

# **Faculty of Information and Communication Technology**

## ANALYSIS EFFECTIVENESS VOIP OVER MPLS-IPVPN FOR STATE GOVERNMENT NETWORK

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## MASTER OF COMPUTER SCIENCE (INTERNETWORKING TECHNOLOGY)

2017

C Universiti Teknikal Malaysia Melaka

### ANALYSIS EFFECTIVENESS VOIP OVER MPLS-IPVPN FOR STATE GOVERNMENT NETWORK

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A dissertation submitted in fulfillment of the requirements for the degree of Master Science Computer (Internetworking Technology)

Faculty of Information and Communication Technology

## UNIVERSITI TEKNIKAL MALAYSIA MELAKA

2017

C Universiti Teknikal Malaysia Melaka

## DECLARATION

I declare that this dissertation entitled "ANALYSIS EFFECTIVENESS VOIP OVER MPLS -IPVPN FOR STATE GOVERNMENT NETWORK" is the result of my own research except as cited in the references. The thesis has not been accepted for any degree and is not concurrently submitted in candidature of any other degree.

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## APPROVAL

I hereby declare that I have read this dissertation and in my opinion this dissertation is sufficient in term of scope and quality for the award of Master of Computer Science (Internetworking Technology).

Signature	:
Supervisor Na	me: Prof Madya Dr. Faizal Bin Abdullah
Date	:

#### DEDICATION

This thesis is dedicated first and foremost to my wife, without whose support and encouragement it would never have been completed. I also sincerely thank my parents for instilling in me the intellectual curiosity to pursue my education. I am grateful for the mentorship of my thesis committee. My advisor, Prof Madya Dr. Faizal Bin Abdullah and Dr. Othman Bin Mohd provided valuable feedback, guidance, and direction. They are all excellent role models and distinguished researchers. I greatly appreciate the opportunity to have studied under them. This study would not have been possible without the data collection and support provided by the 1Melaka\*Net Network IT Department Team Jabatan Ketua Menteri Melaka (JKMM). I greatly appreciate their taking the time and resources to support this study and ensure the IT Department JKMM community's privacy while enabling research. Additionally I would like to thank my classmates Mr. Nasran, Miss Aminah Abod and Mr. Rasyid, who worked with me on these class projects. Vince Bowman, a very talented undergraduate senior, developed a packet parsing tool for use in future work.

## ABSTRACT

The performance of the Voice over IP (VoIP) protocol is of interest when planning for public access. The technologies, which work well for limited use often, fail to scale-up to the user requirements. High-quality VoIP services are required as for the Internet communications to be an alternative towards Public Switched Telephone Network (PSTN). The deployment of VoIP in the Internet network does not promise a good Quality of Service (QoS), since Internet is a kind of best-effort networks. The privacy consideration is also of importance when provisioning voice services on the Internet; particularly from the business use perspective. A full study, which comes up with a definitive set of recommendations would require considerable work over a substantial period of time, however some information on the performance may be obtained by reenacting the most commonly occurring conditions in the lab to ascertain the sensitivity of the VoIP to its key QoS parameters. The aim of this project is to analyze effective VOIP technique and analyze the performance of VoIP communications by observing the QoS parameters variation with respect to the some of the pertinent communication aspects on State Government Network. The aspect chosen in this regard comprise the call signaling protocols, the networking environments, and VPN protocols. Some similar studies have also been used as a comparative measurement towards the results obtained from this research.

Keywords: VOIP, MPLS, IPVPN, performance, Quality of Service (QoS)

### ABSTRAK

Prestasi suara melalui IP (VoIP) protocol adalah kepentingan apabila merancang untuk akses awam. Teknologi yang bekerja dengan baik untuk kegunaan terhad kerap, gagal untuk skala-up kepada keperluan pengguna. Perkhidmatan VoIP yang berkualiti tinggi diperlukan kerana untuk komunikasi Internet menjadi alternative ke arah Public Switched Telephone Network (PSTN). Penempatan VoIP dalam rangkaian internet tidak menjanjikan kualiti yang baik perkhidmatan (OoS), kerana Internet adalah sejenis rangkaian usaha terbaik. Pertimbangan privasi juga penting semasa memperuntukkan perkhidmatan suara di Internet; terutamanya dari perspekti fkegunaan perniagaan. Satu kajian penuh, yang datang dengan satu set definitive cadangan akan memerlukan kerja yang besar dalam tempoh yang lama, namun beberapa maklumat mengenai prestasi boleh diperolehi dengan semula menggubal keadaan yang paling biasa berlaku di makmal untuk memastikan sensitiviti VoIP untuk parameter QoS utamanya. Tujuan projek ini adalah untuk menganalisis teknik VOIP berkesan dan menganalisis prestasi komunikasi VoIP dengan memerhatikan QoS parameter perubahan berkenaan dengan beberapa aspek komunikasi penting di rangkaian Kerajaan Negeri. Aspek dipilih dalam hal ini terdiri dari pada panggilan isyarat protokol, persekitaran rangkaian, dan protocol VPN. Beberapa kajian yang sama juga telah digunakan sebagai ukuran perbandingan terhadap keputusan yang diperolehi daripada kajian ini.

