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Performance analysis of topologies for Web-based Real-Time Communication (WebRTC)

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AALTO UNIVERSITY SCHOOL OF ELECTRICAL ENGINEERING	ABSTRACT OF THE MASTER'S THESIS

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Real-time Communications over the Web (WebRTC) is being developed to be the next big improvement for rich web applications. This enabler allow developers to implement real-time data transfer between browsers by using high level Application Programing Interfaces (APIs). Running real-time applications in browser may led to a totally new scenario regarding usability and performance. Congestion control mechanisms may influence the way this data is sent and metrics such as delay, bit rate and loss are now crucial for browsers. Some mechanisms that have been used priorly in other technologies are implemented in those browsers to handle the internals of WebRTC adding complexity to the system. This new scenario requires a deep study regarding the ability of browsers to adapt to those requirements and to fulfill all the features that are enabled.

We investigate how WebRTC performs in a real environment running over an actual web application. The capacity of the internal mechanisms to adapt to the variable conditions of the path, consumption resources and rate. Taking those principles, we test a range of topologies and use cases that can be implemented with the actual version of WebRTC. Considering this scenario we divide the metrics in two categories, host and network indicators. We compare the results of those tests with the expected output based on the defined protocol in order to evaluate the ability to perform real-time media communication over the browser.

Keywords: Internet, WebRTC, Real-time Communication, Network Topologies, Performance Analysis, Congestion Control, Browsers, HTML5

Preface

Thank you everybody.

Otaniemi, 9.3.2012

Albert Abelló Lozano

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Abbreviatures

AEC	Acoustic Echo Canceler
AMS	Adobe Media Server
API	Application Programming Interface
AVPF	Audio Video Profile with Feedback
DNS	Domain Name System
DOM	Document Object Model
DoS	Denial of Service
DTLS	Datagram Transport Layer Security
FEC	Forward Error Correction
FPS	Frames per Second
HTML	HyperText Markup Language
HTTPS	Hypertext Transfer Protocol Secure
IAT	Inter-Arrival Time
ICE	Interactive Connectivity Establishment
IETF	Internet Engineering Task Force
IM	Instant Messaging
IRC	Internet Relay Chat
ITU	International Telecommunication Union
JSEP	JavaScript Session Establishment Protocol
JSFL	JavaScript Flash Language
JSON	JavaScript Object Notation
LAN	Local Area Networks
MCU	Multipoint Control Unit
MEGACO	Media Gateway Control Protocol
NAT	Network Address Translation
NR	Noise Reduction
NTPd	Network Time Protocol daemon
OWD	One-Way Delay
QoS	Quality of Service
REMB	Receiver Estimated Maximum Bitrate
RFC	Request for Comments
RIA	Rich Internet Application
RTC	Real Time Communication
RTCP	Real Time Control Protocol
RTMFP	Real Time Media Flow Protocol
RTP	Real-time Transport Protocol
RTT	Round-Trip Time
SCTP	System Control Transmission Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRTP	Secure Real-time Transport Protocol
SSRC	Synchronization Source identifier
STUN	Simple Transversal Utilities for NAT

TFRC	TCP Friendly Rate Control
TURN	Traversal Using Relays around NAT
UDP	User Datagram Protocol
VoIP	Voice over IP
VVoIP	Video and Voice over IP
W3C	World Wide Web Consortium
WebRTC	Real Time Communications for the Web
WHATWG	Web HyperText Application Technology Working Group
WWW	World Wide Web
XSS	Cross-site scripting

1 Introduction

Video communication is changing rapidly, the dramatical increase of internet communication between people is forcing technology to support mobile and real-time video experiences in different ways. The existence of reliable, low cost and simple platforms for real-time communication is becoming an essential part for the future of consumer behavior, business structure and innovation.

Technology and media are changing the way people interact with each other, communication over the internet is encouraging users to interact, talk and see content in a cost-effective and reliable way. This way of communication is adding new features to services that have never been available before, they are now able to have high-engagement communication, where richer, more intimate communication is possible. At the same time, this is changing traditions and habits of communication between people and transforming personal relationships. Distances are now shorter, bringing individuals and groups together around the world, allowing people to connect with friends and meet new people in different ways.

Furthermore, it also helps businesses to have lightweight communication alternatives that will increase their efficiency besides the size of the company. Thus, encouraging front end designers designers and device developers to turn their products into multi-functional and inter-operable communication devices.

Video has been available in the World Wide Web (WWW) since the 1990s, it has evolved to be less CPU consuming and has adapted to the new link rates while affordable digital and video cameras have become a must have feature in nowadays computers. Those two enablers along with the increased demand for richer applications for the WWW are some of the reasons behind Real Time Communications for the Web (WebRTC).

WebRTC is a suite of tools that enable human communications via voice and audio where Real Time Communication (RTC) should be as natural in a web application as browsing images or visiting websites. With this simple approach, WebRTC aims to transform something that has been traditionally complex and expensive to an open application that can be used by everybody, enabling this RTC technology in all existing web applications and giving the developers the ability to innovate and allow rich user interaction in their products.

