jitter

Wired nodes=20, Wireless Nodes = 4, 802.11g



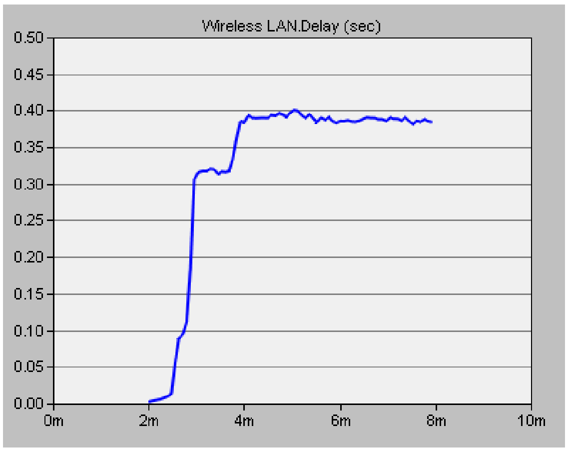
(e) Voice Packed End-to-End Delay

Wired nodes=20, Wireless Nodes = 4, 802.11b

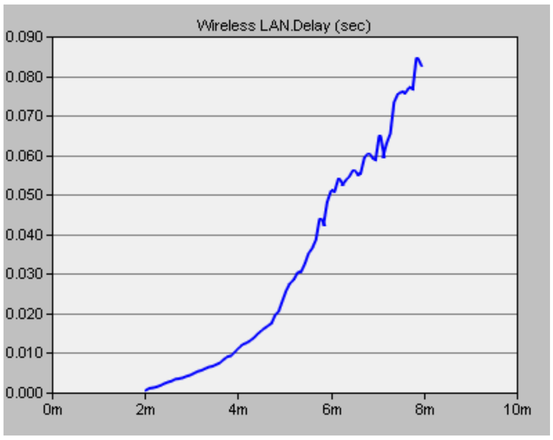


(f) Voice Packed End-to-End Delay

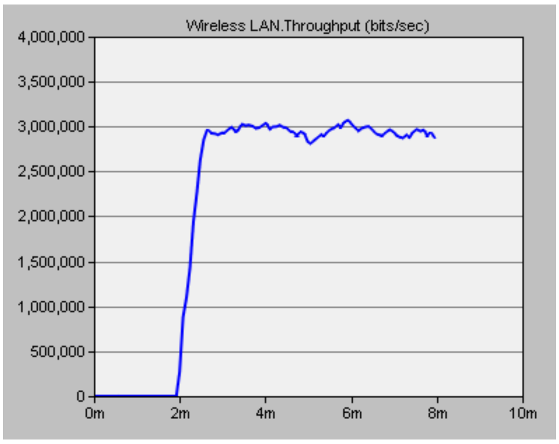
Wired nodes=20, Wireless Nodes = 4, 802.11g



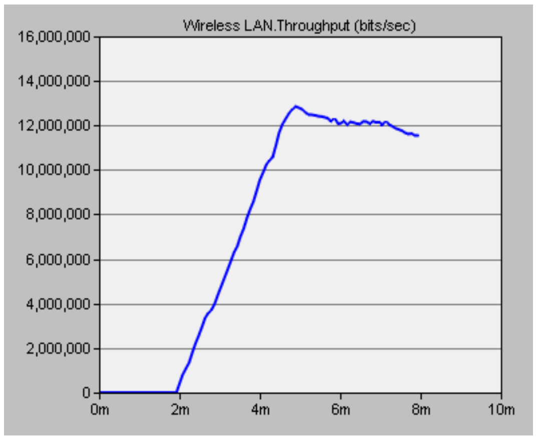
(g) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 4, 802.11b



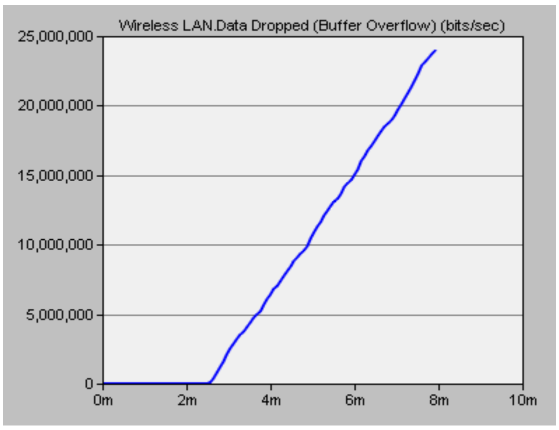
(h) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 4, 802.11g



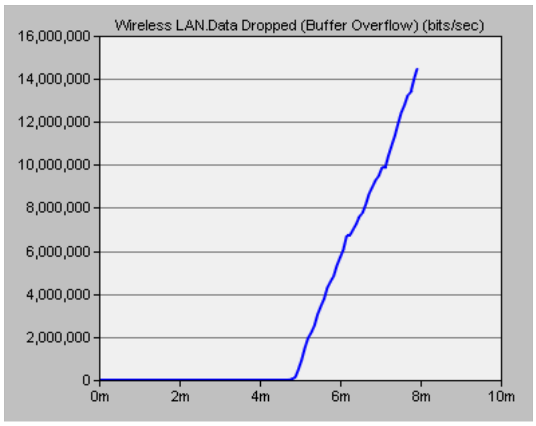
(i) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 4, 802.11b



