The Quality Comparison of WebRTC and SIP Audio and Video Communications with PSNR

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Abstract

Video and audio communications have become part of all areas of work. Two real-time communication protocols commonly used for IP-based video and audio communicationsare Session Initiation Protocol (SIP) and real-time web communications (WebRTC). Both protocols have been widely used in softphone and video conferencing applications. Themain objective of this research is to make an analysis of the performance of a client server application for video and audio communications developed by SIP and WebRTC. The SIP system consists of a softphone on the client side using Bria and a FreePBX server, for WebRTCapplications, using JavaScript and a server at Node.js. The results showed that the WebRTC audio and video communication provided better quality in terms of PSNR. This is due tothe different codecs used between WebRTC and SIP. WebRTC uses VP8 as video codec, SIP uses H.246 as video codec, WebRTC uses G.711 as audio codec, and implemented SIP uses G.729 as audio codec.

Keywords: Video Communication, WebRTC, SIP, PSNR, FreePBX, NodeJs, Video, Audio.