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