Kata kunci: VOIP, MPLS, IPVPN, prestasi, Kualiti Perkhidmatan (QoS)

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### ACKNOWLEDGEMENT

Alhamdulillah. Thanks to Allah SWT, whom with his willing giving me the opportunity to complete my Master Project entitle To Analysis Effectiveness VOIP Over MPLS -IPVPN For State Government Network. I would like to express my sincere gratitude to my supervisor, Prof Madya Dr. Mohd Faizal Bin Abdullah, for the continuous support to complete my project, for his assist, sound advice, patient and immense knowledge. Hits guidance helped me in all the time of final year project and writing of this report. Deepest thanks and appreciation to my beloved wife, mother and siblings for giving me support and motivation through my study years. Last but not least, my thanks goes to my friends for the stimulating discussion, for the sleepless nights we were working together before the deadlines and for all the fun we had together.

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### **CHAPTER 1**

#### **INTRODUCTION**

This research project is part of the qualification of Master Computer Science in Internetworking Technology Courses. The research project focuses on the study the effective technique of Voice Over IP (VoIP) in Multiprotocol Label Switching (MPLS) on Internet protocol (IP) virtual private networks (IPVPN). This study assesses the performance and quality of VoIP service in the MPLS IPVPN network. The experimental results on which technique can maintain a good performance on VoIP applications in an MPLS Network Services managed low end-to-end delay, low jitter, low packet loss, regardless of traffic conditions.

#### 1.1 Background of Study

Voice over Internet Protocol (VoIP) is a suite of combined technology, it enables voice communication in the medium of the Internet Protocol (IP). VoIP can provide the result of the involvement of many sophisticated use of the Internet and online services is offered on the same network. VoIP technology running at the Public Switched Telephone Network (PSTN) and provides low service fees for Internet infrastructure advantages of this area. IP networks based on packet switching method allow more users to share network resources from non-PSTN.

VoIP is a suite of technology term for transmission technologies; provide voice communication in an IP network medium such as the internet (Fjellskal et. all, 2012). The basic step in the Internet phone call is the conversion of voice signals into digital format that outputs

the translation of the signal into Internet Protocol (IP) packets for transmission over the Internet. The process is reversed at the receiving end.

The Telecommunications Industry Association (TIA) 2005 states that residential VoIP consumers are more than three times in 2005 and predicted growth of more than 40% during 2009. TIA also predicted this will make more than 18 million VoIP connections. This fact shows that VoIP is being not only increasing speeds, but also it is here to stay. The adoption of VoIP in small to large businesses has also been great. Traditional communication systems are being replaced at a rapid pace by enterprise business communication tools that offer feature-rich and cheaper way of communicating with your contacts.

Recently, VoIP technologies have advanced to provide tremendous opportunities for service providers, as one can use a single IP network for both data and voice communication in cost-effective and reliable manners. Service providers are now adopting VoIP technologies, to provide new services and applications to accommodate their customer's needs. One major VoIP infrastructure deployment issue for service providers is to maintain high quality of communication services to the customers.

Multi-Protocol Label Switching (MPLS) can be reflected as a good packet switching technology that ensure the Quality of Service (QoS), useful for multimedia applications, next generation communication service reliability and efficient use of network resources Nisha Chauhan et. all, (2015).

Fast Virtual Private Networks (VPN) will run on public network infrastructure as the backbone WAN supplement instead of using expensive leased or dial-up connection in a private network environment. The question arises, is VPN a good solution for wide range combination of PSTN networks, Internet Service Providers (ISPs), IP, Asynchronous Transfer Mode (ATM) and Frame Relay networks. According to the subscribers end, communication through a private or public network should be different in performance (QoS and Security) from the communication (post, fax or sensitive documents) via the PSTN in an organization. In the first case of communication through private or public network; the information is provided directly to the right destination, in safe and reliable manners. MPLS-based VPN is the best solution for all scales of companies currently deployed VPNs to public or private site-to-site communication. MPLS offers sophisticated communications networks with IP QoS that enable multiple classes of public or private services for businesses. In these organizations vital applications are treated with higher priority than other applications.

