# 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