WebRTC in context communication

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Extended Abstract

Browsing the web isn't just about text and images anymore. Users want to interact with services and with each other. They need to communicate fast and easily. WebRTC presents itself as an technology that enable (web)developers to deliver applications with an higher level interaction. This presentation shows the work done within GN4SA8 to create an "in context" video centered application that allows users to create a video websnippet to embed on their website, LMS or any other web service.

For the past 20 years, videoconference has been available in dedicated self-contained boxes. With faster and better CPUs available on the PC market, several approaches to videoconference and webconference have been tried with success, but always using private and proprietary protocol stack. In fact, if a company or university wanted to create a video chat application they would have to buy or build their own application from scratch requiring audio and video engineers. With the proliferation of protocols and solutions, more and more islands have been created where each users can only talk with themselves, making the bridging hard and expensive. As result, the market is filled with generic communication tools, there isn't almost any in-context communications services. To produce such applications for user niches wasn't profitable enough to justify all the development effort.

This is where WebRTC presents itself as a disruptive technology, it provides easy semantic to use audio, video and data peer to peer connections. W3C have created the specification and Google has developed and provided an open-source stack that can be used on any application. Today the effort of creating an audio & video chat application is much less than before. Not surprisingly, Google and Mozilla Foundation have integrated this WebRTC stack within their browser and made it available to developers using the javascript API, as defined by W3C. As result, any web developer today is able to create an audio, video and data real time application.

SA8-WebRTC has embraced this new set of tools as an opportunity to explore new services to the NREN community. WebRTC videoconference communicator applications are ready and available, such as Rendez-Vous from RENATER, but these don tap WebRTC's full potential, where easy to use API allows the development of personalized in-context applications.

Focused on the academic usage of in-context application, SA8-WebRTC team has started working on ideas that could provide value added to the user base, such as teachers and students, potentiate NRENs services and explore WebRTC technology.

From the several ideas, the more interesting was the "Tutoring Application", where a teacher can "publish" a virtual session websnippet beside their on-line course or content and the student uses that to access the tutoring using a self-service request form. The teacher then includes the code on is website and the integration is complete. The application provides a teacher interface that he can access from any device to talk with their students, allows him to define the tutoring time window, get information about the queued students and statistics about the sessions that took place. The application exploits the fact that general webconferencing tools don't provide queueing mechanisms or any in-context or user case dependent logic. Besides this "Tuturing use case", easily this system can be used on Helpdesk, Tele-medicine, Call Center or in any other situation where an expert/operator needs to be presented to a population of users, one at a time.

The development team of two 0.2 FTE develpers had proposed to build everything in under two months. Starting November, the first steps were:

- build mockups of the applications and run it thru some users;
- define the architecture and the API for the multi-client signaling;
- prepare the development framework with multi-language and federation authentication support.

The infrastructure was then built over a LAMP with NodeJS. The webserver delivers the webapplication (http), while NodeJS provides the application signaling to manage the video chat session and the application dynamic over "websocket". After the negotiation over the websocket, the WebRTC API is called to make the p2p connection where the meeting occurs.

Before the end of the year, a prototype was available and the first run of tests with users had started.

By then end of January, the first software package version was ready to be distributed by NRENs and Universities.

Without WebRTC this application would not have been possible to build, more over in this time frame.

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Author Biography:

He studied computer science at Lisbon's University and complemented his studies when he enrolled on a Masters on Science, Technology and Innovation management. Part of the NREN community since 1998. He has developed videoconference, webcast, webconferencing, telepresence and other innovative video-based services. He was a member on the several TERENA Task Forces and contributed to TERENA's SIP Handbook. Currently manages the video services infrastructure at FCCN and is an active of GN4 project.

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