Analisis QoS pada WebRTC dengan Mekanisme Congestion Control di Jaringan Lokal

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Abstract
With real time video and audio communication changing the way people communicate over the Internet significantly, this is supported by the emergence of Web Real-Time Communication (WebRTC) which allows the use of web browsers for streaming multimedia transmissions. But there are challenges, namely how to regulate the quality of delivery time conditions, traffic crowded to determine the efficiency of the implementation of WebRTC in the network, the QoS parameters used are throughput, packet loss, delay andjitter.In this final project, the results of the video stream that can be transmitted using WebRTC technology, are compared with one that is not controlled by congestion, namely UDP transmission using iPerf tools. By applying the congestion control scenario, it is expected to measure the quality of multimedia streaming services.

Keywords: WebRTC, Quality of Service, DSCP, Real-time Communication, UDP, Video Conference