ABSTRACT

This final task is made a system to integrate a VoIP client with a web-based VoIP softphone client applications that already exist. Later VoIP client web based application can be accessed via a web browser and can be used to communicate with existing VoIP applications. For the server that handles the signaling in system was tested using two scenarios VoIP server.

Web-based VoIP client application was built using WebRTC API in browser and WebSocket protocol as support SIP session initiation process. With WebRTC browser technology to conduct real time communication of voice and video conversations, such as video chat, without the need for plugins. While WebSocket technology are used to handle part of the session initiation with its ability to provide full-duplex communication channel through a single TCP connection between client and server.

From the results of testing the integration between web-based VoIP client application VoIP softphone Linphone earned both a SIP client can communicate using SIP server that supports WebSocket connection. An average of three scenarios testing the post dial delay largest value obtained 0.46 seconds meets ITU-T E.721 standard. However, the implementation of a web-based VoIP client still has obstacles difference SDP standard. In the system under test differences found in the use of standard protocols SDP media profile, security key management, and parameter synchronization source identifier.

Keywords : VoIP , SIP , WebRTC , WebSocket , PDD