The reason for this postulation is to investigate the operation of VoIP over MPLS-IPVPN infrastructures for guaranteed Quality of Service (QoS) that is influenced by a number of important factors including delay, load, throughput, packet loss, bits error ratio, bit errors per packet and voice-encoding scheme. This complex interaction of these parameters defines the overall call quality experienced by the consumer. VoIP over MPLS-IPVPN research should define voice service types that are comparable to the existing PSTN services and could be provided at a lower cost.

#### **1.2 Problem Statement**

The main issues in VoIP are commonly like a combination two traditional PSTN. These issues are inherent to VoIP and less can be done to enhance this technology. However, the parameter such as delay, load, throughput, packet loss, bits error ratio, bit errors per packet can be enhanced by careful planning and solid network design. VoIP has the problem of delays and the quality of hearing that is not clear even to operate on a network infrastructure based on internet protocol.

### **1.3 Research Question**

- i) How the interior and exterior routing protocols VoIP network works with MPLS VPN?
- ii) What is the influencing parameter of VoIP network with MPLS VPN?
- iii) How the MPLS VPN based on Routing Information Protocol Version 2 (RIPv2) or
  Open Shortest Path First (OSPF) interior routing protocol and Border Gateway
  Protocol (BGP) exterior routing protocol with IP QoS will be the best solution for
  VoIP traffic about VPN delay, load and throughput, and Site-to-Site Flow delay
  and LSP delay, and End to-End Queuing delay?
- iv) Which of the proposed scenarios will be the best solution;
  - a. MPLS VPN with RIPv2 routing protocol or
  - b. MPLS VPN with OSPF routing protocol.

All of this scenario will use the same QoS parameters and service reliability in order to get the customer satisfaction and confidence.

### **1.4 Research Objective**

This thesis will focus on the implementation of Quality of Service (QoS) in MPLS VPN backbone with VoIP, using the bandwidth management tool and do the analysis behavior of VoIP traffic in the MPLS VPN network with QoS.

The following steps will be involved to answer the questions and to get the results.

- i. To study the protocol in MPLS-IPVPN Network Environment for support VOIP
- ii. To compare technique and network protocol in the organization on VOIP.
- iii. To identify proposed scenarios will be the best solution with regard to MPLSVPN with QoS parameters.

### 1.5 Research Scope

For this research, there are several research scopes of this study:

- This project topology will use the State Government Network design environment. This project will be conducted by using MPLS-IPVPN over Metro-E Network type/technology.
- ii. This study focuses only on a VoIP network traffic.
- iii. This project focuses consist of three (3) locations as listed below:
  - a. Two (2) Customer site branch
  - b. One (1) Internet Service Provider network cloud
- iv. The project will be conducted by using the simulation in environment network.

### **1.6** Research Significance

The nature of administrative prerequisites of VoIP will be recognized from the writing in light of the attributes of VoIP. MPLS-IPVPN systems will be made to bolster VoIP in the virtual situations. Levels of postponement, jitter, and misfortune are to be watched and answered to analyze VoIP conveyed by the two systems under different heaps of foundation activity and VoIP movement. Also, a dynamic lining administration (AQM) is to be designed in MPLS- IPVPN systems to examine its viability, in correlation with that in alternate MPLS-IPVPN systems arrange without AQM.

### 1.7 Project Report Overview

In order to perceive the flows of research works, there are eight chapters to be illustrated in visualizing the work under study. As below depicts Figure 1.1 is the sequence of research activities involve in this research.



Figure 1.1: Research Structural Process

**Chapter 1**: Introduction, this chapter provides justification with an overview and also background information of this research, problem statement, objective, scope, project significance and expected output.

**Chapter 2**: Literature Review, this chapter provides justification with an overview of VOIP. Also, it will explain details about Quality Of Services (QOS), Transport Protocol started with analyses the VoIP and quality of service issue in the Internet. It will followed by the discussion about previous researches that related to this topic.

**Chapter 3**: Research methodology, this chapter discusses the methodology which used to achieve the research objective, include research method, research methodology and model, and flow chart of this analysis. This chapter also stated the research tools, project requirement and project schedules and millstone.

**Chapter 4**: Implementation, it is part of collecting data in this research, where design, configuration of VOIP and capture the voice traffic will be done in this chapter.

**Chapter 5**: Findings and Analysis, it is the main part of this research, where the result of detection and analysis of the new characteristics will be discussed in depth in this chapter. This chapter focuses on testing process by tested and validate results from the analysis result in the previous phase.

**Chapter 6**: Conclusion and Future Work. This chapter will provide the summary of this research, contributions, limitation and future work.