COMPARATIVE STUDIES ON THE CODECS PERFORMANCE OF BEST CODECS TECHNIQUES FOR VOICE AND VIDEO IN INTERNET PROTOCOL VERSION 4 (IPV4) AND INTERNET PROTOCOL VERSION 6 (IPV6)

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(COMPUTER SCIENCE)

UNIVERSITI PERTAHANAN NASIONAL MALAYSIA

2024

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Thesis submitted to Center for Graduate Studies, Universiti Pertahanan Nasional Malaysia, in fulfillment of the requirements for the Degree of Master of Science (Computer Science)

2024

ABSTRACT

Voice over Internet Protocol (VoIP) has become universal in recent days and become the first choice of small to medium companies for voice and video communication over IP network to cut down the cost and use the IT resources in much more efficient way. Another popular technology that ruled the world after the year 2000 is 802.11 wireless networks. The wireless medium has special requirements than the 802.3 (Ethernet) and special consideration take into account while implementing the video and voice over IP (VoIP) via wireless medium network. One of significant issues 802.11 and 802.3 is the bandwidth availability. Implement of VoIP over 802.11 must consider availability of bandwidth to retain the good voice quality. IPv6 is considered to be the next-generation Internet protocol. We will use wireless network platform to conduct and achieve the objective of this research. Therefore, this study is to analyze voice and video performance and measure Quality of Service (QoS) delivered by IPv6 using the best codecs approach compared to IPv4. It focuses more on quality of voice on voice and video codecs such as packet loss, delay time and jitter in IPv6 through soft phone. X-lite application is used for communication between source and destination party. In our experiment, there are four types of audio codecs have been tested such as G.711 a law, Speex, BroadVoice-32, iLBC and two video codecs such as H.263 and H.263+. Network Management System (NMS) is used to monitor and capture the performance of voice and video (VoIP) in IPv6 environment. Based on the finding result, it shows that all audio and voice codecs suffered higher delay and packet loss for IPv6 compared to IPv4. The significant result of jitter IPv6 is slightly lower than IPv4. Therefore, the quality of voice and video (VoIP) might be decreased if implemented in wireless IPv6 environment.

ABSTRAK

Voice over Internet Protocol (*VoIP*) telah menjadi universal dalam beberapa hari kebelakangan ini dan telah menjadi pilihan utama syarikat kecil dan sederhana untuk komunikasi suara dan video melalui rangkaian IP yang dapat mengurangkan kos dan penggunaan sumber IT dengan cara yang lebih cekap. Teknologi popular lain yang mendominasi dunia selepas tahun 2000 adalah rangkaian tanpa wayar 802.11. Medium tanpa wayar ini mempunyai keperluan khas berbanding dengan rangkaian tanpa wayar 802.3 (*Ethernet*). Pertimbangan khas ini diambil kira semasa melaksanakan video dan suara melalui IP (*VoIP*) melalui rangkaian medium tanpa wayar. Salah satu isu penting bagi rangkaian tanpa wayar 802.11 dan 802.3 adalah ketersediaan jalur lebar. Pelaksanaan VoIP melalui rangkaian tanpa wayar 802.11 mesti mempertimbangkan ketersediaan jalur lebar untuk mengekalkan kualiti suara yang baik. IPv6 dianggap sebagai protokol Internet untuk generasi seterusnya. Kami akan menggunakan platform rangkaian tanpa wayar untuk menjalankan dan mencapai objektif kajian ini. Oleh itu, kajian ini adalah untuk menganalisis prestasi suara dan video serta mengukur Kualiti Perkhidmatan (*QoS*) yang disampaikan oleh IPv6 menggunakan pendekatan codec terbaik berbanding dengan IPv4. Ia lebih memfokuskan pada kualiti suara pada codec suara dan video seperti kehilangan paket (*packet loss*), masa kelewatan (*delay time*), dan *jitter* dalam IPv6 melalui *soft phone*. Aplikasi X-lite digunakan untuk komunikasi di antara sumber dan destinasi. Dalam eksperimen kami, terdapat empat jenis codec audio yang telah diuji seperti G.711 a law, Speex, BroadVoice-32, iLBC dan dua codec video seperti H.263 dan H.263+. Sistem Pengurusan Rangkaian (*NMS*) digunakan untuk memantau dan menangkap prestasi suara dan video (*VoIP*) dalam persekitaran IPv6. Berdasarkan hasil penemuan, ia menunjukkan bahawa semua codec

audio dan suara mengalami kelewatan dan kehilangan paket yang lebih tinggi bagi IPv6 berbanding IPv4. Hasil yang jelas dari IPv6 adalah sedikit lebih rendah daripada IPv4. Oleh itu, kualiti suara dan video (*VoIP*) mungkin menurun atau berkurangan jika dilaksanakan di dalam persekitaran IPv6 tanpa wayar.

