ABSTRACT

Voice over Internet Protocol (VoIP) is a technology that can pass the voice traffic, video and data packet form over an IP network. By using a VoIP phone, a lot can be taken advantage of them is certainly cost much cheaper than traditional phone rates.

The selection of the data compression is very important due to voip audio codec affects the quality of VOIP service. One purpose of this selection of audio codecs is to reduce the bandwidth so that we can save bandwidth.

In this Final project, has conducted measurements and testing of the performance of two iLBC codec and SIP VoIP SPEEX on using Asterisk 1.6 and obtained the average value - the average delay for the iLBC codec is 124ms, 23ms jitter, packetloss 0%, MOS 4.4 and R Factor for the codec 93 while SPEEX delay 202ms, 32ms jitter, packetloss 0%, MOS and R Factor 4.4 VOIP SIP 92. As for the minimum bandwidth can still digunkan codec - codec is 16kbps.

From the results of this study is expected to produce a clear reference and can help us to choose a good codec while we want to build a VoIP network.

Keywords : audio codecs, VOIP SIP, iLBC, SPEEX, delay, jitter, packet loss.