ABSTRACT

Telecommunication technology is growing rapidly. This rapid development was followed by diverse communications services. These diverse services are needed by the user. Required communication services has reached the voice and video services. In addition to diversity, surely also expected to have the quality of service according to the standard and have low cost. Voice and video communication services must also be real-time and reliable. To meet the desired communication services, so that's why need to be developed in the network services based on IP (Internet Protocol), then the service will be evaluated the quality of some parameters.

In this final project, implementing Mobicents application servers that support WebRTC. Mobicents server as a stand-alone application server can support WebRTC services. Communication is done utilizing the browser as a medium. Later in its implementation, will be the delivery of voice and video between clients. From these results, it will be analyzed the performance of the WebRTC services.

In this final project seen its performance QoS parameters that include delay, jitter, throughput and packet loss and parameters for server performance such as CPU usage and memory usage. From the measurement results obtained that the background traffic variations from 0 Mbps to 80 Mbps given, the average value of delay is 5.3753254 ms , jitter is 0.4697894 ms, throughput is 275,567.2277 Bps and packet loss is 0.020305%. The results still meet the standard of ITU-T. The highest result of CPU usage is 37.5 % when start server process and the results of memory usage increase from 17.88 % to 30.7 %. In this final project, also implemented on different networks. QoS value in the wired network still meet the standard of ITU-T.

Keyword: WebRTC, Internet Protocol, Mobicents, QoS