ACKNOWLEDGEMENT

Bismillahir-Rahmanir-Rahim first and foremost, I am very grateful to Allah the Al-Mighty because gives me the strength to complete this mater thesis. I would like to show my gratitude to my beloved family especially my wife Nor Amirah Binti Hasnan, who have always supported and inspired me all the way until this master thesis is completed. Without their support and prayers this master thesis would not be completed.

I owe my deepest gratitude to Prof. Dr. Mohd Nazri Bin Ismail and Prof. Ts. Dr. Omar Bin Zakaria La Isms for his supervision, advice and guidance from the very early stage of this master thesis as well as giving me extraordinary experiences through the work. Above all and the most needed, they provided me unflinching encouragement and support in various ways.

Many thanks go in particular to my boss and colleagues at Bank Kerjasama Rakyat Malaysia Berhad (Bank Rakyat) for sharing their moral support and tolerance in order to smooth the journey to complete this master thesis.

Last but not least, I would like to spend some time to express my sincere gratitude and appreciation to those people who helped me either directly or indirectly throughout the preparation of this master thesis.

APPROVAL

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LIST OF ABBREVIATIONS

Abbreviation	Definition
CODECS	- coder-decoder, a software for compressing and decompressing audio and video
VOIP	- Voice Over Internet Protocol
IPV4	- Internet Protocol version 4
IPV6	- Internet Protocol version 6
IPTV	- Internet Protocol Television
NMS	- Network Management System
PSTN	- Public Switch Telephone Network
SIP	- Session Internet Protocol
QOS	- Quality of Service
TCP/IP	- Transmission Control Protocol/Internet Protocol
TCP	- Transmission Control Protocol
UDP	- User Datagram Protocol
ITU-T	- International Telecommunications Union-Telecom
MOS	- Mean Opinion Score
RTP	- Real-time Transport Protocol
RTCP	- Real-time Transport Control Protocol
GUI	- Graphical User Interface
BSS	- Brekeke SIP Server
NAT	- Network Address Translation
LAN	- Local Area Network

WAN	-	Wide Area Network
WLAN	-	Wireless Local Area Network
RSVP	-	Resource Reservation Protocol
WFQ	-	Weighted Fair Queuing
MPLS	-	Multiprotocol Label Switching
V2OLTE	-	Voice over Long Term Evolution
LTE	-	Long Term Evolution
KPI	-	Key performance indicators
VOD	-	Video on Demand
AVOD	-	Advertising Supported Video on Demand
PBX	-	Private Branch Exchange

CHAPTER 1

INTRODUCTION

1.1 Background

Recently, people started using VoIP technology to communicate with each other. Nowadays a lot of PSTN technology or Public Switch Telephone Network users switch their phone to VoIP. It involves the voices transmission via packets switched Network (PSN). As we know, the concept behind video conferencing is to integrate video and voice to connect remote users with each other by compressing audio and video stream to be transmitted through a digital network. There are three main elements of VoIP which is the sender, the IP network and the receiver [1]. The most popular VoIP technology is Skype. This technology allows users to do video conference, video call, chatting, file sharing either for individual or business purposes. There are needs some components to setup the video conference to make it become successful. The input device such as video camera/webcam, microphone, headphone/earphone, and network connection also the output device such as monitor, and speaker is very important for setup video conferences. The last components to make VoIP conference more compatible are the codecs to ensure data can be encoded and decoded accordingly and indirectly VoIP becomes more stable.

For this research, it used SIP (Session Internet Protocol) as a service to establish, modify and terminate VoIP calls. It acts as a signaling protocol for Internet applications. SIP is used for video conferencing, online gaming, peer-to-peer application, instant messaging, virtual reality, and voicemail. It also has the capability to support mobile applications.

In addition, the function of SIP is an IP telephony protocol signaling for initiating a two-way communication to establish a phone call. Example, using softphone as a medium to initiate a conference and it use a SIP server (Session Initial Protocol) as signaling for VoIP. It is like when we are using a regular phone to enter the number for calling, hear the phone ringing or a busy signal. The main difference according to a regular phone's protocol with VOIP which the use of SIP protocol in internet as device for transporting the call, thus, there is no regular circuit in the network is established [2].

