PERFORMANCE ANALYSIS OF VOICE CODEC FOR VOIP

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Thank To My Lord, Allah,

My Parents And My Family Members,

My Related And All Friends.

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"In the name of Allah, the most Beneficent, the most Merciful"

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ABSTRACT

Recently, VoIP (Voice over Internet Protocol) is a great interesting voice communication over the Internet, with high level quality of service (QoS) along with circuit switch and cellular. The objective in this project is to assess to what extend today's internet in meeting this expectation via studying VoIP performance and its QoS. However, the methodology in this project is, first the CODECs are selected by some criteria then apply them on SIP server to finally come out with the result from the simulation in order to make comparison and analysis the QoS. This work implements VoIP protocols for two connected user using SIP server with its three CODEC algorithms. After define the main problems in this area set of parameters are taken into account due to their affection to the performance of the voice, such as jitter, packet loss, packet delay and throughput. This project is simulated three existing CODECs (converting the voice from analog to digital and compressing the packets) using the most common CODECs with VoIP, they are G.711, G.723 and G.729. However, the simulation will use NS2 platform with vary values of packet size and number of calls. Finally, the main objective from this project is to obtain a high quality of voice by make a proper decision for choosing the codec voice. As conclusion, G.711 is a preferred technique when the quality is required because of the high throughput from its packets, while G.723 perform well with the high bandwidth means it can handle many user. Finally, G.729 the high level compression is the proper technique for many user and heavy data only when the quality is not taken into account.

ABSTRAK

Belakangan ini, VoIP (Voice over Internet Protocol) adalah suatu komunikasi suara melalui internet yang menakjubkan, dengan kualiti servis (Qos) pada tahap tinggi seiring dengan suiz litar dan selular. Objektif projek ini adalah untuk mengenalpasti perkembangan Internet pada hari ini yang mampu memenuhi kehendaknya melalui kajian terhadap pencapaian VoIP dan QoS. Tujuan projek ini adalah, yang pertama CODECs inalah memilih beberapa criteria kemudian mengoplikasikan ia dalam SIP server dan akhirian mengeluarkan keputusan dari simulation untuk membuat perbandingan dan analisis QoS. Projek ini menunjukkan protokol-protokol VoIP untuk dua pengguna yang berhubung melalui pelayan SIP dengan tiga algorithma CODEC. Setelah mengenalpasti masalah utama dalam bidang ini, suatu set parameter telah diambilkira bergantung kepada kesannya terhadap persembahan suara seperti jitter, kehilangan paket, kelewatan paket dan throughput. Projek ini akan menggunakan tiga CODEC yang ada (menukarkan suara dari analog ke digital dan memampatkan paket tersebut) menggunakan CODEC dengan VoIP yang paling sering digunakan seperti G.711, G.723 dan G.729. Walaubagaimanapun, simulasi ini akan menggunakan platform NS2 dengan nilai saiz paket dan bilangan panggilan yang berbeza. Akhir sekali, objektif utama daripada projek ini adalah untuk mendapatkan suara yang berkualiti tinggi dengan memilih jenis CODEC yang sesuai. Kesimpulannya, G.711 merujuk kepada teknik apabila kualiti diperlukan kerana the high throughput dari packets, sementara G.723 membuat yang the terbaik dengan the high bandwidth bermakna ia mampu menangani ramai pengguna. Akhirnya, G729 levvel yang tertinggi compression ilalah teknik yang terbaik untuk ramai pengguna dan data berat hanya apabila kualiti tidak masuk dalam akaun.

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PROJECT OVERVIEW

1.1. Introduction

Voice communication recently became daily used whether by cellular, land phone PSTN or voice over Internet protocol (VoIP), the last one has been invented recently, so fast VoIP became very wide among the people. Nowadays, a human can communicate easily to all over the world whereby their IP contributed many services and protocols. Unfortunately, until now there is no clear guarantee to transmit the data over the network (Doherty, Anderson, 2006). Therefore, sending the samples of voice (packets) still does not reach the satisfaction of users, due to certainly the VoIP does not be as clear as the human conversation. Recently, after the VoIP has been invented, the next challenge for the researchers and engineers was the quality of voice service.

Starting from converting sender’s voice from analog to digital, then sending that digital through the network, terminating by converting back the digital frame to analog at the receiver side, many algorithms and techniques were occurred in VoIP life cycle.

On the other hand, we need to improve certain techniques and codec in order to obtain high quality of voice. Many works, tools and techniques have been published and built with fully concerned about how to choose a proper codec techniques, the used codec should has suitable properties and features which aim to send a light and clear voice sample.

