

Live Streaming of Audio and Video through the Browser in an Enterprise

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Abstract

Media streaming services generate a large portion of the Internet traffic every day with the biggest one being video on demand services. This thesis will focus on the use of streaming media in a collaborative environment within an enterprise setting. There are currently many such services existing today. A lot of these services are costly and use old legacy systems and are therefore not easy to upgrade to adopt the benefits of the latest technologies.

The aim of this thesis is to find the most suitable way for enterprises to implement live media sharing in the web browser. The goal is to detect different problems. In particular this thesis will look into the difficulties of streaming live audio with high quality over the different protocols.

The HTML5 standard introduces many new APIs that give web browsers the ability to communicate directly with each other in real-time. But there are many solutions and the different providers doesn't seem to agree on a single solution. This thesis will examine using these new APIs to see how we can integrate the browser in current media collaboration systems. The goal being to determine the feasibility of using these new APIs and evaluate their usage. This thesis hope to aid in developing solutions for media sharing in the browser that can be used in an enterprise setting.

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Acronyms

Chapter 1

Introduction

Chapter 2

Background Research

Chapter 3

Industrial Case

Chapter 4

Problem Description

As the use of mobile devices increase in business use, Visual Solutions wants to explore the possibility of developing collaboration tools that run directly in the browser using the new API¹'s drafted by the W3C². This would allow their solutions to extend to more platforms. This chapter aims to explain the problems and what can be done to overcome them.

As described in the previous chapter 3. Visual Solutions has chosen to look at the new Web API's that are under development for doing collaboration services. While there long have been Flash, and other services like Microsoft Silverlight's Smooth Streaming and Apple's HTTP³ Live Streaming. WebRTC⁴ seems to have some important advantages over the other technologies.

- Universal mobile support
- Easy integration, with a high focus on security.
- No need to download clients or plugins.

By looking over the drafts done by W3C, WebRTC seems to be well suited for doing collaboration directly in the web browser.

4.1 Problem statement

WebRTC seems to be the best solution for doing video conferencing on the web, and both desktop and mobile web browsers have some degree of support which improves on a nightly basis.

¹Application Programming Interface: Specifies how some software components should interact with each other

²World Wide Web Consortium:

³Hypertext Transfer Protocol:

⁴Web Real-Time Communication: <http://www.w3.org/TR/webrtc/>

While other solutions like Flash may have better support at the moment, there is a clear indication that the web is moving towards open standards. And a lot of effort is being put into WebRTC at the moment by the big players like Google and Firefox to make this the new standard, however there are still disagreements about which audio/video codecs is going to be used as a standard.

There has been done a lot of functional testing to see if the experiments work. What kind of features and codecs work and what does not. This is done because all the browsers that support WebRTC only support different subsets of the features specified by the W3C.

Find out if WebRTC is a mature and usable technology to be used in a business environment.

Figure out what will happen in the near future with the support for WebRTC on different platforms?

4.2 Research question

This thesis will try to answer the following question:

How can Visual Solutions best integrate WebRTC with their current application Virtual Arena for doing collaboration?

Supporting questions:

- What is the current support for WebRTC in the different browsers?
- How well does WebRTC connections scale with more MediaStreams and peers, and if this can be improved using other models than the standard P2P model
- How will the support for this technology improve in the near future?

4.3 Research method

The research method used in this thesis, is a quantitative experimental method, using systematic investigation and analysis of the different implementations in browsers of the drafted API's by the W3C. Doing experiments to examine the actual workings of the implementations to uncover support for the features.

Practical experiments consists of test cases that was run in Chrome, Firefox, and Chrome for Android. These experiments were performed because there is a lot of incomplete information on the current support.

Many smaller experiments were conducted in the beginning to gain knowledge of the workings of WebRTC. Lots of articles, blogs, books and papers were read.

Looking at trends in the development, this thesis will also be predicting support for the API's in the near future.

4.4 Existing work

There exists a lot of tutorials on creating very simple implementations of WebRTC on the client side, usually setting up one-to-one connections using a third-party service such as Pusher or OpenTOK for doing signaling. More advanced guides show how you can setup a Node.js server to do signaling using WebSockets.

This is great, but to be able to do what we need, we need to create an advanced server that acts like a client. While most commercial solutions that could potentially be of some help most are closed. But there is one open source project called Licode which utilizes the WebRTC's native libraries and is compatible with the standard and protocols. It's written in C++ and is acts like a MCU⁵. It's open source, but not very well documented. It also doesn't offer what we need in terms of including external RTP⁶ streams, but it says in their roadmap that this is planned for the future.

