ANALYZING VOIP WITH VIDEO CONFERENCING IN IPV6 PLATFORM

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UNIVERSITI TEKNIKAL MALAYSIA MELAKA

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This report is submitted in partial fulfillment of the requirement for the
Bachelor of Computer Science (Computer Networking)

FACULTY OF INFORMATION AND COMMUNICATION TECHNOLOGY
UNIVERSITI TEKNIKAL MALAYSIA MELAKA
2010

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DECLARATION

I hereby declare that this project report entitled

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is written by me and is my own effort and that no part has been plagiarized without citations.

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DEDICATION

This thesis is dedicated to my family especially to my beloved parents, Chan Kok Weng and Yap Fee Khoon, who always give me encouragement all the way since the beginning of my studies.

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ABSTRACT

This project is mainly about the comparison performance between IPv6 and IPv4 network in VoIP in the presence of varying levels of background UDP traffic and it is successfully be tested in different operating system which are Windows XP and Linux Ubuntu 10. The packet that will focus on this project is only the video packet. This project will analyze the VoIP by make 3 calls for each traffic in every bandwidth. The durations of each calls is 5 minutes. The performance measures are maximum and mean delta (time between the arrival of video packets), maximum and mean jitter and packet loss. The traffic of this project is using the virtualized traffic, means that the traffic will be generate by using traffic generator software.

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ABSTRAK

Projek ini terutamanya adalah mengenai prestasi perbandingan antara IPv6 dan IPv4 rangkaian dalam VoIP di atas pelbagai peringkat lalu lintas UDP dan telah berjaya diuji dalam sistem pengoperasian yang berbeza, Windows XP dan Linux Ubuntu 10. Paket yang akan fokus pada projek ini adalah hanya paket video. Projek ini akan menganalisakan VoIP dengan membuat 3 panggilan untuk setiap lalu lintas di setiap bandwidth. Masa yang digunakan dalam setiap panggilan adalah selama 5 minit. Jenis-jenis prestasi yang diuji dalan projek ini adalah maksimum dan min *delta* (waktu antara penghantaran paket video), maksimum dan min *jitter* dan *packet loss*. Lalu lintas dalam projek ini adalah dengan menggunakan virtual lalu lintas, ini bermakna lalu lintas yang akan dihasilkan dengan menggunakan perisian *Traffic Generator*.

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LIST OF ABBREVIATIONS

VoIP	-	Voice over Internet Protocol
IPv4	-	Internet Protocol version 4
IPv6	-	Internet Protocol version 6
VTC	-	Video Teleconferencing
MCU	-	Multipoint Control Unit
UDP	-	User Datagram Protocol
LAN	-	Local Area Network
WAN	-	Wide Area Network
WLAN	-	Wireless Local Area Network
VLAN	-	Virtual Local Area Network
MIPv6	-	Mobile Internet Protocol version 6
MOS	-	Mean Opinion Score

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CHAPTER I

INTRODUCTION

1.1 Project Background

VoIP (Voice over Internet Protocol) is a general term for a family of transmission technologies for delivery of voice communications over Internet Protocol networks while videoconference is a set of interactive telecommunication technologies which allow two or more locations to interact via two-way video and audio transmissions simultaneously. Nowadays, there are many types of VoIP in the market such as Windows Live Messenger, SKYPE, ICQ and so on. Most of these VoIP are integrated with video call function and run in IPv4.

IPv6 is the next generation network layer protocol, which was at first called IPng, future internet next generation that was designed as a replacement for the current IPv4

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protocol due to the limitations and shortcomings of IPv4 and more importantly, the exhaustion of IPv4 addresses.

The exhaustion of address space is one of the issues as the momentum of VoIP grows significantly. The 32-bit address space in IPv4 precludes its scalability to a large user base. IPv6 addresses this problem of IPv4 with a very large address space that consists of 128 bits. Therefore, it is now possible to support 2^{128} unique IP addresses, a substantial increase in number of computer that can be addressed with the help of IPv6 addressing scheme.

Also, in order for VoIP to be widely deployed, security concerns such as eavesdropping and hacking must be addressed as well. The other issues that impact VoIP are:

- End-to-end integrity of signaling and bearer paths details.
- IP voice packet delivery across firewall
- NAT (network address translation) addressing issues that cause several networking problem in end-to-end nature of the internet.
- Preventing denial or disruption of service.

As the growing popularity of VoIP will make it a significant component of traffic in the future internet, it is one of the interests to compare the VoIP performance. The result would help to determine if there are any differences in VoIP performance over IPv6 due to overhead resulting from the larger IPv6 header. Performance is measured using maximum and mean values of delta, maximum and mean jitter and packet loss.

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1.2 Problem Statements

The problem statements in this project are:

- Performances of VoIP in IPv4 is not as stable as in IPv6 environment
- Every platform of operating system may affect the performance of VoIP in IPv4 and IPv6 environment

1.3 Objective

There are few objectives that will be achieved from this project:

- Analyze the performance of VoIP in IPv4 and IPv6 environment in term of value of delta with different operating system.
- Analyze the performance of VoIP in IPv4 and IPv6 environment in term of value of jitter and packet loss with different operating system.
- Compare the performance of VoIP and show the suitable recommendation for different operating system platform.

1.4 Scope

The scopes of this project are:

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- This project will only cover VoIP with video conferencing in IPv6 and comparison with IPv4
- This performance will be test in Windows XP and also Linux Ubuntu 10.
- The target users are for network technicians that are in charge of networks in companies, offices, schools and organizations.

1.5 Project Significance

There are several benefit and significance of this proposed project. The results of the performance will show the comparison between IPv6 and IPv4 in VoIP network. Through the comparison, we can know the different of performance in both technologies. Besides that, the result also will show the limitation of VoIP in IPv6. By doing this research, we also can find the different performance of VoIP when run on different Operating System. Therefore, VoIP developer can improve the limitation of it in the future.

1.6 Expected Output

At the end of the analysis, we will get the result of the performance from VoIP which run in IPv6 and IPv4 platform. Based on the result, the improvement of current VoIP development can be enhanced for future.

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1.7 Conclusion

As we know, IPv4 addresses will exhausted soon and IPv6 will slowly replace it in our future life. Therefore, analysis of VoIP in IPv6 is a must to get the result of limitation, so that the developer can easily improve the VoIP due to the result given.

From this chapter, the problem statements, objectives, scope, project significance and expected output are being identified in order to analyze the VoIP with video conferencing in IPv6 platform.

After finishing this chapter, it will proceed to the second chapter which is the Literature Review and Project Methodology. This chapter will discuss and analyze the existing systems.

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