ABSTRACT

Currently, various IP-based software has emerged that lead to the Next Generation Network. Some of the software includes OpenSIPS, Trixbox, OpenIMS, and Asterisk. Then came a new Kamailio VoIP platform which has reliable performance. A problem arose when the VoIP platform was expanding but still wanted to use the previous VoIP platform. Therefore it is necessary to do inter-server interconnection so that communication can be done.

In the final project entitled "Performance Comparison Analysis of Kamailio VoIP Server and Asterisk Server and OpenSIPS Server" an interconnection will be carried out to connect between servers so that VoIP video call services can take place and the reliability of Kamailio servers can be utilized, which will be applied to the lecturer room. Faculty of Applied Sciences, Telkom University. Call routing techniques are used as the most reliable method of connecting separate servers. Then proceed by measuring the performance which includes Post Dial Delay, Delay Process, QoS, and MOS.

In this test, it was found that the average PDD value of the Asterisk server was 272 ms, the OpenSIPS server was 581 ms, and the Kamailio server was 158 ms with an average PDD value on an interconnection system of 600 ms. Obtained the smallest delay process value on the Kamailio server with a value of 0.789 ms.

Keywords: *VoIP, Video Call, Packet Switch, Asterisk, OpenSIPS, Kamailio.*