(j) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 4, 802.11g

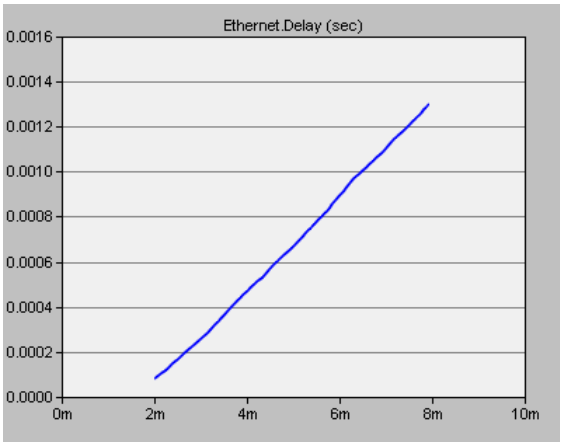


(k) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 4, 802.11b



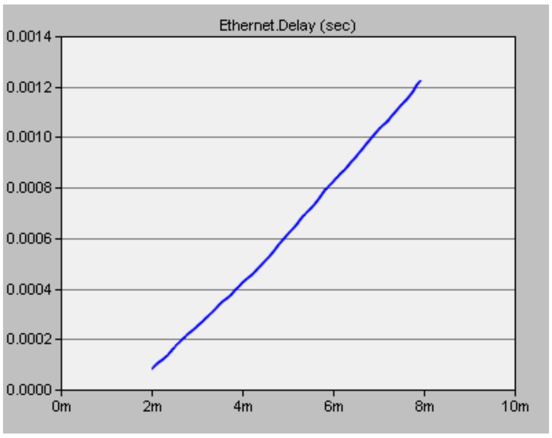
(l) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 4, 802.11g

Figure 6.5 Wireless LAN Performance (Wireless Nodes = 4)



