Performance Studies of VoIP over Ethernet LANs

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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 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 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. We preformed our simulation in OPNET simulation tool to validate our experiments.

*Keywords**:** VoIP – Ethernet LANs,SOHO*

I. 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], [4].

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 [5] as Ethernet based LANs is very common in enterprises and other organizations for data networking [6]. 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 research paper.

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 are shared 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

2. Related Work

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 paper, there is one independent simulation scenario tests G.711, G.723, and G.729 encoder schemes to investigate the performance differences.

2.1 Network Performance Studies

In reference [8], 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.

Table 1. Leading Researchers and Their Contributions in VoIP Performance Study

Researcher	Contribution	Year	Description/key concept
D. Kotz & K. Essien	This study can help understand usage patterns in WLAN	2005	This study investigated nearly two thousand users from Large WLAN
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
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
Salah & 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.
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

2.2 Network Modelling

This section will introduce simulation environments; the configurations of specific devices and related technologies are required to support VoIP will also be presented. This paper describes VoIP network simulation scenarios.

2.3 Strengths and Weaknesses of OPNET

The simulation tool adopted in this paper 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 are 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.

4. 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

4.1 Modelling Assumptions:

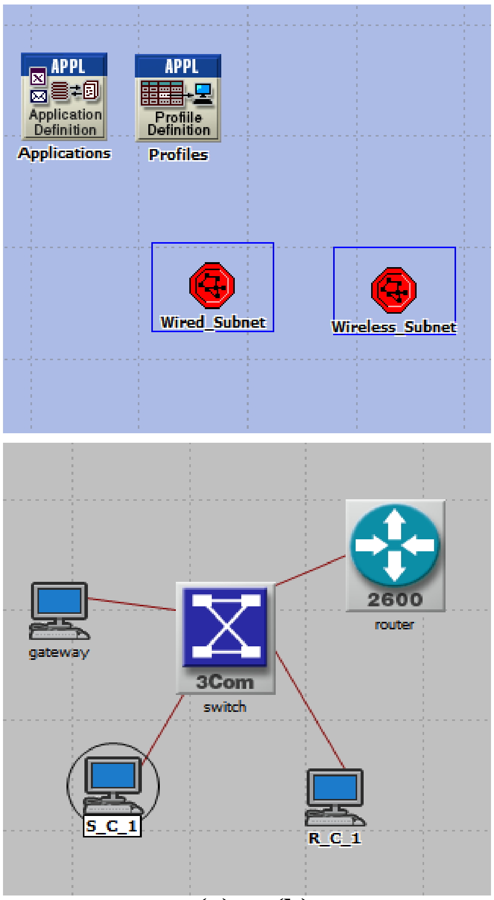
- The local area networks operate at 100Mb/s throughout the simulations.
- There is no other network traffic besides VoIP traffic in this study. Each simulation experiment considers 8 minutes of simulation time.
- This study also assumes that there are only peer-to-peer voice calls throughout the simulation,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

Various scenarios were simulated in this paper. 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:

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 1 shows the OPNET representation of network topology for Scenario 1.



(a) (b)

Figure 1. OPNET Representation of VoIP Network model

5. Results and Analysis

A detailed simulation network modelling was 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 paper presents simulation results for performance prediction of VoIP.

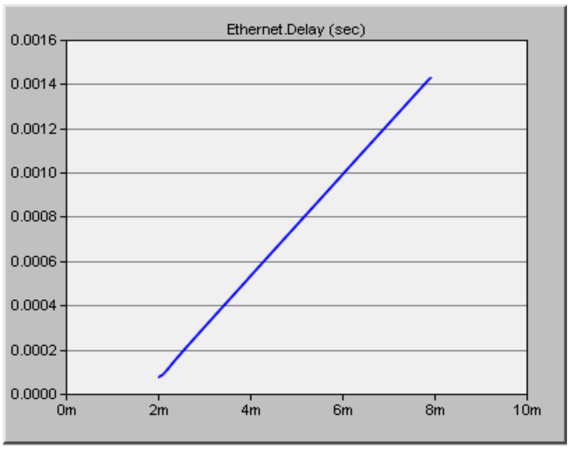
The duration of OPNET simulation was set to 8 minutes (duration time for campus network model was set to 4 minutes due to memory limitation). The VoIP traffic started at 120 seconds after the simulation is initially started. Every simulation stops at 8 minutes, the statistical and graphical results are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

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 2 shows network packet delay. The OPNET default reported delay configuration is the sample mean. Figure 2 (a) shows Ethernet delay is less than 1ms when there is only two VoIP clients on a wired Ethernet LAN. As seen in Figures 2 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 7ms Ethernet delay. The Ethernet delay increases to around 9 ms for 120 nodes (Figure 2 (i)).

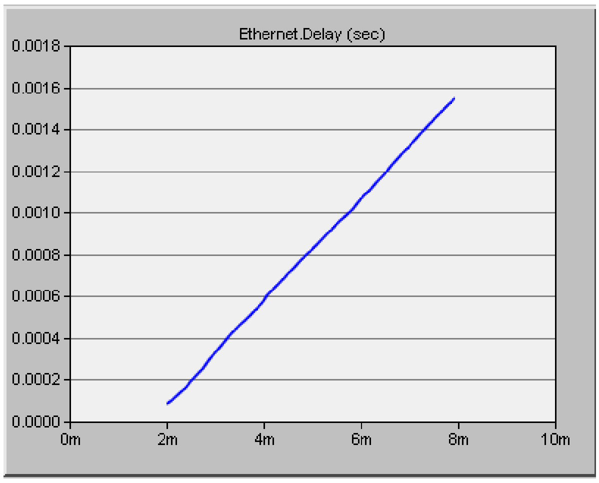
However, as seen in Figure 2 (j) and (k), Ethernet packet delay rapidly increases to more than 1 second when the number of VoIP clients increased to 200 nodes and 400 nodes.



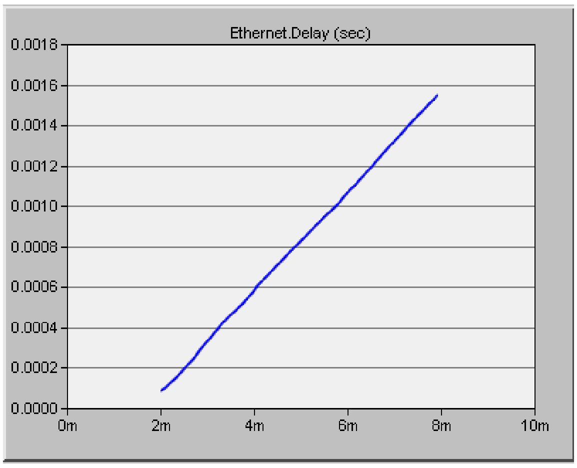
(a) (N=2)

(b) (N=4)

(c) (N=8)



(d) (N=10)



(e) (N=16)

(f) (N=20)

Figure 2. Network Packet Delay

Figure 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.

6. Conclusion and Future Work

This paper 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 paper 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.

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