IEEE Africon, Mauritius, 9-13 September 2013

Determining the Signalling Overhead of two common WebRTC methods:JSON via XMLHttpRequest and SIP over WebSocket

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Abstract

Web Real-Time Communication (WebRTC) introduces real-time multimedia communication as native capabilities of Web browsers. With the adoption of WebRTC the Web browsers will be able to use WebRTC to communicate with one another (peer-to-peer), and with WebSocket servers such as Mobicents SIP Servlets and other server technologies that support WebSocket communication to enable SIP-to-WebRTC communication. This paper outlines the potential of WebRTC and discusses the two common methods of doing real-time communication in Web browsers through WebRTC. The methods are JavaScript Object Notation (JSON) via XMLHttpRequest (XHR) and Session Initiation Protocol (SIP) via WebSocket. A three-user WebRTC video chat prototype application was developed and used to evaluate both methods. Additional signalling overhead introduced into a browser by each method was determined. The results showed WebRTC-SIP/WS has more overhead than WebRTC-JSON/XHR. This signalling overhead findings are useful in that they could help application developers make decision on their choice of technologies and protocols when developing WebRTC-supported applications.