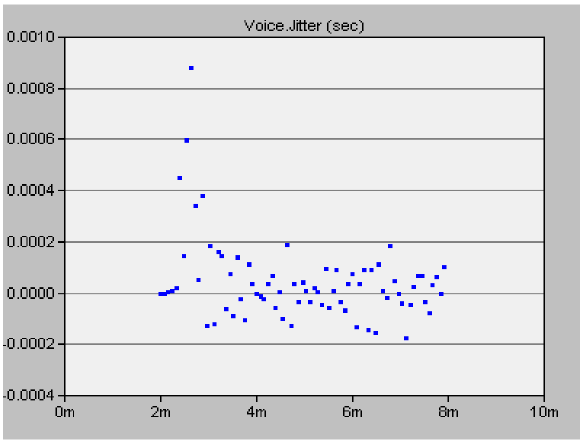
(a) Ethernet delay

Wired nodes=20, Wireless Nodes = 6, 802.11b



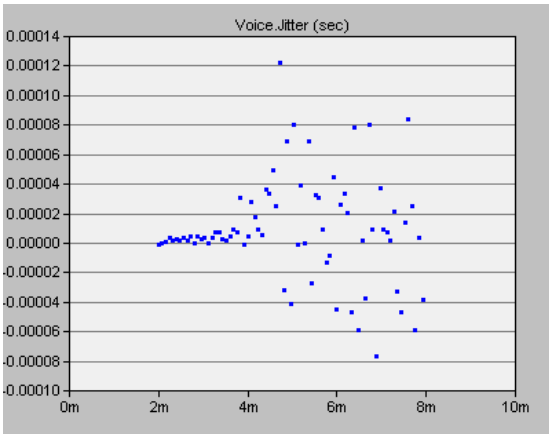
(b) Ethernet delay

Wired nodes=20, Wireless Nodes = 6, 802.11g



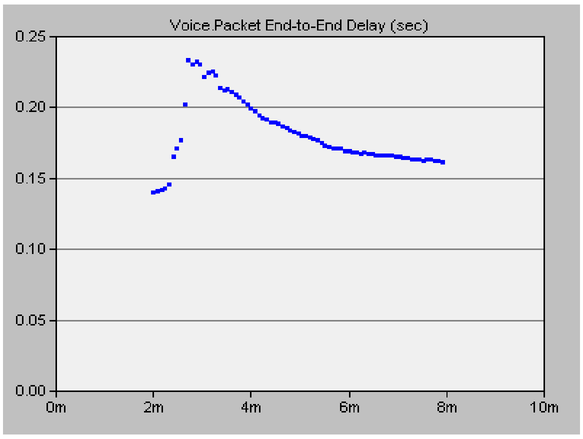
(c) Voice Jitter

Wired nodes=20, Wireless Nodes = 6, 802.11b



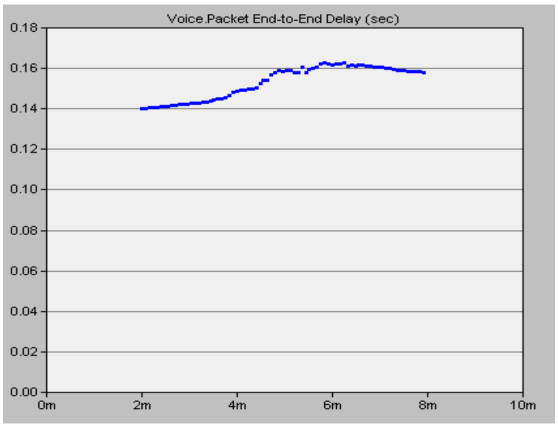
(d) Voice Jitter

Wired nodes=20, Wireless Nodes = 6, 802.11g



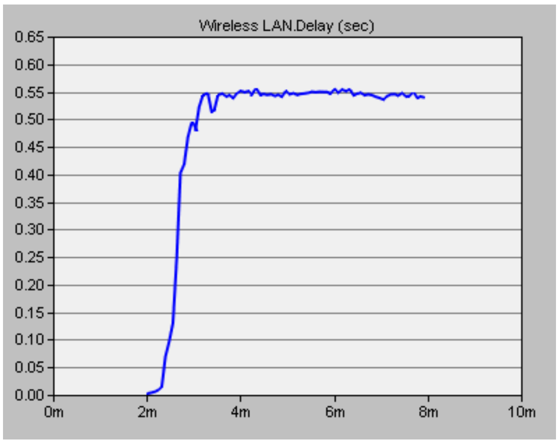
(e) Voice Packed End-to-End Delay

Wired nodes=20, Wireless Nodes = 6, 802.11b

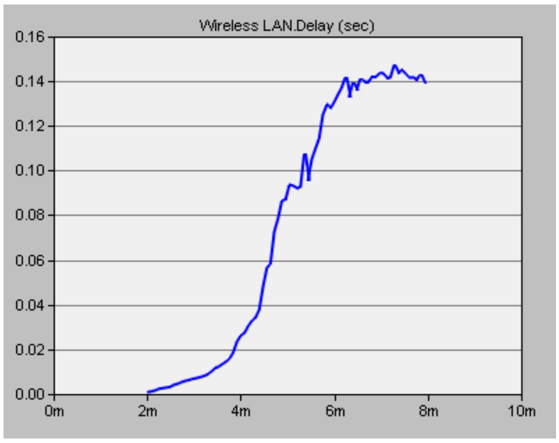


(f) Voice Packed End-to-End Delay

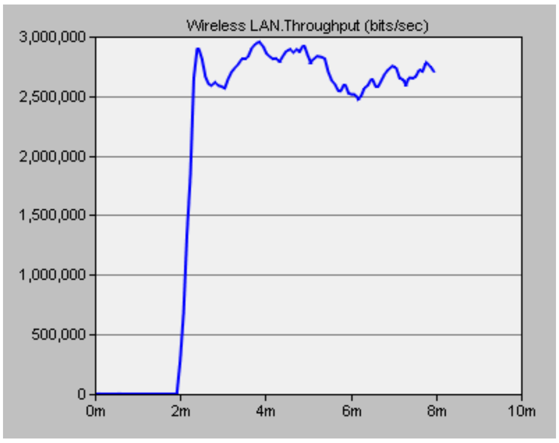
Wired nodes=20, Wireless Nodes = 6, 802.11g



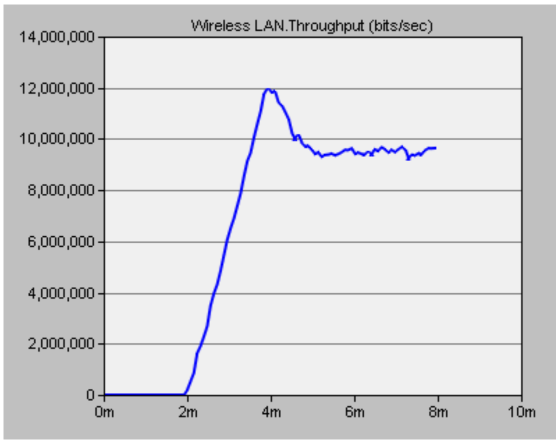
(g) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 6, 802.11b



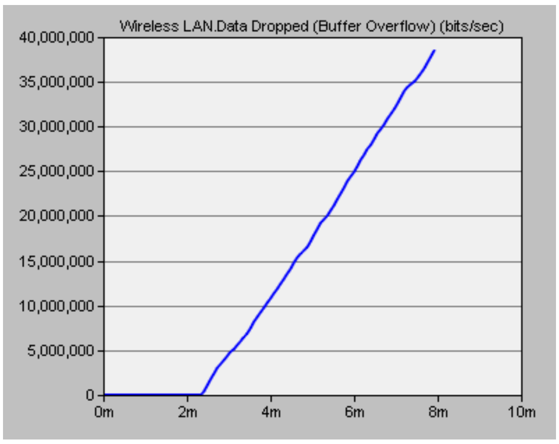
(h) Wireless LAN Delay
Wired nodes=20, Wireless Nodes = 6, 802.11g



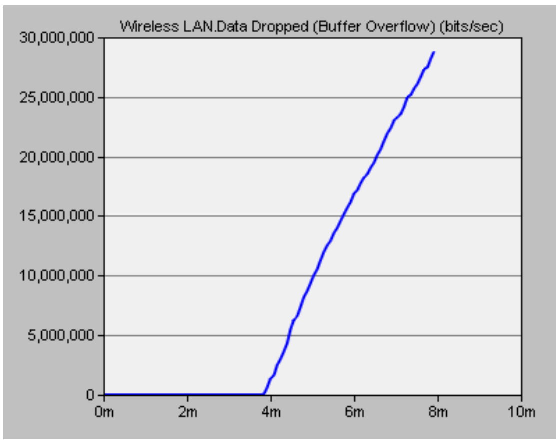
(i) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 6, 802.11b



(j) Wireless LAN Throughput
Wired nodes=20, Wireless Nodes = 6, 802.11g



(k) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 6, 802.11b



(l) Wireless LAN Data Dropped
Wired nodes=20, Wireless Nodes = 6, 802.11g

Figure 6.6 Wireless LAN Performance (Wireless Nodes = 6)

Table 6.1 Summary of experimental results (wireless nodes)

Number of Wireless Nodes	Wireless Protocol	Voice Jitter	Packet End-to-End Delay	Wireless LAN Delay	Ethernet Delay	Voice Traffic Sent (pps)	Voice Traffic Received (pps)	Packet loss ratio
2	802.11b	0.006ms	151ms	156 ms	0.69ms	9879	9098	8%
	802.11g	0.001ms	142ms	9.6ms	0.64ms	10039	9990	0.05%
4	802.11b	0.019ms	166ms	315ms	0.65ms	10,980	8,670	21%
	802.11g	0.004ms	146ms	30ms	0.62ms	10,918	10,110	7.4%
6	802.11b	0.045ms	182ms	468ms	0.66ms	11,791	7,969	32%
	802.11g	0.009ms	153ms	74ms	0.61ms	11,904	9,635	19%

In the 802.11g mode, the maximum data rate is 54 Mb/s; this rate is much larger than the 11 Mb/s of 802.11b. However, the simulation results show that 802.11g protocol gives us a much better performance than 802.11b but it seems that still cannot support VoIP services under this network model as the packet loss ratios are still too high. Because 802.11g has higher throughput than that in 802.11b can be achieved packets with large payload. However, there is only very small payload in VoIP. Therefore, although the 802.11g has data rate of 54 Mb/s, which is much higher than 802.11b, it still cannot gain that much improvement as far as VoIP capacity is concerned, because it cannot reduce the dominant overheads. The simulation results indicate that it can only support 2 wireless VoIP clients under 802.11g protocol with 20 wired VoIP clients in this network model. Therefore, as wireless networks provide limited bandwidth compared to a wired LAN, compressing codecs which can save more bandwidth are strongly recommended for wireless VoIP networks.