There is no good service on the web that provides live data on how far the support has come on different browsers. There is one service that will check for WebRTC and WebSockets connectivity on <http://www.check-connectivity.com/>. You will find blogs and articles that provide tables showing who has implemented which APIs, but most will be outdated. The best solution is to test yourself using your own code.

If WebRTC is not supported in a user's browser, there may be possible for the user to either update their browser or adjust a flag in the browser's experimental settings.

4.5 Expected results

- From this thesis we want to have provided a solution to how Visual Solutions can integrate WebRTC with Virtual Arena. We want to provide information about all the necessary technologies that need to be implemented.
- We would also like to map the current support for WebRTC on different browsers.

⁵Media Control Unit:

⁶Real Time Protocol: defines a standard packet format for delivering audio and video over IP networks.

- Also we would like to look at how well WebRTC connections scales with more MediaStreams and peers, and if this can be improved using other models than the standard P2P model.

Chapter 5

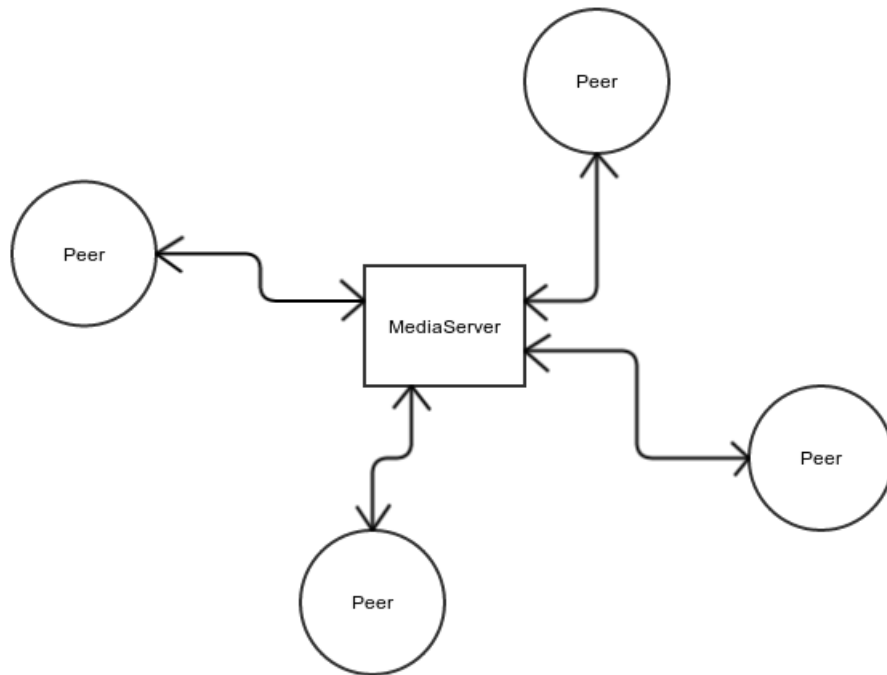
WebRTC and Virtual Arena

5.1 How can we integrate RTC in the Web with Virtual Arena?

This chapter is about the challenges of making two different media systems work together. It will discuss a possible solution and the pros and cons of this. Then we will look at how Visual Solutions can best integrate this solution and how their vision can be solved.

5.1.1 How does Virtual Arena work?

Virtual Arena supports one-to-one, one-to-many and many-to-many collaborative scenarios. The application uses a MCU that acts as a media server. With this server Virtual Arena can support a lot more incoming and outgoing streams than in a simple peer-to-peer scenario. It also applies mitigation strategies for scenarios with limited bandwidth. Without going into too much detail of how the application is actually put together the main setup looks something like this:



Communication between peers and the media server is pure RTP streams with all the information that is needed provided in the header of the packets being sent. The most important piece of information is the SSRC¹ with this, the media server can identify the incoming stream and mix it in with the other streams and forward it to all the other peers.

So basically all we need is to be able to listen for incoming RTP streams on the receiving end. And the other way we have to deliver pure RTP streams with the correct header information included, most importantly the SSRC.

5.1.2 How does WebRTC work?

WebRTC is a very complex synergy of components and protocols. But from a frontend developers point of view, all of this is packaged into three main Javascript API's: `getUserMedia`, `RTCPeerConnection` and `RTCDataChannel`. These are defined by the W3C.

The exchange of real-time media between two browsers follows a process like this:

1 Input devices are opened for capture as the media source. This is done using the `getUserMedia` API.

¹Secure Real-time Transport Protocol:

2 Now we have to signal the other users that we want to connect to them. using `RTCPeerConnection` we send an SDP² offer to the other clients, which generates an SDP Answer. The SDP here includes ICE³ candidates. Which opens ports in the firewall. There is a fallback if both clients are on symmetric NAT⁴'s and a connection isn't possible to use a TURN⁵ server that acts like a packet mirror, channeling all the packets through the TURN server.