The problem that occurred in IPv4 addresses has been improved to a new version of the Internet Protocol in IPv6. IPv6 protocol has many improvements such as the increasing of the address space from 2^{32} to 2^{128}, mobility, more reliable security and quality of service [3].

In this research It focuses on the comparing between IPv4 version and IPv6 version during the data transfer process which means the data will collect and analyze after VoIP conference or VoIP calls. The result will show the differences in performance over IPv6 compared to IPv4 in VOIP. The use of codecs will determine the quality for video and voice of the VoIP conference. VoIP technology relies on the performance of codecs because codecs will transform into digital voice packets. The selection of codecs involves voice quality, processing power and bandwidth requirements.

The QoS (Quality of Service) is used to measure the performance for both versions (IPv4 & IPv6) of VoIP. It can be measured by means being able to hear and speak with clearly and continuous voice without the disruption of noise. Opinion Score will be applied to make sure that the quality of voice or video communication is either good or bad. During the VoIP session, noisy, jitter, delay and echo will interrupt the conference or call. By using the QoS the problem can be solved efficiently and consistently. Figure 1.0 below shown the basic of VoIP network architecture in IPv4 and IPv6

Therefore, improving VoIP QoS (Quality of Service) has become the industry's research priority. IP network always faces sudden traffic, when a sudden traffic appears, the network will congest, resulting in delay and jitter that will seriously affect the real-time VoIP services [5]. For real-time voice communications services, QoS becomes difficult to guarantee. In order to make VoIP provide equivalent quality of service to PSTN technology, IP networks require VoIP to provide new mechanisms to ensure a reasonable and capability of QoS. The performance and quality of the VoIP calls and conferences will depend on each other [4], [6]:

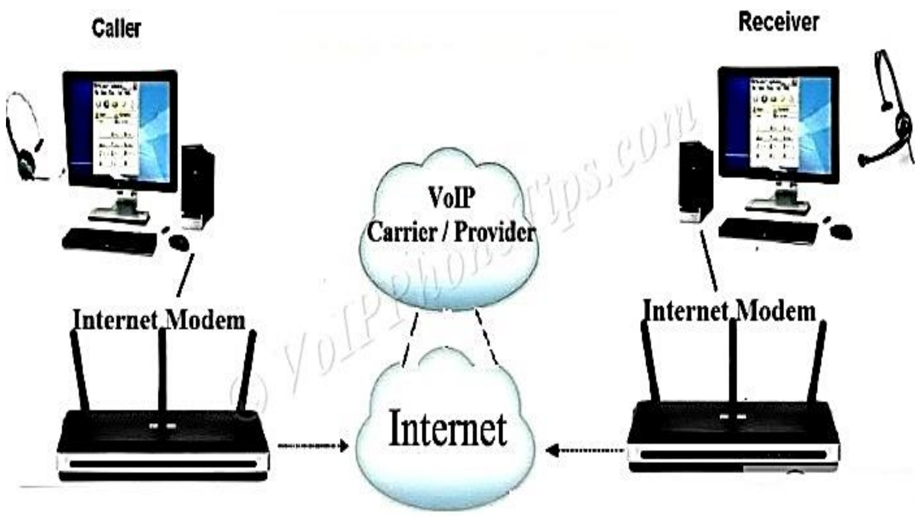

Figure 1.0: Basic VoIP network Architecture. From *How does VoIP work? most VoIP frequently asked question*. Most VoIP Frequently Asked Question. (n.d.). https://www.voipphonetips.com/#axzz8aYbDCIBR

