

Integration of WebRTC with SIP – Current Trends

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Abstract— Web real time communication (WebRTC) is an important part of modern convergence networks. Interactive voice communication, video conferencing, chat messaging are today very popular and widely used Internet Protocol (IP) network services and applications. The Session Initiation Protocol (SIP) is a crucial communication technology that brought new and standardized control mechanism. The reviewed literature addresses the various issues in context of WebRTC such as interaction with SIP-Based Conferencing Systems, signaling solution and multimedia conference models. The various issues highlighted in integration of WebRTC with SIP are interaction with SIP-Based Conferencing Systems, signaling solution, multimedia conference models and security issues. Integration of WebRTC with SIP creates a good platform for e –base learning via multimedia communication. Encryption standards have been developed for secure media communication. Signaling overheads assist developers in the choice of technologies and protocols. A multimedia server technology - Kurento could push current WebRTC capabilities beyond plain peer-to-peer communication. Abstract protocols were developed for conference creation. Multiplatform video conferencing systems based upon web browsers were analyzed. Several issues still to be addressed are testing of open source products for web based multimedia sessions, use of encryption for media and signaling, validation of use of Kurento for consolidation of WebRTC ecosystem. Simulation approach can be used for testing purpose by organizing the conditions similar to the real life situation.

Keywords— WebRTC; SIP; signaling; e –base learning

I. INTRODUCTION

Real time communication is an important part of modern convergence networks. Interactive voice communication, video conferencing, chat messaging are today very popular and widely used Internet Protocol (IP) network services and applications. The Session Initiation Protocol (SIP) is a crucial communication technology that brought new and standardized control mechanisms. This allows the transformation of IP networks into real multimedia communication platforms which are now able to provide real time communication and presence services. SIP allows to integrate and mix together different types of communication services (internet and telecomm, real time and interactive with data/non-interactive services) into a new kind of communication environment. The integration of SIP real time service with the web service can create a new communication environment which will enable to create a vast number of new multimedia communication services. The integration task requires deeper knowledge of system, protocol, proper programming tools, skills and need of special software. Web Real Time Communication (WebRTC) requires a kind of signaling mechanism. It also requires a support of protocols which are not yet widely embedded within present SIP clients. Thus a new type of integrated communication environment needs to be developed by extension of SIP platform with WebRTC functions for enabling the web based initiated multimedia sessions [1].

A. Web Real Time Communication (WebRTC)

WebRTC is a new web technology which allows browser and mobile applications with functionalities such as audio/video calling, chat, peer-to-peer (P2P) file sharing and all that without any additional third-party software or plugins [3].

Major components of the WebRTC API are as shown in figure 1:

- **MediaStream** - Allows a web browser to access the camera and microphone;
- **RTCPeerConnection** - Sets up audio or video calls;
- **RTCDataChannel** - Allows browsers to send data through peer-to-peer connections.

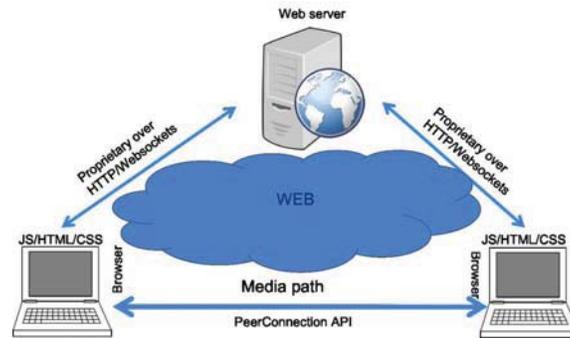


Fig. 1. WebRTC components

A typical WebRTC setup involves two browsers establishing a WebRTC call through a controlling Web Server and having the media streams between the browsers directly. The media streams have to be secured (achieved using DTLS-SRTP) and should traverse NAT/firewall (achieved using ICE negotiations and STUN/TURN mechanisms).

B. Session Initiation Protocol (SIP)

SIP is an IP based application protocol which provides signaling and control functionalities for a large range of multimedia communications including voice, data, images, messaging, presence, file transfers and etc. SIP defines some logical functional entities, which are categorized either as SIP endpoints or as SIP servers. SIP endpoints are User agents (UA), Back-to-back User Agent (B2BUA) or SIP gateways. SIP servers can have role of a Registrar server, Proxy server or Redirect server. From the protocol point of view SIP is the signaling protocol; from the implementation point of view is SIP the communication technology which is using and adopts wider range of protocols, technologies and security mechanisms. It mainly includes: real time protocol/real time control protocol (RTP/RTCP) for media transport and secure real time protocol (SRTP) for its media security; domain name system (DNS) for SIP addressing; SIP and session description protocol (SDP) for signaling and media capabilities handshaking, secured by using the datagram transport layer security/user datagram protocol (DTLS/UDP) or transport layer security/transmission control protocol (TLS/TCP) protocol stack, SIMPLE, MSRP and XCAP for instant messaging and presence ICE/STUN/TURN for NAT and firewall traversal. The generalized illustration of Network connection during a VoP call using a PBX is shown in figure 2.