6.3 Scenarios 3: Impact of Encoder Schemes

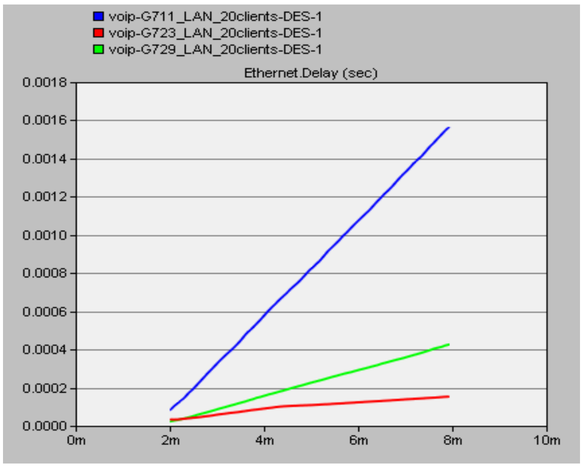
This study also carried out another scenario that measured the codec schemes. In this experiment, the codec schemes are changed from G.711 to G.723 (5.3k) and G.729 (8k).

In this scenario, the main findings are listed in table 6.2, the simulation results of this scenario are reported in figure 6.7, figure 6.8, and figure 6.9. Figure 6.7 (a) shows Ethernet delay of three codec schemes. Figure 6.7 (b) shows voice jitter of three codec schemes. Figure 6.7 (c) shows voice packed end-to-end delay of three codec schemes.

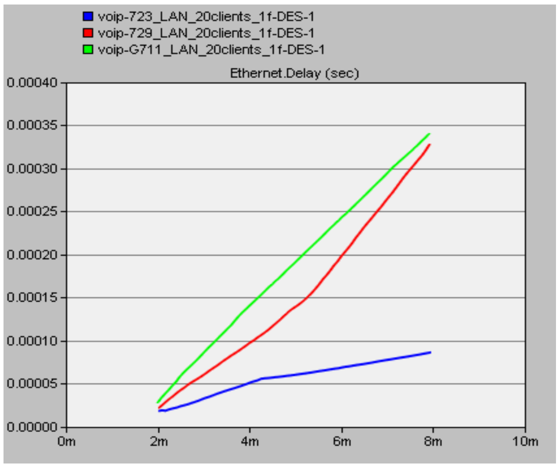
Figure 6.7 shows Ethernet delay of different codec schemes. The default setting of the attribute “voice frames per packet” is set to five. As can be seen in 6.7 (a), G.711 codec scheme yields the highest Ethernet delay among the three codecs e, the maximum value of the Ethernet delay is 1.57ms. G.723 codec scheme gives the lowest Ethernet delay among the three codec schemes (0.15ms).The Ethernet delay of G.729 is slightly higher than G.723 (0.43ms).

Table 6.2 Summary of experimental results (encoder schemes)

Voice Codec Schemes		Voice Jitter	Voice Packet End-to-End Delay	Ethernet Delay	Voice Traffic Sent (packets/sec)	Voice Traffic Received (packets/sec)
Voice Frames per Packet = 5	**G.711**	0.0000ms	141.6ms	1.57ms	24,120	24,120
	G.723 (5.3k)	0.0000ms	340.1ms	0.15ms	8,013.3	8,013.3
	G.729 (8k)	0.0000ms	140.4ms	0.43ms	24,400	24,400
Voice Frames per Packet = 1	**G.711**	0.0000ms	60.3ms	0.34ms	121,502	121,500
	G.723 (5.3k)	0.0000ms	100.1ms	0.09ms	40,367	40,367
	G.729 (8k)	0.0000ms	120.3ms	0.33ms	30,325	30,325



(a) Ethernet Delay
(Voice Frames per Packet = 5)

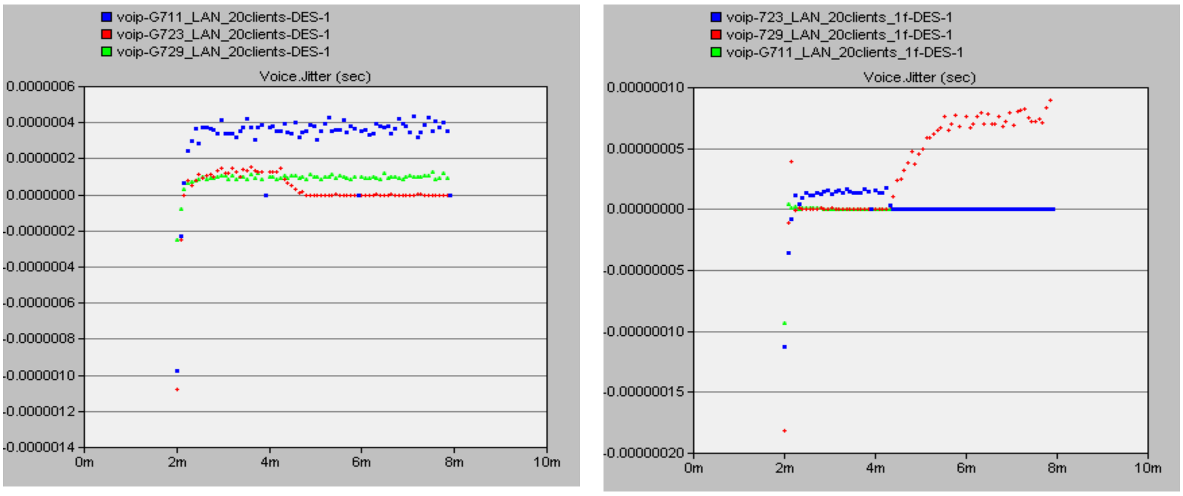


(b) Ethernet Delay
(Voice Frames per Packet = 1)

