

## I. INTRODUCTION

Multimedia applications and services are one of the highest needs of the internet with data from calculations in network traffic since 2019 [1]. Of the various existing multimedia applications and services, Web Real-Time Communication (WebRTC) and the Session Initiation Protocol (SIP) protocol are two protocols that are interesting to discuss. WebRTC is a technology that exists in the field of communication using an Application Programming Interface (API), with JavaScript in the browser application. With this technology, WebRTC allows users to communicate using video and audio directly from the web without the need for extensions or additional tools [2] [3].

Currently, WebRTC still uses an API which is designed to communicate from browser to browser only, because of this architecture, WebRTC is not recommended for communicating with the conference model because it can cause a load on the existing connection in the media API, it can affect user comfort when doing communication. The SIP protocol is communication in a network between multimedia devices using two protocols, which can be called the Real-time Transport Protocol (RTP) or the Real-time Transport Control Protocol (RTCP) and the Session Description Protocol (SDP).

This SIP protocol will be run by the GUI (Graphical User Interface) on the FreePBX server that manages Asterisk (PBX) [4]. This study aims to conduct a network performance comparison of WebRTC and SIP audio and video communications.

This study will use WebRTC based on Ubuntu 18.04 LTS and node.js as a server, and for the SIP protocol using FreePBX as a server. This test is conducted to determine the network quality of these 2 applications and has various parameters to be tested as throughput, jitter, and packet loss.