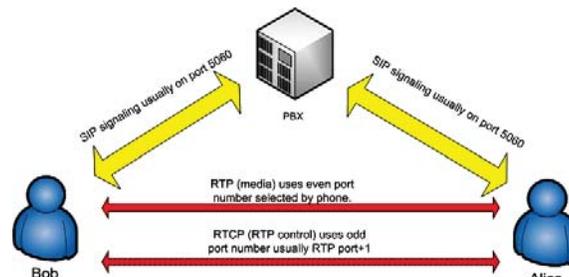


Fig. 2 Illustration of Network connection during a VoIP call using a PBX

C. WebSocket (WS)

WebSocket is a fast, web-based communication protocol which enables persistent and bidirectional TCP (or secured TLS) communication between a web client and a remote server.

II. LITERATURE SURVEY

The literature referred addresses the various issues in context of WebRTC such as interaction with SIP-Based Conferencing Systems, signaling solutions and multimedia conference models.

Segeč [1] has provided technological overview describing the aspect of integration of two important internet technologies WebRTC with SIP. These together are used to easily extend any web based e-learning or collaboration environment for unlimited communication experiences. These allow the teachers to communicate

with students through audio & video. The analysis of interworking issues identifies the system entities together with opportunities for deployment of open source product.

Amirante [2] has illustrated the concept of integrating interactive multimedia features into web applications. This has led to the creation of the World Wide Web Consortium (W3C) WebRTC and the Internet Engineering Task Force (IETF) RTCWEB working groups. Such groups are jointly defining both the application programming interfaces and the underlying communication protocols for the setup and management of a reliable communication path between any pair of next-generation web browsers. The major issue is concerned with co-existence of legacy SIP based systems with the upcoming browser enabled architectures. This issue is tackled by identification of interoperability requirement & real world example associated with allowing access to the Meetecho collaboration framework from WebRTC browsers is presented.

Some of the important projects currently undertaken are sipml5 project by Doubango Telecom, *Lynckia*, an MCU-based approach to WebRTC videoconferencing currently under development at the Universidad Politecnica de Madrid and Voice and Video on the Web (VVoW) at Illinois Institute of Technology.

Sredojev et al. [3] has described the WebRTC technology & implementation of WebRTC client server & signaling. Signaling methods and protocols are not specified by the WebRTC standards. A novel signaling mechanism has been designed & implemented. The corresponding message sequence chart of the WebRTC communication behavior describes a communication flow between peers and the server. The server application is implemented as a WebSocket server. The client application demonstrates the use of the WebRTC API(application peripheral interface) for achieving real-time communication. WebRTC technology reduces the costs of using VPN (Virtual Private Network) to connect remote branches and real-time communication is supported without the need for additional third-party software or plugins. Secure communication with employees within an enterprise and its remote branches is enabled by state of the art encryption standards.

Adeyeye et al [4] 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 signaling overhead introduced into a browser by each method was determined. The results showed WebRTC-SIP/WS has more overhead than WebRTC-JSON/XHR. These signaling 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.

Fernández et al [5] has introduced Kurento, a media server technology based on open source software capable of demonstrating how this convergence could take place by combining a SIP/HTTP based signaling plane and a powerful media server infrastructure built on top of the GStreamer software stack. The presented technology is suitable for sending and delivering real-time multimedia through different protocols and formats and capable of providing advanced processing capabilities, which include media mixing, transcoding and filtering. Thus, Kurento could push current WebRTC capabilities beyond plain peer-to-peer communication.

Elleuch [6] introduces two specific conference models VoIP and MMoIP(video communication service) adapted to support WebRTC communication between browsers for both small scale and large scale conference. Many conference models can support multi-party communication between endpoints over an IP networks. Otherwise, the use of WebRTC technology require direct P2P media negotiation between participants as well as signaling interaction with the Server to manage the conference. The proposed abstract protocols implements conference creation and browser joining/leaving and can be mapped using SIP/DSP primitives.

Vápeník et al. [7] has dealt the videoconferencing systems ran by web browsers where certain contemporary videoconferencing solutions and technologies used for multimedia streaming were analyzed. WebRTC technology terminology along with its benefits has been specified. Certain options for security enhancements for visual stream transmission via computer network were sketched. This system was built as a web videoconferencing system with security features. It is used to allow communication in four forms – video, audio, text chats and files sharing among users. In addition to the primary security control of information exchange and, of course, the secure transmission of video, audio and data, the system was extended to the two-level security videoconferencing virtual rooms. Unique token and conference password are the two levels of system security. The videoconferencing system has been tested on most operating systems - Windows 7, Windows 8, UNIX and Android. Thus, it is multiplatform and in addition is functional on multiple distributions of web browsers.

III. GAPS IN STUDY

Use of open source products for web based initiated multimedia sessions is to be tested in remotely based online learning & education system.

