UNIVERSITI TEKNOLOGI MARA

PERFORMANCE ANALYSIS OF VOICE OVER IP (VOIP) OVER MOBILE BROADBAND (HSDPA)

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بسم اللهالرحمنالرحيم

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ABSTRACT

This paper focuses on Voice Over IP (VoIP) over mobile broadband. The objective of this paper is to analyze the performance of VoIP over HSPA through two parameters which are the jitter and mouth-to-ear delay parameters. The drive test was conducted to collect real measurement data in live HSDPA network. The data are analyzed to examine the behavior of jitter and mouth-to-ear delay where both of the parameters impact the quality of the conversation made by VoIP. From the experiment result, it is clear that with small packet size and large buffer jitter may bring the large latency and may cause the mouth-to-ear delay. As known, the large number of mouth-to-ear delay dramatically degrades the quality of VoIP call

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CHAPTER 1

INTRODUCTION

1.1 BACKGROUND

During the last few years, the voice over data network services have gained increased popularity. Quick growth of the Internet Protocol (IP) based networks, especially the Internet, make the Voice over IP become widely used.

The number of worldwide VoIP customers reached 38 million at the end of 2006 and it is projected that there will be approximately 250 million by the end of 2011 [1]. VoIP is a reality nowadays and each day, more and more individuals use this system to phone around the world because there are many common programs that make it easy to use VoIP such as Skype, MSN Messenger, VoIPcheap, and so on.

QoS (Quality of Service) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped along the transmission. Things to consider are;

- Latency: the time from the start of packet transmission to the start of packet reception
- Jitter: Variations in delay of packet delivery
- Packet loss: Too much traffic in the network causes the network to drop packets
- Burstiness of Loss and Jitter: Loss and Discards (due to jitter) tend to occur in bursts