Many web services already use RTC technology to allow communication (e.g Google Hangouts) and most of them require the user to download native apps or plugins to make it work. With WebRTC, real-time communications between users should be transparent for them, since downloading, installing and using plugins can be complex and tedious. On the other side, the usage of plugins is also complicated from the development point of view and restricts the ability of developers to come out with great features that can enrich the communication between people.

WebRTC project major guidelines are based on working Application Programming Interfaces (APIs) that are open source, free and standardized.

1.1 Background

WebRTC is an effort to bring standard APIs to JavaScript developers, allowing them to code RTC functionalities into web applications. This is commonly seen as a call system over the web or video calling applications.

Web application APIs are defined into the HyperText Markup Language (HTML) version 5 and help developers to add features to their web applications with minimal effort using JavaScript functions. APIs can be defined as a collection of methods and callbacks that help developers to access available technologies in the browser, they are used in web development to access the full potential of browsers and compute some part of the dynamic web applications on the client side. WebRTC API is being drafted by the World Wide Web Consortium (W3C) alongside with the Internet Engineering Task Force (IETF) . This API has been iterated through different versions to increase its usability thanks to the feedback given by web developers.

The first W3C announcement of WebRTC was done in a working group in May 2011 [?], and the official mailing list started in April 2011 [?]. During the first stage of discussion, the main goal was to define a public draft for the version 1 of the API implementation and a timeline with the goal to release the first final version of WebRTC. The W3C public draft of WebRTC was published the 27th of October 2011 [?]. During this first W3C draft, only media (audio and video) could be sent over the network to other peers, it defined the way browsers would be able to access media devices without the use of any plugin or external software.

WebRTC project also joined the IETF with a working group in May 2011 [?]. The initial milestones of the IETF initially marked December 2011 as the deadline to provide the information and elements required for the W3C to design the first version of the API. On the other side, the main goals of the IETF working group covered the definition of the communication model, session management, security, NAT traversal solution, media formats, codec agreement and data transport [?]. In June 2011 Google publicly released the source code of their WebRTC API implementation [?].

WebRTC APIs are integrated within the browser and accessible using JavaScript in conjunction with the Document Object Model (DOM) interfaces. Some of the APIs that have been developed for WebRTC are not part of the HTML5 W3C specification but are included into the Web HyperText Application Technology Working Group (WHATWG) HTML specification.

WebRTC APIs work combining two different coding languages, HTML and JavaScript, HTML is the de facto format for serving web applications and JavaScript is becoming the most popular scripting system for web clients to allow users to dynamically interact with the web application. The actual version of WebRTC is a merge between different API that integrate with each other to provide flexible RTC in the browser. The final goal is to allow developers to create plugin-free real-time web applications that can be cross compatible with different browser vendors and operating systems.

1.2 Challenges

WebRTC is a suite of protocols that share the available device resources with many other applications. Due to the short experience in WebRTC environments sharing the available resources, we might find some lack of documentation or previous literature regarding congestion analysis compared with other technologies.

The aim to test and help to develop new protocols such as WebRTC is unfortunately accompanied by a lack of information that may affect some of the statements made in this thesis. Hopefully, this should not affect its development neither its conclusions.

Considering the fact that WebRTC is still being developed at the moment of writing this thesis, some of the statements made here might be different in the upcoming versions of WebRTC, meaning that some of the analyzed issues could have been solved.

General WebRTC challenges are related to two different issues: technical problems and political decisions. Firstly, congestion mechanisms for RTC have always been complicated to implement due to the need of a fast response against path disturbances and link conditions. In the course of this thesis we might find limitations in WebRTC when having constrained links.

Network Address Translation (NAT) may also arise as a problem, succeeding when setting up a communication path through restrictive environments is crucial in RTC protocols.

WebRTC is being standardized in different W3C and IETF working groups, meaning that there is a common interest to reach consensus in all aspects of this protocol. This is positive as it is supposed to adapt the specifications to all possible needs but it also delays some decisions that might affect the development of the APIs.

During the development of the thesis we focus in the technical challenges of the protocol.

1.3 Contribution

Investigate how WebRTC performs in a real environment trying to evaluate the best way to set multiple peer connections that handle media in different network topologies. Measure the performance of WebRTC in a real environment, identifying bottlenecks related to encoding/decoding, media establishment or connection maintenance. All this should be performed in real-time over a browser by using the already existing WebRTC API.

By using metrics related to RTC protocols we expect to understand the way WebRTC performs when handling in different environments.

1.4 Goals

WebRTC uses and adapts some existing technologies for real-time communication. This thesis focuses in studying:

- WebRTC performance in different topologies and environments using real sources of video and audio that are encoded with the codec provided by the browser.
- Usage of WebRTC to build a real application that can be used by users proving that the API is ready to be deployed as well as it is a good approach for the developer needs when building real-time applications over the web. This is done in conjunction with other new APIs and technologies introduced with HTML5.
- Testing of different WebRTC topologies with different network constraints to observe the response of the actual existing API.