3 Once connection is successful, a DTLS⁶ connection is opened and all the media from input devices are encoded into packets and transmitted using SRTP⁷-DTLS streams.

4 At the destination, the packets are decoded and formatted into a `MediaStream`.

5 The `MediaStream` is sent to output devices

5.1.3 What are the differences?

While Virtual Arena provides a very simple and not so secure solution media level. This because in a closed enterprised environemnt this is not needed. Most of the security will lie in the firewall anyways.

But since WebRTC is a very open solution and is supposed to work with public users firwall configurations, a lot more complex security architecture is needed.

So while WebRTC packages all their streams in a new format called a `MediaStream`. At the transport level everything is encrypted using DTLS on top of SRTP.

5.1.4 What do we need?

There are several problems here. Since Virtual Arena operates on a low-level not much is needed. All we need is to listen to a specific ip-address on a specific port on UDP transmitted data. In the packet headers we will find an SSRC that we use to identify the incoming packets.

With WebRTC on the other hand, we are only allowed to work on a higher level. We cannot access `MediaStreams` directly without first opening a connection with another peer. This is a problem. This means that we first

²Session Description Protocol:

³Interactive Connectivity Establishment:

⁴Network Address Translator:

⁵Traversal Using Relays around NAT:

⁶Datagram Transport Layer Security:

⁷Secure Real-time Protocol:

have to initiate a normal RTC Connection before we can look at any identifiers such as the SSRC. We cannot listen on specific ports for incoming RTC connections, and we cannot specify a specific port to transmit data. Also as of now the current implementation of WebRTC is not designed to add local external streams to the connection. This means that getting WebRTC to work with any external application, we have to create some sort of gateway server.

By having done tons of experiments I propose this solution:

One gateway server that can listen to UDP-packets from the Visual Solutions MCU and acts as a peer in WebRTC. This is not an ideal approach, but the most viable at the moment. Ideally one would integrate WebRTC over the whole spectrum, which is both inconvenient and leads to complications.

For a single person that acts as a PeerConnection in WebRTC to listen in on a conversation from Virtual Arena, one would need to initiate a fake RTC Connection using two peers, then create a MediaStream from the incoming UDP packets and inject that MediaStream into the RTC Connection. On the returning side one would have to take the MediaStreams and break them into pure RTC packets with an SSRC identifier in the header and send them in return to the MCU.

First problem is receiving the incoming conversation from Virtual Arena this can be done setting up a socket that listens for incoming UDP packets. This is not a problem and can even be done in a pure Javascript environment using node.js

Then one would have to create a MediaStream Object from these packets. This is not currently possible using chrome API's or Firefox. In chrome it is possible to create a pure audio MediaStream using the new WebAudio API, and in Firefox it should be possible to create a MediaStream from video using `getStreamUntilEnded()`, but this is currently broken. In the future however this should be possible using the drafted `captureMedia` API.

It is however possible to inject external MediaStreams into an RTC Connection.

For returning data from a PeerConnection to the MCU one would have to record the stream and return the data over an UDP⁸ connection. This should be done using the proposed MediaStream Recording APIs, but none of the browsers have currently implemented these yet.

⁸User Datagram Protocol:

Chapter 6

What is the current state of support of Real-time Communication Between Browsers

Chapter 7

How well does Real-time
Communication scale on
mobile devices?

Chapter 8

What will happen with the support of Real-time Communications in the near future?

Chapter 9

HTML5

Chapter 10

The Approach(experiments)

Chapter 11

The Implementation(how implemented approach)

Chapter 12

Evaluation

Chapter 13

Conclusion

summarize your thesis again as in the introduction. Describe how your evaluation revealed that your system is successful. Describe future work in this area.

We have looked at some basic theory about streaming media in the web browser, and explained our experiments and their capabilities. In chapter x we looked at the testing data collected and did an analysis of the results. The results were discussed and conclusions were made. This chapter will summarize the key elements of the results, look at what contributions have been made, and suggest some future work.

To simulate a low latency environment all early experiments were done on a local network. After collecting data using different mechanisms of protocols and streaming media, they were compared with the results from the same experiments using a simulated external network. Data about traffic, latency, and cpu-load were gathered and analyzed. Then we discussed the results with regard to inconsistencies, trends, synergies between different factors, and potential conclusions that could be made.

Contributions to reseach

Future Work Furter testing when different vendors have agreed on a common platform

Appendix A

Appendix Title