**Development of a browser based real-time peer-to-peer remote support application**

WebRTC (Web Real-Time Communication) is an API definition by the W3C (World Wide Web Consortium). It enables users to communicate via audio and/or video stream directly in the browser without the need of proprietary software or external plugins.

At the beginning of this thesis, the core components of WebRTC will be discussed, as well as the various possibilities it offers to developers.

A concept will be created that describes the aspects and components that are necessary to develop a remote support application with WebRTC that works peer-to-peer directly through the web browser, without any kinds of plugins needed.

It should be possible for a user to call another user through the web browser and connect to them via audio/video stream. The called user can see what the calling user sees on his/her device and highlight any parts of the video screen with coloured circles to help them find a solution for a problem. Additionally, the users should be able to communicate through a text-based chat on the same web page.

When users visit the web page, they can provide their name and e-mail address. The name will be visible to every other user who has opened the web page, while the provided e-mail addresses are not visible to any other user. If a user calls another user through the web page, this called user will also receive an automated e-mail message containg a clickable link inviting them to the audio/video chat session. For now, no kind of user management will be used. Everyone who visits the server’s web page will be visible to everyone else.

To set up a peer-to-peer connection, the public IP address of both users in the call must be known. In a real life environment, this might be a problem if one or both of the users is located in a network that uses Network Address Translation (NAT). To solve this problem, WebRTC uses Session Traversal Utilities for NAT (STUN) and Traversal Using Relay NAT (TURN) servers to discover a user’s public IP address. This is not necessary if both users are located in the same network. For this prototype, it is assumed that both users are part of the same network.

To demonstrate the practical functionality of the concept, a prototype will be developed. The prototype consists of a simple management server, whose sole purpose is that users can discover each other via a web page to start an audio/video call through WebRTC. After the call has been initiated via WebRTC, all data exchange and signaling happens directly peer-to-peer, without the management server.

The developed prototype will undergo tests regarding the quality of the audio/video stream connection on a mobile browser in an area with limited WLAN reception. Because WebRTC offers the possibility to pass constraints on the camera’s picture quality, different levels of picture quality will be assessed within these tests.

Because WebRTC is still under development, there are only a few web browsers that fully support it. Therefore, only Chrome, Firefox and their mobile versions will be used.

Conclusively, all insights and experiences from the prototype development as well as the test results will be discussed in detail.

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