The final conclusion covers an overall analysis and usage experience of WebRTC, providing some valuable feedback for further modifications on the existing API.

1.5 Structure

Not sure about here

2 Real-time Communication

Real-time Communication can be defined as any method of communication where users can exchange information and media with low latency, real-time aspect of it can be also defined as live. The purpose of RTC is widely seen as a way to intercommunicate between people or software. This can be done in a two-way scenario where data is transmitted between both sides, being both users receivers and senders, or in a one-way configuration with one unique source of data and one or multiple receivers. In the first configuration, latency is very important in order to achieve good quality communication between both users whereas the second scenario can tolerate some latency in the link but data transmission must be constant. In two-way communication data can be transmitted using multiple technologies, the topologies used can be either peer-to-peer or using a centralized relay. Some other ways of transmitting data include multicast or broadcast.

In broadcast and multicast mode data is transferred to multiple peers in a network but does not require real time response in most cases.

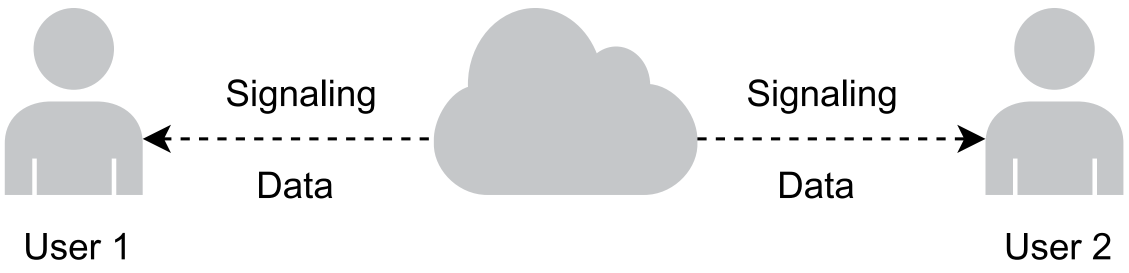


Figure 1: Real time communication between two users over the Internet.

Figure **??** describes an RTC scenario for two users, the technology providing the communication may differ in each situation but the goal is always the same. RTC has a common characteristic that is always common in all technologies, there must be a signaling or agreement between the two entities, either with the central node or with the other user. This procedure is used by the protocols to check the capabilities of the two entities before proceeding to send the media. In the signaling part, codec agreement and keep-alive methods are decided at the same time as all the multiple features that will be enabled in the new session, making it crucial to configure the media and data to be transmitted.

On the other hand, once signaling is done data flow can be sent to the receiver, this flow may include media (audio or video) and data. The transmission may require also some extra signaling messages to be exchanged in order to maintain the link or adapt the constraints to the actual network conditions, at the same time, features might change during the session.

RTC scenarios can be given either over the Internet or using traditional techniques, some real-time scenarios are: telephony, mobile phone communication, radio, instant messaging (IM) , Voice over IP (VoIP) , Video and Voice over IP (VVoIP) , Internet Relay Chat (IRC) and videoconferencing.

All the previous ways of communication work in real time and rely in Figure ?? topology but with different technologies. In this thesis we manly work with Internet RTC using media and data.

2.1 Session Initiation Protocol (SIP)

SIP allows communication between two different users with audio/video support in real-time. SIP final Request for Comments (RFC) was published in June 2004, this document describes the original functionalities and mechanisms of SIP [?]. From an overview perspective, SIP is an application-layer control protocol for multimedia sessions which can establish, maintain and terminate media sessions. During the development of the standard, different new functionalities were added to the drafts such as conferencing and the possibility of adding/removing media from existing sessions.

This protocol works alongside with other existing technologies such as Real-time Transport Protocol (RTP) , Session Description Protocol (SDP) and Media Gateway Control Protocol (MEGACO) . Using SDP for the session negotiation between the end-points and RTP for the media transport, all these protocols are widely used in other technologies and usually provide legacy for older devices and outdated versions. Meanwhile SIP can locate and deliver a message to a user, SDP can provide the required information for the session establishment and RTP can transport the data.

SIP architecture relies in a trapezoid form where the Domain Name System (DNS) is used to locate the other peers of the system. Once that peer is located and session is negotiated, media flows peer-to-peer to the endpoint. In order to build this system different agents are needed, SIP Proxies, SIP Redirect and SIP Registrar. SIP Proxies transmit the SDP and SIP messages from one peer to the other to establish the signaling (Figure ??). SIP Registrar are the machines that collect and save all the user information from the end points.

