Voice Quality Performance Measure in VoIP using SIP Protocol Application

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ABSTRACT

Voice Quality Performance Measures in VoIP using SIP protocol Application

Voice over Internet Protocol (VoIP) is a technology that delivers voice using the Internet Protocol (IP). VoIP is a term that used in IP telephony for a set of facilities for managing the delivery of voice information using the internet protocol. Using VoIP method can avoid the tolls charged by ordinary telephone service. In this scope, this research is focuses on quality of transmission data using Session Initiation Protocol (SIP) that use UDP or TCP protocol. This research also focuses on measurement of the performance based on the quality of voice, jitter, delay and throughput. This project is doing with some approach and methodology.

Methodology that been used in this project is planning, where information, software and hardware requirement are determined by using Windows XP as a platform, Express Talk as the SIP application, and some VoIP analyzer such as Ethereal, Acterna PVA-1000 analyzer, and Qcheck. Next, for implementation and testing showed that the SIP application is connected between two nodes. This testing measured the performance of Quality of Service (QoS) for jitter, delay, voice quality that captured by Ethereal. Then, the data from Ethereal will be analyzed by Acterna PVA-1000 VoIP analysis. For the throughput, Qcheck is used as an analyzer. Data gathered and illustrated in the table and graph. For analysis, data will analyze and conclude the findings. SIP application is implemented and the performance of quality voice is measured. Testing is doing ten (10) times a week. The average result is compared with previous study. Then, the results showed that the performance of VoIP using SIP application is similar with other application or protocol.

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CHAPTER 1

INTRODUCTION

1.0 PROJECT INTRODUCTION

Voice over Internet Protocol (VoIP), also known as Internet Voice, is a technology that allows you to make telephone calls using a broadband Internet connection instead of a regular (or analog) phone line. VoIP also is a method for taking analog audio signals, like the kind you hear when you talk on the phone, and turning them into digital data that can be transmitted over the Internet. VoIP can turn a standard Internet connection into a way to place free phone calls. The practical upshot of this is that by using some of the free VoIP software that is available to make Internet phone calls, you are by passing the phone company (and its charges) entirely. In term of quality, IP networks employ the same types of bandwidth-saving schemes as the Frame Relay network, including fragmentation, jitter buffering, prioritization, voice compression, silence suppression and echo canceling.

SIP, the Session Initiation Protocol, is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP provides methods to control sessions, but does not specify the applications and services that will use those sessions; as a result, SIP does not guarantee application behavior. SIP is independent of the media used, allowing the flexibility to initiate sessions for different media types.

SIP is relatively new on the VoIP standards scene. SIP is an application-layer control protocol that makes up for many of H.323's inherent faults. The standard, developed by the Internet Engineering Task Force (IETF), addresses the call setup and teardown, error