ABSTRACT

Voice Over Internet Protocol (VoIP) *is a technology capable of passing voice, video, and data traffic in the form of packets in real-time with an Internet protocol network*. *This VoIP can take advantage of the existing internet infrastructure to communicate like using a regular telephone and is not subject to regular telephone charges to communicate with other VoIP users anywhere and anytime*, *where analog voice signals are converted into digital voice signals and then package the voice*. *In the process*, *a codec is needed*. Codecs *to be tested include the* G711 A-law, G711 u-law, G722, G729, and G723. *In analyzing the quality of codecs that are compatible with the Asterisk* server *by calculating the byte rate of codec payload*, *Delay*, *Throughput*, and *Packet Loss*, Wireshark *software is used*. *Analysis of the* codec *quality aims to be used as a reference in choosing a good codec in building Voice Over Internet Protocol* (VoIP). G711 A-law codec *has the best value of the* 5 codecs *that have been analyzed, the* G711 A-law codec *has the highest Throughput value of 5 trials, which is* 42,803 *bytes in the call* 3 *device and streaming test, for the Delay value which is 4 ms in the call* 3 *test. The device, and* 0% *Packet Loss value.*

Keywords : VoIP, Codec