DNS provides the IP address for both proxy servers and allows the messages to be exchanged between both peers, SIP uses the following three-way handshake: INVITE, 200OK and ACK. Those messages carry the SDP data inside in an object format, when ray@upc.cat receives the INVITE message from bob@aalto.fi builds the 200OK response carrying the SDP object that provides compatibility check between both peers and which options and codecs to use. SIP provides some more messages to update the already existing session or to close it. The media transport is done using RTP and RTCP over User Datagram Protocol (UDP) [?].

SIP is a pure VoIP confederated technology that helped the community to learn about real-time P2P communication.

2.2 Real Time Media Flow Protocol (RTMFP) and Adobe Flash

RTMFP and Adobe Flash are proprietary technologies provided by Adobe, both services work together to deliver multimedia and RTC between users.

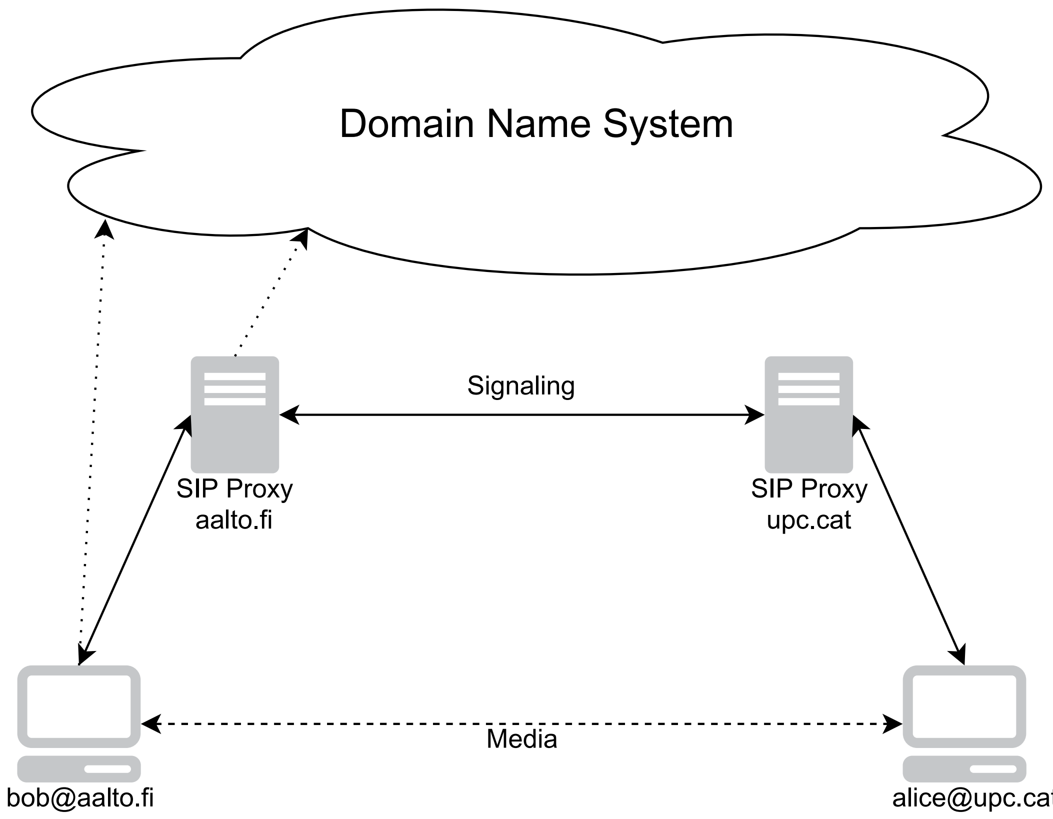


Figure 2: SIP architecture for end-to-end signaling.

Adobe Flash is a media software that uses a plugin to work on top of the browser, it is used to build multimedia experiences for end users such as graphics, animation, games and Rich Internet Applications (RIA) . It is widely used to stream video or audio in web applications, in order to enable this content we need to install Adobe Flash plugin in our computer. It also uses a different programming language that do not comply with any standards called JavaScript Flash Language (JSFL) and ActionScript. RTMFP and Adobe Flash require a plugin to work with any device, this obliges the user to install extra software that is not included in the browser, these two technologies are not standardized and are difficult to enable in some mobile devices. Adobe Flash Player is available in most platforms, except iOS devices, and is present in about 98% of all internet-enabled desktop devices. This plugin allows developers to access media streams from external devices such as cameras and microphones to be used along with RTMFP.

RTMFP uses Adobe Flash to provide media and data transfer between two end points. This system works over UDP [?]. RTMFP provides a full suite of methods and functions that allow the browser to access the necessary mechanisms to run real-time media communication, those methods are included into the plugin that must be installed prior usage. RTMFP is a private and licensed protocol. It also handles congestion control on the packets and NAT transversal issues. One of the biggest differences is that, compared with SIP, RTMFP does not provide interdomain connectivity and both peers must be in the same working domain to be able to communicate. This protocol is implemented by using Flash Player, Adobe Integrated Runtime (AIR) and Adobe Media Server (AMS) [?].