Figure 6.7 Ethernet Delay of Different Codec Schemes

As shown in figure 6.7 (b), the Ethernet delay decreases as the “voice frames per packet” is set to one. G.711 codec scheme still yields the highest Ethernet delay among the codec schemes, the maximum value of the Ethernet delay is 0.34ms, and G.723 codec scheme yields the lowest Ethernet delay among the three codec schemes (0.09ms). The Ethernet delay curve of G.729 is still in the middle of two other schenmes, but this time the Ethernet delay of G.729 codec scheme is only slightly lower than G.711 (0.33ms).

The codec schemes’ comparison values of voice jitter are shown in figure 6.8. As seen in 6.8 (a),similar to the Ethernet delays, when the parameter of the attribute “voice frames per packet” is set to five, G.711 voice scheme gives the highest voice jitter and G.723 voice scheme has the lowest voice jitter curve. All these codec schemes yield acceptable voice jitters which are less than 1ms.



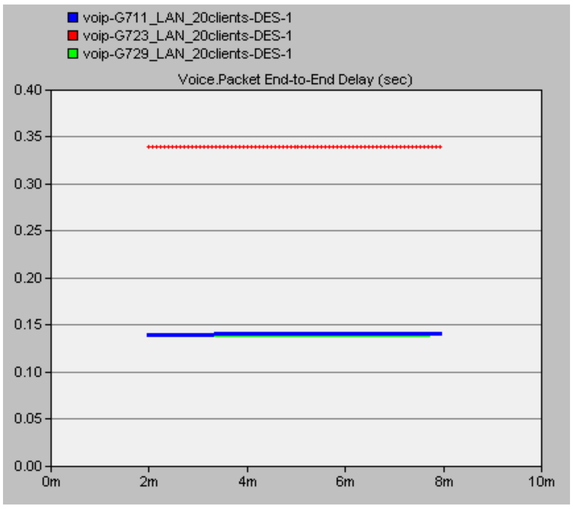
(a) Voice Jitter
(Voice Frames per Packet = 5)

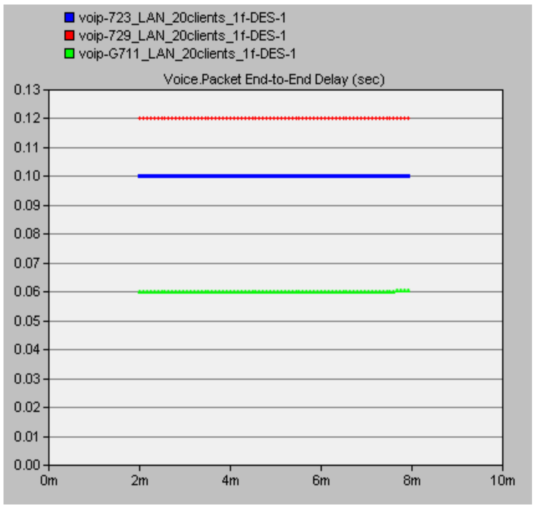
(b) Voice Jitter
(Voice Frames per Packet = 1)

Figure 6.8 Voice Jitter of Different Codec Schemes

Although the simulation results of the voice jitter are very close to 0ms. But as can be seen in figure 6.8 (b), not all the voice jitter curves decrease when the "voice frames per packet" is changed to one. Voice jitter of G.729 codec scheme becomes worse, and it yields the highest voice jitter curve among the codec schemes. However, the voice jitter curves of G.711 and G.712 decrease and very close to zero. All the jitter values are in the acceptable area.

Figure 6.9 shows the Voice packet End-to-End delay of three codec schemes. As can be seen in figure 6.9(a), when the parameter of the attribute "voice frames per packet" is set to five, G.729 codec scheme yields the highest Voice Packet End-to-End Delay around 340ms; it is more than 200ms which is considered as unacceptable value. G.711 and G.723 codec schemes have similar values both around 140ms which are less then the 200ms limit.





(a) Voice Packet End-to-End Delay
(Voice Frames per Packet = 5)

(b) Voice Packet End-to-End Delay
(Voice Frames per Packet = 1)

Figure 6.9 Voice Packet End-to-End Delay of Different Codec Schemes

However, when the parameter of the attribute "voice frames per packet" is set to one, all these codec schemes have good voice packet end-to-end delay values. G.729 codec scheme still yields the highest voice packet end-to-end delay, but it around 120ms which is an acceptable value. The voice packet end-to-end delay also inproved under G.711 and G.723 codec schemes. In this scenario, the simulation results indicate G.711 should be considered as the most appropriate voice scheme for this VoIP network model as it provides the best quality. Therefore, G.711 codec scheme is a good choice to the VoIP network model with 20 VoIP clients.

6.4 Scenarios 4: Impact of Traffic Arrival Distributions

In this scenario, the main findings are listed in table 6.2, the simulation results are reported in figure 6.8, which includes a number of figures. Figure 6.8 (a) shows Ethernet delay. Figure 6.8 (b) shows Voice Jitter. Figure 6.8 (c) shows Voice Packed End-to-End Delay.

This network model includes only wired component with 20 VoIP nodes. VoIP services performed smoothly among three different traffic distributions. The simulation results show that the VoIP has the best performance over Ethernet LAN under exponential traffic distribution.