1.2 Problem Statement

VoIP (abbreviation for Voice over Internet Protocol) is a digital signal different from the analog voice signals. It uses the data packet to do real-time delivery in the IP Network architecture. The advantage of VoIP is it uses the global and wide environment of Internet to provide better and cheaper services more than traditional environment [4]. However, IP telephony's voice quality is poor compared to traditional telephone, one of the important reasons is that Internet bandwidth always changed. For example, when network congestion occurs, packet loss is much and indirectly leading to voice distortion. Bandwidth refers to the Internet can transmit information how many bytes per second. The greater the bandwidth, the more favorable transmission of data services, but whatever transmission medium's, the bandwidth is limited. Delay, it's the time difference between received data packets and sending packets. There are various types of delay algorithm such as processing delay, network delay and jitter buffer delay [10]. If the packet loss rate is too high, it increases the drops rates packets in the transmission of data packets in the network. Packet loss rate should be less than 5%, if more than 10% it will affect the quality of service. Normally the loss reason is the line network routing error or failure, transmission delay is too long or the network congestion that caused packets to be discarded. Jitter known as delay variation; it's referred to various delays in the network leads to data packet arrival rate changes. If the network jitter is increased, some voice packets may be discarded because of late. In addition, delay variation should be less than 10% as well. Normally Jitter is caused by queuing delay, variable packet size; intermediate links and routers on the relative load [7]. Packets out of order occur when the network is poor, voice packets in the transmission process is exposed to out of order and affecting the play of receiving end. However, according to the timestamp of each voice packet, it can easily be determined by the order of the sending voice packets. A commonly used solution is jitter buffer at the receiver, reorder the packets of disorder, and reproduce the originator of the order [8]. Echo (Electric echo) on traditional phone system, there is 2-4 wire conversion. In the telephone voice transmission, mixer need to complete conversion of 2-4 impedance mismatch and resulting mixer voice "leakage." Whereas on electrical network delay, if exceeds 25ms will have impact on the speaker. Acoustic echo is the speaker's sound picked up by the microphone sent back to the remote, which makes remote callers hear the echo [9]. Therefore, in this study need to determine the best CODEC selection for voice and video to increase the quality of the conversation between two parties. The best CODEC selection for real time voice and video can minimize packet loss, jitter, delay, packet out of order and echo [4].

1.3 Objective of Study

Below are the objectives of this study:

i) To explore the VoIP characteristic and performance measurement over IPv4 and IPv6.

ii) To compare the CODEC performance for VoIP in IPv4 and IPv6 environment.

iii) To propose the best CODEC techniques for VoIP in determine the best VoIP CODECS selection calls and conferences quality over IPv4 and IPv6 environments.

1.4 Research Goal

The goal of this research is to propose and determine the best CODECs for VoIP calls and conferences via IPv4 and IPv6. This study investigates and explores the effective performance during conversation between sender and receiver. This goal required a simple test-bed setup for the best CODECs selection for VoIP IPv4 and IPv6. The robustness of the proposed selection CODECs was taken into consideration as it determined if the proposed approach able to produce good results when tested on different experiments and scenarios.

The contributions and benefaction of this research are:-

i) To strengthen an important knowledge on V2oIP performance in wireless IPv6 environment using best codecs selection over IPv4 and IPv6.

ii) The results of the V2oIP performance in IPv4 and IPv6 environment are useful and can be used as a benchmark for commercial ISPs in next generation networks, especially in defense and security industries which is always challenging.

1.5 Scope of Research

Voice over IP (VoIP) research areas are divided into two categories which is TCP/IP version 4 (IPv4) and TCP/IP version 6 (IPv6). The scope of this research focuses on the best CODECS selection for VoIP calls and conferences over IPv4 and IPv6. Softphones will be used in both parties for the calls and conferences. VoIP management tools will be used to capture the VoIP quality over IPv4 and IPv6 using various CODECS selection in wired and wireless (wi-fi) environment. Mean Opinion Score (MOS) technique will be applied to rate the VoIP calls and conferences quality. The quality of the VoIP call and conferences will measure based on delay, latency, packet loss and jitter.

1.6 Organization of the Thesis

The outline of this thesis paper is organized as following structure:

Chapter 1 provides the introduction, aims and objectives, research methodology, motivation and contribution of this research are discussed, and also discusses about the scope of the thesis.

Chapter 2 presents a comprehensive literature review, it covers the general overview of the V2oIP technology, IPv6 platform, VoIP Codecs, Quality of Service (QoS) on V2oIP such as delay, jitter, packet loss, queuing delay, V2oIP metrics, Session Initiation Protocol (SIP). Finally, it includes a comprehensive review on the related work to this study.

Chapter 3 describes speech Codec for V2oIP softphone as well as Codecs quality evaluation methods. Chapter 3 also discusses how to improve VoIP quality during the planning.

Chapter 4 dedicates a discussion of the experiment setup, network scenarios and the parameters required to configure them.

Chapter 5 explains the real test bed experimental result on V2oIP via IPv6 and IPV4 platform. This study is using Wireless Network medium as case study.

Chapter 6 concludes the entire thesis work and future research works.