Media transfer in this protocol is encrypted, this issue has been addressed clearly in RTMFP by using proprietary algorithms and different encryption methods. The RTMFP architecture is similar to WebRTC concept, it also allows reconnection in case of connectivity issues and works by multiplexing different media streams over the same media channel when handling conferences or multiple streams. For the signaling part, Adobe uses a service called Cirrus (Figure ??), this service allows architectures such as: end-to-end, many-to-many and multicast [?].

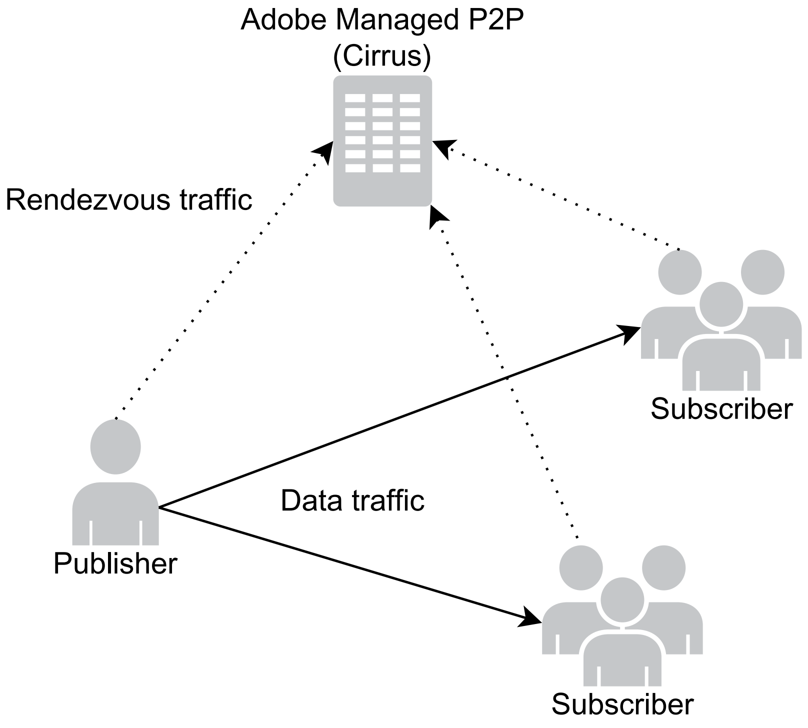


Figure 3: RTMFP architecture using Cirrus.

Some of the most valuable features is the possibility to easily integrate P2P multicast topologies where one source sends a video to a group of receivers.

2.3 WebRTC

WebRTC is part of the HTML5 proposal, it is defined in a W3C draft [?], and enables RTC capabilities between Internet browsers using simple JavaScript APIs. Providing video, audio and data P2P without any plugins. This API replaces the need of any plugin for P2P communications in browsers, WebRTC uses already existing standardized protocols, inherited from SIP, to perform RTC. The project was open sourced by Google to keep working with the IETF in order to standardize the technology [?].

WebRTC provides interoperability between different browser vendors, this allow the APIs to be accessible by the developers assuring high degree of compatibility (Figure **??**). Some of the major browsers that actively implement WebRTC APIs are: Google Chrome, Mozilla Firefox and Opera.

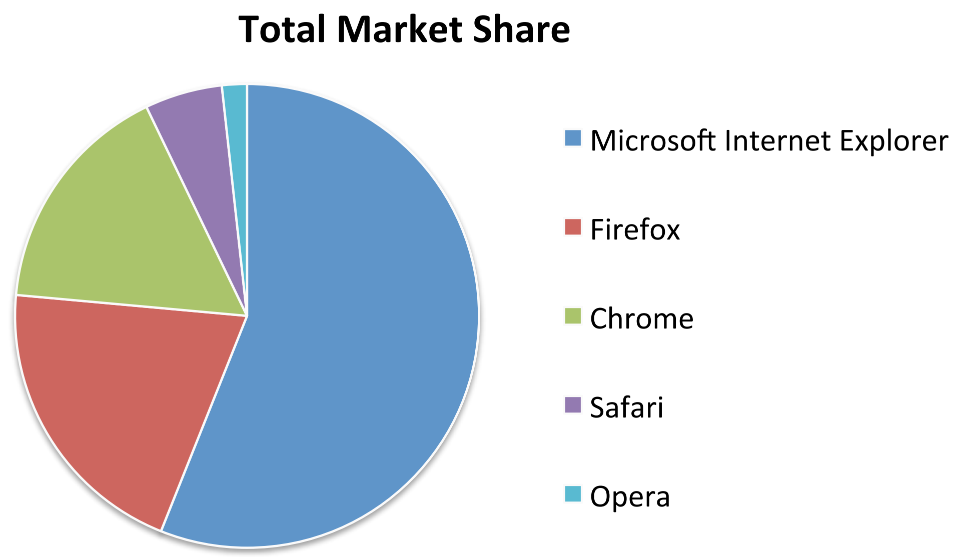


Figure 4: Market share of browser vendors by April 2013 [**?**].