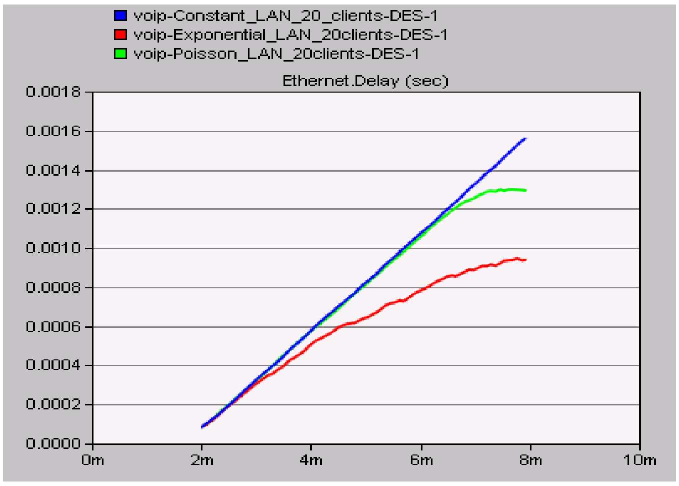
Table 6.3 Summary of experimental results (traffic arrival distributions)

(Wired nodes=20, Wireless nods = 0)

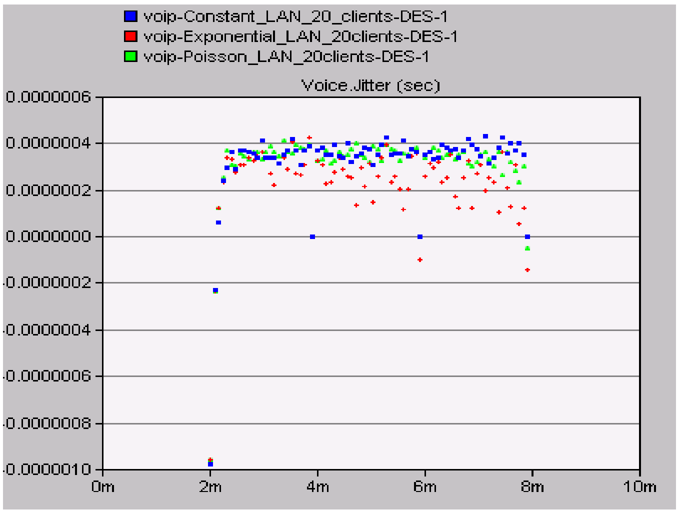
Traffic Distribution	Voice Jitter	Voice Packet End-to-End Delay	Ethernet Delay	Voice Traffic Sent (packets/sec)	Voice Traffic Received (packets/sec)
Exponential	0.0000ms	141.0ms	0. 9ms	13,936	13,936
Poisson	0.0000ms	141.3ms	1.30ms	20,295	20,295
Constant	0.0000ms	141.6ms	1.57ms	24,120	24,120

As seen in figure 6.10(a), the Ethernet LAN delay has the smallest value (0.09ms) under exponential arrival distribution, this is very low delay compares to the values under Poisson traffic distribution (1.3ms) and constant traffic distribution (1.57ms). Therefore, Ethernet delay could be ignored as all these values are around 1ms.

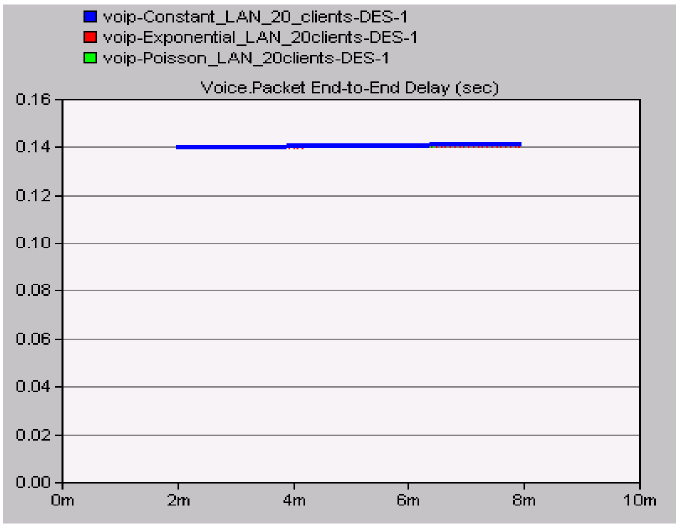
Figure 6.10 (b) shows the comparison graph for the three traffic distributions. Jitter values of all three traffic distributions are acceptable values close to zero that could be ignored. However, in general, the exponential traffic distribution gives the best jitter pattern over other traffic distributions. The red spots in 6.10 (b) are overally lower than blue and green spots.



(a) Ethernet Delay (Wired Nodes = 20)



(b) Voice Jitter (Wired Nodes = 20)



(c) Packet End-to-End Delay (Wired Nodes = 20)

Figure 6.10 Performance of the VoIP Traffic Distributions

Figure 6.10 (c) show Packet End-to-End Delay for the three different traffic distributions. As can be seen in (C), all the three traffic distributions yield almost the same values. However, comparing to the maximum values shown in table 6.3, the exponential distribution has the lowest voice packet End-to-End delay (141.0ms), this is only slightly lower than Poisson traffic distribution (141.3ms) and constant traffic distribution (141.6ms).

The number of packet sent form exponential traffic distribution (13,936) is the smallest comparing to Poisson distribution (20,295 packets) and constant distribution (24,120 packets), and the default constant traffic distribution sent the most voice packets during the simulation. Therefore, it is very clear to explain why VoIP services can be performed best under exponential traffic distribution. However, considering the voice packets sent from the constant traffic distribution are nearly twice as from the exponential traffic distribution, but the results under each distribution are very similar, a fast conclusion is that the traffic arrival distributions have little impact on the VoIP performance.

6.5 Simulation accuracy and Validation

The OPNET simulation was executed on a computer in Auckland University of Technology. The computer has Windows XP service pack 2 operating system with an Intel Core 2 CPU 6420 2.13GHz processor and 1.99 Gbyte of memory. The elapsed time for the simulation run was up to 3 hours. Due to the time and resource limitation, the simulation time for the scenarios (more than 100 VoIP clients) decreased to 6 minutes instead of 8 minutes.

