

Abstract - Multimedia services are one of the internet needs with high data traffic network count since 2019. Two of the multimedia services, Web Real-Time Communications (WebRTC) and Session Initiation Protocol (SIP), have been widely used in applications for conducting video conferencing. The main objective of this research is to analyze network performance by an application with a client-server for audio and video communications developed with WebRTC and SIP protocols. The SIP system uses the FreePBX server, and the softphone application uses Bria. Whereas WebRTC uses JavaScript with servers on Ubuntu using Node.js. The analysis application uses a star topology and runs on a local network using Wi-Fi. After testing, the results show that the throughput, jitter, and packet loss of WebRTC are better than SIP. This result is caused by several factors including the type of codec, the type of platform, and the way of signaling and routing of each protocol

Keywords: WebRTC; SIP; Video Communication; Codec; Network Performance.