With WebRTC, developers can provide applications for nearly half of the desktop devices available, mobile devices will integrate WebRTC as part of their HTML5 package to also enable RTC soon [**?**].

WebRTC is composed by two important APIs that enable real-time features, GetUserMedia and PeerConnection. Both of them are accessible by JavaScript on the browser. WebRTC APIs rely on the top of the *WebKit* rendering engine in Chrome and Opera, in April 2013 Google announced that is going to stop using *WebKit* as the rendering engine that is behind displaying web pages in Chrome. Instead, it's creating its own rendering engine named *Blink* [**?**].

2.3.1 Device Access API

WebRTC uses an API called *GetUserMedia* to access media streams from local devices (video cameras and microphones). This API itself does not provide RTC, furthermore, provides the media to be used as simple HTML elements in any web application. *GetUserMedia* allows developers to access local media devices using JavaScript code and generates media streams to be used either with the rest of the WebRTC APIs or with the HTML5 video element [**?**]. *GetUserMedia* is already interoperable between Google Chrome, Firefox and Opera [**?**].

GetUserMedia proposal was first attached directly to the WebRTC working group but has been published in a different draft, the usage of this API removes the need of using Adobe Flash to access the media device and also the plugin requirement.

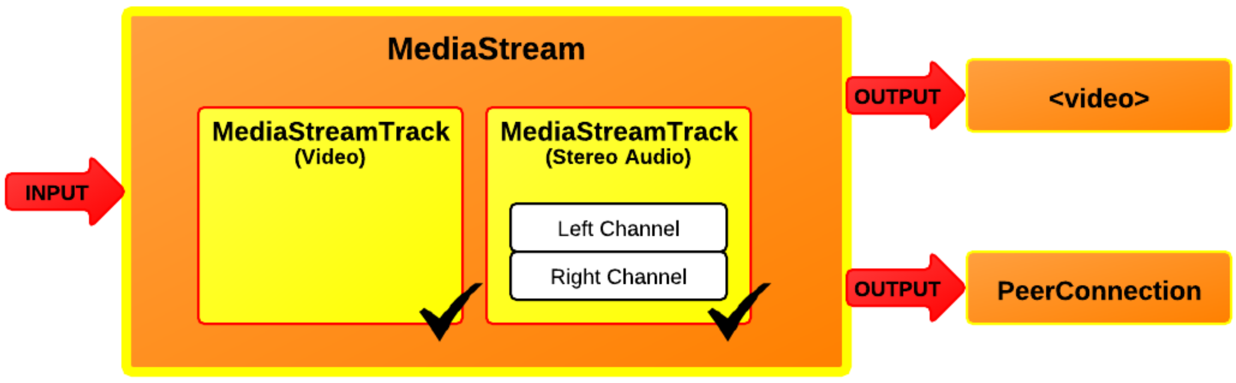


Figure 5: Media Stream API. Source [?].

Figure ?? illustrates how the browser access the media and delivers the output to JavaScript. We use this function to build WebRTC enabled applications for RTC video conferencing. The video tag is an HTML5 a Document Object Model (DOM) element that reproduces local and remote media streams.

MediaStreams are objects returned by the *GetUserMedia* API, those objects are composed by *MediaStreamTracks* that carry the actual data for the media. The goal of using this architecture is to be able, in a near future, to include multiple sources of video and audio over the same stream. *MediaStreams* handle the synchronization between all the *MediaStreamTracks* included for proper playback in the application level.

GetUserMedia API works using a fallback model, this means than the JavaScript method returns an object that can be played in an HTML web application, a simple example of this method can be seen in the following code.

Listing 1: Simple example of video and audio access using JavaScript

```
navigator.webkitGetUserMedia(cameraConstraints(), gotStream, function() {
        console.log("GetUserMedia failed");
});

function gotStream(stream) {
        //Stream is the MediaStream object returned by the API
        console.log("GetUserMedia succeeded");
        document.getElementById("local-video").src =
            webkitURL.createObjectURL(stream);
}
```

With the previous code, we are using the video and audio media from our devices to be played in an video HTML element identified as *local-video*.

GetUserMedia also allow developers to set some specific constraints to the media acquisition. This help applications to better adapt the stream to their requirements, those *cameraConstraints()* are stored into a JavaScript Object Notation (JSON) library and provided to the API through the *navigator.webkitGetUserMedia* method.

2.3.2 Networking API

WebRTC uses a separate API to provide the networking support to transfer media to the other peers, this API is named *PeerConnection* [?]. *PeerConnection* API bundles all the internal mechanisms of the browser that enable media and data transfer, at the same time it also handles all the exchange signaling messages with specific JavaScript methods.

Signaling messages are exchanged using a topology similar to Figure **??**, with the messages being sent either by WebSockets or other similar HTTP polling protocols. Messages are built using a modified bundled version of SDP, WebRTC signaling messages are similar to SIP as they use SDP bodies for the agreement.