In order to increase the accuracy of the simulation results, this dissertation referred to the previous detailed VoIP deployment study in [15], five simulation replications were run by different initial seeds with OPNET, and any integer value could be an initial seed. Five simulation replications were sufficient to mitigate the randomness of system [32], [33], and each simulation replication produced very similar graphical results.

6.6 Limitations of This Study

There are still some limitations for this study. Firstly, this dissertation only considered VoIP services and ruled out all the other devices such as file transferring services, email services and database services etc. However, these services are common in real world network. Secondly, although simulation performed quite well under OPNET terminology, although OPNET simulator is very close to the reality, however, it is still the simulation not the reality. Moreover, the OPNET only supports SIP protocol for VoIP connection, which means the OPNET does not support the models for the recent VoIP gateway and gatekeeper, so their performance are not measurable.

Chapter 7

Conclusion and Future Work

This study evaluated the performance of VoIP over Ethernet LANs through four various network scenarios using OPNET simulation tool. This study also measured different real time communication parameters, such as packet end-to-end delay, jitter, and tried to determine the maximum number of VoIP calls which the network can support. This study presented statistical and graphical analysis to enable us find out comparison patterns. This is very important and useful way as the comparison pattern can tell us what is happening in VoIP conversation for each network scenario.

The number of VoIP clients has significant impact on VoIP performance for both wired and wireless LAN. Especially for wireless LAN, the impact of increasing the number of wireless nodes will be more than impact of the increasing the number of wired nodes. The simulation results presented in this study show that the main bottle necks are the switches and routers (includes wireless routers), thus upgrading switches and routers and design a good VoIP network become very important.

In today's world, VoIP services have deployed on both small network and large-scale network, many organizations face the situation of increasing trend of networking services; network expanding through the existing network often includes both wired and wireless network components. In addition, providing VoIP services with other data services over the same network infrastructure is just one of the needs. Some previous studies show that the performance of VoIP services over wireless networks is much worse than over wired network. The simulation results of this study clearly shows VoIP service has different impact over wired and wireless network when the number of VoIP nodes is increasing.

The performance IEEE 802.11b is clearly worse than IEEE 802.11g for the same load conditions. However, we conclude that both IEEE 802.11b and IEEE 802.11g protocols bring significant delay and jitter that cannot meet all voice requirements for the network model in this dissertation. The impact also comes from the choice of the voice encoder schemes and type of traffic arrival distributions. The simulation results of this dissertation show that the VoIP services perform best under G.711 voice encoder scheme and exponential traffic arrival distribution.

This dissertation presented the VoIP network simulations. It takes a lot of time and effort to get acquainted with OPNET Modeler. This study referred to many relevant earlier studies and works to overcome problems and difficulties. Moreover, it provides a lot of insight into the VoIP performance over Ethernet LANs by using the OPNET tool. The results of the simulation are quite satisfactory.

The major factors that affect VoIP quality such as delay, jitter and packet loss, are measured by simulation. The simulation results presented in this dissertation can help organizations understand how well VoIP will perform on a local network prior to adopt VoIP, it also help researchers and designers to design a network for VoIP deployment. Various issues related to the deployment of VoIP are also discussed. These issues include VoIP security and traffic characteristics and QoS requirements.

This study only considered peer-to-peer voice calls. VoIP conferencing and messaging options are suggested as future research. This study considered VoIP traffic only. In future studies, more realistic traffic applications such as background traffic, FTP, and Email can be considered.

References

[1]. Skype official website: http://about.skype.com/

[2]. Windows Live Messenger URL:http://get.live.com/messenger/overview

[3]. Google Talk URL: http://www.google.com/talk/

[4]. K. Bhumip (2003) Implementing voice over IP. *John Wiley & Sons, Inc.*

[5]. V. Theoharakis, & D. N. Serpanos (2002). Editors, Enterprise Networking: Multilayer Switching and Applications. *Idea Group Publishing*, Hershey, PA, USA

[6]. ITU-R Rec. H.323 (1999). Packet-Based Multimedia Communications Systems.

[7]. T. Nguyen, F. Yegenoglu, A. Sciuto, & R. Subbarayan (2001). Voice over IP Service and Performance in Satellite Networks. *IEEE Communications Magazine*, Volume: 39, Issue 3, page(s): 164-171.

[8]. S. K. Das, E. Lee, K. Basu, & S. K. Sen (2003). Performance Optimization of VoIP Calls over Wireless Links Using H.323 Protocol Computers. *IEEE Transactions*, Vol. 52, No. 6 Page(s):742 – 752

[9]. G. A. Thom (1996). H.323: the multimedia communications standard for local area networks. *Communications Magazine, IEEE*, Volume 34, Issue 12, page(s): 52-56

[10]. L. Hong, & P. Mouchtaris (2000). Voice over IP signaling: H.323 and beyond. *Communications Magazine, IEEE*, Volume 38, Issue 10, Page(s):142 – 148.

[11]. L. Milandinovic, & J. Stadler (2002). Multiparty Conference Signaling using SIP. International Network Conference, 2002

[12]. J. Rosenberg, H. Schulzrinne, G. Camarillo, A.Johnston, J. Peterson, R. Sparks, M. Handley, & E. Schooler (2002). SIP: Session Initiation Protocol. RFC 3261, *IETF*.