The integration of Opus codec Asterisk, in order to have it available in the list of supported codecs when negotiating a media session, needs extensive testing. Mechanism of handling ICE (Interactive Connectivity Establishment) in chrome, in contrast with less standard version, needs to be established. The support of the different WebRTC endpoint implementations, like Firefox Nightly, as well as browsers exploiting the webrtc4all (<http://code.google.com/p/webrtc4all/>) plugin to implement the required functionality, before it is natively supported and widespread, needs to be explored.

Secure communication with employees within an enterprise and its remote branches is enabled by state of the art encryption standards in WebRTC technology. An additional file sharing option needs to be developed and encryption will be used for both media and signaling.

The integration of various protocols in WebRTC video chat opens enormous opportunities for web as well as application developers. On one hand, web developers can develop websites and applications that would run in a browser using HTML5 with the APIs exposed to WebPages. On the other hand, application developers can develop applications that work with a browser internals (e.g. a XULRunner or Chrome Application), thereby directly communicating with the underlying protocols and mechanisms in that browser. Thus, extending the SIP Servlet application to JAIN-SLEE and comparing latency in setting up a two-way communication between an IMS-based WebRTC and the regular (non IMS-based) WebRTC is required.

Contribution of Kurento to the consolidation of the WebRTC ecosystem by showing a pathway towards more advanced and universal real-time communication services, needs to be validated.

IV. RESEARCH APPROACH

There are two basic approaches to research, viz. quantitative approach and the qualitative approach. Quantitative approach involves the generation of data in quantitative form which can be subjected to rigorous quantitative analysis in a formal and rigid fashion. This approach can be further sub-classified into inferential, experimental and simulation approaches to research.

1. Inferential approach focuses on forming a database from which characteristics or relationships of population can be inferred from a sample.
2. Experimental approach is characterized by the greater control over the research environment and in this case some variables are manipulated to observe their effect on other variables.
3. Simulation approach involves the construction of an artificial environment within which relevant information and data can be generated. This permits an observation of dynamic behavior of a system or its subsystem under controlled conditions.

Qualitative approach to research is concerned with subjective assessment of attitudes, opinions and behavior. Simulation approach can be used for testing purpose by organizing the conditions similar to the real life situation

V. CONCLUSION

Current literature has highlighted the various issues in integration of WebRTC with SIP such as interaction with SIP-Based Conferencing Systems, signaling solution, multimedia conference models and security issues. Integration of WebRTC with SIP creates a good platform for e -base learning via multimedia communication. Encryption standards have been developed for secure media communication. Signaling overheads assist developers in the choice of technologies and protocols. A multimedia server technology - Kurento could push current WebRTC capabilities beyond plain peer-to-peer communication. Abstract protocols were developed for conference creation. Multiplatform video conferencing systems based upon web browsers were analyzed. Several issues still to be addressed are testing of open source products for web based multimedia sessions, use of encryption for media and signaling, validation of use of Kurento for consolidation of WebRTC ecosystem. Simulation approach can be used for testing purpose by organizing the conditions similar to the real life situation.

REFERENCES

- [1] P. Segeč, P. Palúch, J. Papán, M. Kubina, "The integration of WebRTC and SIP: way of enhancing real-time, interactive multimedia communication", ICETA 2014 • 12th IEEE International Conference on Emerging eLearning Technologies and Applications, Starý Smokovec, Slovakia, , pp 437-442, December 2014.

- [2] A. Amirante, Meetecho S.r.l., T. Castaldi, L. Miniario, and S. Pietro Romano, "On the Seamless Interaction between WebRTC Browsers and SIP-Based Conferencing Systems", WEB-BASED COMMUNICATIONS, IEEE Communications Magazine, pp 42-47, April 2013.
- [3] B. Sredojev, D. [Samardzija](#), D. [Posarac](#), "WebRTC technology overview and signaling solution design and implementation", MIPRO 2015, Opatija, Croatia, pp 1006-1009, May 2015.
- [4] M. Adeyeye, I. Makitta, T. Fogwill, "Determining the Signalling Overhead of two common WebRTC methods: JSON via XMLHttpRequest and SIP over WebSocket", IEEE Explore Digital Library, 978-1-4673-5943-6/13, 2013.
- [5] L. López Fernández, M. [Paris Diaz](#), R. [Benitez Mejias](#), F. J. [Lopez](#), "Kurento: a media server technology for convergent WWW/mobile real-time multimedia communications supporting WebRTC", IEEE Explore Digital Library, 978-1-4673-5828-6/13, 2013.
- [6] Wajdi Elleuch, "Models for Multimedia Conference between Browsers based on WebRTC, Sixth International Workshop on Selected Topics in Mobile and Wireless Computing", pp 279 -284, 2013.
- [7] R. Vápeník, M. [Michalko](#), J. [Janitor](#), F. [Jakab](#), "Secured Web Oriented Video conferencing System For Educational Purposes Using WebRTC Technology", ICETA 2014, 12th IEEE International Conference on Emerging eLearning Technologies and Applications, Starý Smokovec, Slovakia, pp 495 -500, Dec 2014.