WebRTC uses multiplexing over one port when sending the traffic, this means that media, data and monitoring messages are sent over the same port from peer to peer, traffic is sent over UDP or TCP [?]. *PeerConnection* API provides signaling and NAT transversal techniques to bypass routers and firewalls, this part is very important to guarantee a high degree of success when establishing calls in different scenarios.

PeerConnection P2P session establishment system works in a constrained environment similar to RTMFP but it has been designed to provide some degree of legacy for other SDP based technologies such as SIP. Figure **??** shows how a WebRTC simple P2P scenario works, the server used for signaling is a web server.

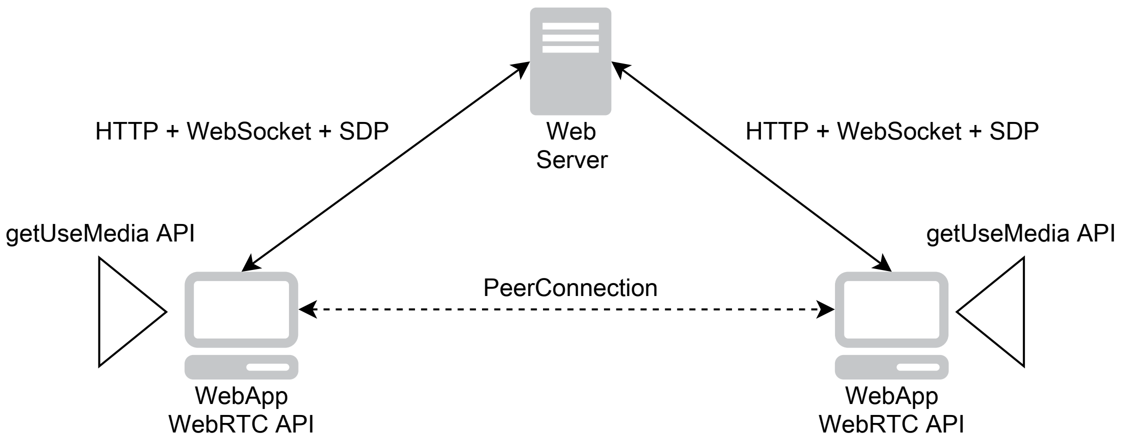


Figure 6: WebRTC simple topology for P2P communication.

Figure **??** does not show relay machines that help to provide NAT transversal solutions. In a real application we could host those relay machines either in the same web server or using external services, those servers must be introduced into the WebRTC *PeerConnection* configuration when starting a new call.

Listing 2: Simple example of *PeerConnection* using JavaScript

```
//XXXX represents the stun server address
var pc_config = {"iceServers": [{"url": "stun:XXXX"}]};
pc = new webkitRTCPeerConnection(pc_config);
```

```
pc.onicecandidate = iceCallback1;

//Localstream is the local media obtained with the GetUserMedia API
pc.addStream(localstream);

function iceCallback1(event){
       if (event.candidate) {
              sendMessage(event.candidate);
       }
}

//When incoming candidate from the other peer we send it to the
   PeerConnection
pc.addIceCandidate(new RTCIceCandidate(event.candidate));

//This is fired when the remote media is received
pc.onaddstream = gotRemoteStream;
function gotRemoteStream(e){
       document.getElementById("remote-video").src =
          URL.createObjectURL(e.stream);
}
```

The previous code describes a simple example on how to use the *PeerConnection* API to perform a P2P connection and start transferring media, this code works in conjunction with the code in section **??**. When building the new *PeerConnection* object we need to pass the JSON object *server* with the stun configuration for the NAT transversal process.

2.3.3 Control and Monitoring

Control and monitoring is an important part of all RTC protocols, this part is usually handled by the software that may adapt the media constraints and configurations to the available resources on the link.

In WebRTC, this is done through the Statistics Model and Constraints defined in the W3C draft [**?**], these methods are part of the actual *ÊPeerConnection* API defined in section **??**. Once the *PeerConnection* is made and media is flowing we need to measure the quality of the connection, this is done by retrieving the stats provided in the Real Time Control Protocol (RTCP) messages that are being sent over the link. The Audio Video Profile with Feedback (AVPF) provides significant improvements in the transmission of RTCP messages that are event driven rather than periodic [**?**] like in other RTP based technologies.

To access the statistical data retrieved from the control messages we need to use the *getStats()* method in the *PeerConnection*, this method allow the developers to access that data in a JSON format that might require some post-processing. Statistical models are useful for the developers to monitor the status of their WebRTC applications and adjust the attributes of the *PeerConnection*.

With constraints developers are able to change media capture configuration by setting Frames per Second (FPS) and video resolution. Other attributes can be set on the *PeerConnection* such as bandwidth requirements, transfer rate is automatically adjusted in WebRTC using its internal mechanisms but we can set a maximum value.

JSON objects for camera and bandwidth constraints must be defined as in the following code.