[13]. Han-Chieh Chao, Y. M. Chu, & G. Tsuei (2001). Codec Schemes Selection for Wireless Voice over IP (VoIP). *Proceedings of the Second IEEE Pacific*

Rim Conference on Multimedia: Advances in Multimedia Information Processing. Pages: 622 - 629

[14]. S. Zeadally, F. Siddiqui, & P. Kubher (2004).Voice over IP in Intranet and Internet environments. *Communications, IEE Proceedings*, Volume 151, Issue 3, Page(s): 263 - 269

[15]. K. Salah, & A. Alkhoraidly (2006). An OPNET-based simulation approach for deploying VoIP. *International Journal of Network Management*, Volume 16, Issue 3, Pages: 159 - 183

[16]. W. Chou, (2007). Strategies to Keep Your VoIP Network Secure. *IT Professional Published by IEEE Computer Society*, Volume 9, Issue 5, Pages 42-46.

[17]. T.J. Walsh, & D.R. Kuhn, (2005). Challenges in securing voice over IP. *IEEE Security & Privacy*, vol. 3, no. 3, pp. 44- 49.

[18]. L. Huijie, & L. Xiaokang, (2005) An OPNET-based 3-tier network simulation architecture. *IEEE International Symposium*, Volume: 2, page(s): 793- 796

[19] J. A. Zubairi, & M. Zuber. SUNY FREDONIA Campus Network Simulation and Performance Analysis Using Opnet. *University of New York College*. URL :http://www.cs.fredonia.edu/zubairi/opnet/op2k_jz.pdf

[20]. A. Van de Capelle, E. Van Lil, J. Theunis, J. Potemans , & M. Teughels (2001). Project driven graduate network education, Networking. I*CN 2001, PTI, Proceedings Lecture Notes in Computer Science*, pp. 790-802.

[21]. M. Baratvand, M. Tabandeh, A. Behboodi, & A.F.Ahmadi (2008). Jitter-Buffer Management for VoIP over Wireless LAN in a Limited Resource Device, Sharif Univ. of Technol., Tehran;, *Networking and Services*, 2008.Fourth International Conference on page(s): 90-95.

[22]. L. Cai, Y. Xiao, X. Shen, & J. W. Mark. (2006) VoIP over WLAN: Voice capacity, admission control, QoS, and MAC. *International Journal of Communication Systems*, pp. 491- 508.

[23]. S. B. A. Latif, M. A. Rashid, & F. Alam, (2007) Profiling Delay and Throughput Characteristics of Interactive Multimedia Traffic over WLANs Using OPNET. *Proceedings of the 21st International Conference on Advanced Information Networking and Applications Workshops*, Volume 02, Pages: 929-933.

[24]. Opnet(TM) Home page, Web site: http://www.opnet.com

[25]. W. An, C. Yuqiang, & W. Jinhua (2006). Simulation Study of TCP/IP Communication Based on Networked Control Systems. *Intelligent Control and Automation*, *WCICA 2006*, The Sixth World Congress, Volume: 1, page(s): 4479-4483.

[26]. http://www.wainhouse.com/files/papers/wr-qos-in-ip-networks.pdf

[27]. A. P. Markopoulou, F. A. Tobagi, & J. Karam (2003). Assessing the Quality of Voice Communications Over Internet Backbones. *IEEE/ACM Transactions On Networking (TON)*, VOL 11, Issue 5, Pages: 747 – 760.

[28]. A.Takahashi, H.Yoshino, & N. Kitawaki (2004). Perceptual QoS Assessment Technologies for VoIP. *IEEE Communications Magazine*, Volume 42, Issue 7, page(s): 28-34

[29]. C. N. Chuah, (2000) Providing End-to-End QoS for IP based Latency sensitive Applications". *Technical Report, Dept. of Electrical Engineering and Computer Science, University of California at Berkeley*.

[30] D. Kotz, & K. Essien (2005). Analysis of a campus-wide wireless network. *In Proc. of the 8th Annual International Conference on Mobile Computing and Networking, Atlanta, GE*, pages 107–118.

[31] L. Zheng, L. Zhang, & D. Xu (2001). Characteristics of network delay and delay jitter and its effect on voice over IP (VoIP). *ICC 2001. IEEE International Conference,* Volume 1, Page(s):122 – 126.

[32] K. Pawlikowski, H. Jeong, & J. Lee (2002). On credibility of simulation studies of telecommunication networks. *Communications Magazine, IEEE*, Volume 40, Issue 1, Page(s): 132-139.

[33] A. Law, & W. Kelton (1991). Simulation modeling and analysis, 3nd edition. *New York: McGraw-Hill*, Pages: 784.

[34] (2000).One Way Transmission Time. *ITU-T Recommendation G.114*.

[35] J. H. James, C. Bing, & L. Garrison, (2004) Implementing VoIP: a voice transmission performance progress report. *Communications Magazine, IEEE*, Volume 42, Issue 7, page(s): 36- 41

[36] D. Butcher, L. Xiangyang, & G. Jinhua, (2007) Security Challenge and Defense in VoIP Infrastructures. *Systems, Man, and Cybernetics, Part C: Applications and Reviews, IEEE Transactions,* Volume 37, Issue 6, Nov. 2007 Page(s):1152 – 1162.

[37] P.C.K. Hung, & M.V. Martin, (2006) Security Issues in VOIP Applications.

Electrical and Computer Engineering, CCECE '06, Page(s):2361 – 2364

[38] W. Wei, C. L. Soung, & V.O.K. Li, (2005) Solutions to performance problems in VoIP over a 802.11 wireless LAN. *Vehicular Technology, IEEE Transactions on*, Volume 54, Issue 1. Page(s): 366 – 384.

[39] T. Alex, C. CElizabeth, & Tharam Dillon (2007) Secure Mobile VoIP. *DEBI Institute, Curtin University Australia, International Conference on Convergence Information Technology.*

[40]. D. Rizzetto, & C. Catania (1999). A Voice over IP Service Architecture for Integrated Communications. *IEEE Internet Computing*, Volume 3, Issue 3, Pages: 53 – 62.

[41]. B. Goode (2002) Voice Over Internet Protocol (VOIP). *Proceedings of three IEEE*, Volume 90, Issue 9, Page(s): 1495 – 1517