This proposal will investigate VoIP existing speech codecs techniques, aiming to study and compare encoding/decoding techniques with the certain parameters delay, jitter, throughput and packet loss, during transmitting the data through the network.

1.2. Problem Background

Any computer system has many methods to play the sound and also should have algorithm to compress the speech data, either for storage that file or send it through the network. However, compression the audio for sending it much more difficult than just storage, due to it is a real time operate, attempting to make the conversation as near as the real talk. On the other hand, when the voice navigates across many devices such as router, server, and the media itself, those things certainly will decrease the quality of voice until it reaches the receiver.

Not only voice compression in the sender side is a problem, but also the receiver, the protocols between the sender and the receiver should be clear, if the sender encodes the voice by such codec, on the other side the receiver should decodes the voice with codec has familiar protocol with the sender (Honathan, 2007). In addition, several codecs are not adapted with WLAN but they are with LAN.

Problems that might happen over VoIP can be divided into three categories according to codec used (Honathan, 2007). Table 1.1 shows simply and generally the weakness of each codec and where the straight. The table reviews each codec how the jitter, delay, placket loss, and throughput have been affected to them.

Table 1.1: Weakness for Each Codec

Parameters	**CODECS**		
	G.711	**G.729**	**G.723**
Jitter	High	Low	High
Delay and loss packet	Low	High	Low
Throughput	High quality	Normal	High

Many techniques were introduced and applied throughout the whole project lifecycle just to make a good decision for choosing a better codec, these techniques are differ in their objectives and appliance, but it all gathers under best effort encoding/decoding voice to reduce the main three common parameters as mentioned above such as jitter, delay, and packet loss.

Encoding/decoding voice are the most serious problems during one way End-to-End VoIP (Oliver, 2000). Since other algorithms are depend on the situation of the network. On the other word, encoding the speech from analog form to digital representation (samples) and packetizing the data are occur in a critical time, parallel with high level of compression. All these problems are taken into account, in order to obtain a light data ready to send.

Users' satisfaction also consider as a big problem, if the users want to compare the quality of VoIP with circuit switch service (Doherty, Anderson, 2006), certainly they would complain from the inaudible conversation and from the delay of voice arriving specially if the calls come from long distance. MOS (mean opinion score) is a common measurement tool for voice quality (Alias and Ong, 2006), as described in the ITU recommendation P.800, the relation between audio performance characteristics and quality score make the MOS standard for network evaluation, from the table 1.2 MOS can be range from 5 (excellent) down to 1 (bad). Whereat MOS of 4 or higher is generally considered toll quality, and a MOS below 3.6 results in many users who are not satisfied with call quality.

Table 1.2: The Mean Opinion Score Scale (ITU)

MOS	**Quality Rating**
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

From all various problems above show how really important to focus on the quality of voice service by choosing better speech codec techniques.

1.3. Problem Statement

Reports (Alias and Ong, 2006; Amir, Tarik and Noor, 2005 and Kee, Yin and Moh, 2005) have shown that VoIP projects tend to attempt many compression techniques (CODECs) to obtain high quality of service. Similarly, they applied that codecs to other network area. Recently, researches are encountered a same area problem which are choosing a proper codec and control the main factors. In contrast to traditional views, the problems within VoIP are linked with each other.

1.4. Project Objectives

The objectives of this project are stated in the following points:

1. Investigate and analyze the audio encoding/decoding techniques in VoIP.

2. Simulate the selected encoding/decoding techniques of VoIP using NS2.

3. Evaluate and test the performance of VoIP codecs.

1.5. Project Scope

The objectives of this study are stated in the previous section. In order to achieve the study objectives, it is important to highlight the study area and its boundaries, which are stated in the following points:

1. Testing and simulation are going to be used on SIP server exactly with NS-2.
2. Only address the coding/decoding techniques.
3. Re-coding techniques will not be addressed in this project.
4. Linux Fedora Core 2 as an Operating system to install NS2.

1.6. Project Justification

The primary measurement of success VoIP quality is to perform high quality of voice measured from users' satisfaction. Any project has its own problems, from this point determining the project success or fail comes from how this project can manage and control the problems.

Initially, codecs often affect the performance of VoIP whether by increase or decrease the quality, due to each codec is familiar with certain network such as wired or wireless, and also some codecs are able to work well with high number of calls while others are not. One more, some codec is utilized for compression the data. On the other hand, codec supplies a high quality of voice but it cost high bandwidth. This project tend to determine each situation and which the recommended codec and where to use. In addition, this project will compare the three codec to make a clear idea where the

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