ABSTRACT

The development of telecommunications technology rapidly making the

need to communicate to get information over the phone higher. On the other hand,

call center are typically used in the office or university to make it easier to get

information, still using circuit based communications. Whereas, in the current

world of global communication, the trend began to shift from circuit or PSTN

network to the model communication via IP or commonly known as VoIP (Voice

over IP). Popular communication in this era is web based communication, where

the web began to be applied to be a call center which are the communications

using IP, so the function of the call center that serves only call can transform into

the contact center which has free calls, chat and FAQ. To support digital voice

data lines operating in the internet protocol requires a hardware called IP PBX

which is the function of the IP PBX can be replaced by software called Asterisk.

The result of this final project is a contact center system that has features

call audio, audio-video calls, chat and Frequently Asked Questions te goals to

allow users to get the information in various ways. Call features both audio and

audio-video using the concept of VoIP with SIP protocol by utilizing Asterisk

servers that have a PBX functions.

From the results of tests performed, the results obtained from the audio

call is one way delay by an average of 68 977 ms, with an average jitter 13 753

ms, and packet loss by an average of 4.8%. MOS values obtained from an audio

call is for 4328 and for video calls is at 2:04. The quality of audio calls can be

categorized into "Good" because it has a MOS value 4.328 of 5, while the quality

of video calls can be said not too good because it has MOS value 2.04 of 5.

Keywords: VoIP, Asterisk, Contact Center, SIP