Listing 3: JSON objects for constraints attributes in WebRTC

```
//Media constraints
var constraints = {
        "audio": true,
        "video": {
                "mandatory": {
                        "minWidth": "300",
                        "maxWidth": "640",
                        "minHeight": "200",
                        "maxHeight": "480",
                        "minFrameRate": "30"
                },
        "optional": []
        }
}

//Bandwidth
var pc_constraints = {
        "mandatory": {},
        "optional": [
         {
                "bandwidth": "1000"
        }
        ]
}
```

Both constraints objects are added to the *GetUserMedia* and *PeerConnection* methods when building the new session. Values are in pixels for the media attributes and Kbit/s for the rate configuration.

2.3.4 Low vs High level API

During the development of WebRTC there has been a lot of discussion in the different working groups about the API layout, those APIs have been designed using the feedback provided by the JavaScript developers.

One of the difficult parts in the standardization process has been to decide the complexity level of the API, how much is available to be accessed by the developers and which configurations or mechanisms should be automatized in the browser. After long discussion, WebRTC is now using JavaScript Session Establishment Protocol (JSEP) [?], this API is a low level API that gives the developers control of the signaling plane allowing each application to be used in different environments, some applications give legacy to SIP or Jingle protocols meanwhile others might only work in a closed web domain.

The media process is done in the browser but most of the signaling is handled in the JavaScript plane by using JSEP methods and functions. Figure **??** represents the JSEP signaling model, this system extracts the signaling part leaving media transmission to the browser. However, JSEP provides mechanisms to create offers and answers, as well to apply them to a session. The way those messages are communicated to the remote side is left entirely up to the application.

One interesting feature that JSEP provides is called *rehydration*, this process is used whenever a page that contains an existing WebRTC session is reloaded keeping the existing session alive. This technique avoid session cuts when accidentally reloading the page or with any automatic update from the web application. With *rehydration*, the current signaling state is stored somewhere outside the page, either on the server or in browser local storage [**?**].

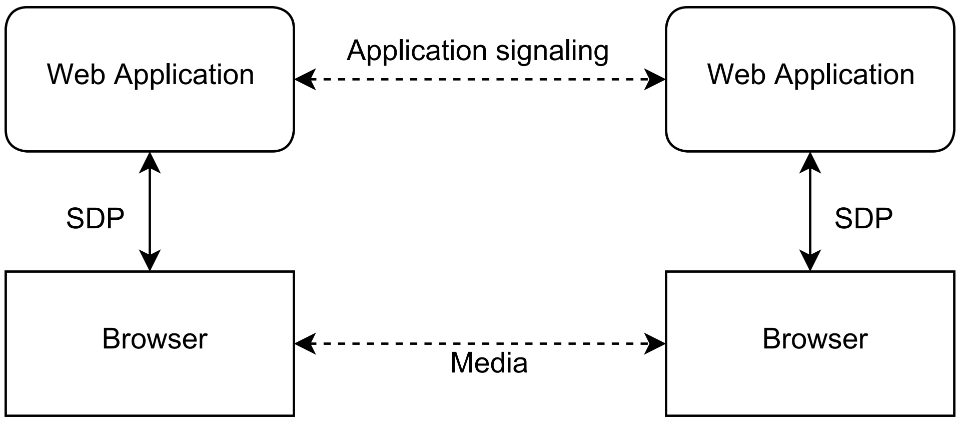


Figure 7: JSEP signaling model.

Low level APIs allow developers to build their own high level APIs that handle all the WebRTC protocol from media access to signaling. Those high level methods are useful to simplify the way JavaScript developers build their applications, building object oriented calls we can have JavaScript libraries that set up and maintain multiple calls at the same time. The benefits of having low level JSEP API for WebRTC are the multiple possibilities to adapt WebRTC to the requirements of each specific application.

In this thesis we use some high level APIs that handle signaling over WebSockets and statistics.

2.3.5 Internals of WebRTC

WebRTC has multiple internal mechanisms that enable the RTC on the browser level by using APIs. Those mechanisms work together to accomplish all the needs for WebRTC features, some of them are related to the network level and others to video acquisition.

One of WebRTC main issues is NAT transversal, this problem usually affect all RTC related technologies. Real-time P2P protocols cannot success in all environments without a NAT transversal solution, in SIP and WebRTC, this problem is addressed with the usage of Simple Transversal Utilities for NAT (STUN) and Traversal Using Relays around NAT (TURN) [?] [?]. Both of these methods are combined with Interactive Connectivity Establishment (ICE) technique that helps WebRTC to decide which is the best way to bypass NATs and firewalls, ICE is widely used in P2P media communications and has proven to be reliable when choosing the best option to succeed with the connexion [?].

TURN and STUN machines are usually placed outside the local network of the clients and help them to find the way to communicate each other by discovering new open paths, the final decision is taken by the ICE mechanism in WebRTC. STUN server provide different IP and port configurations that allow a direct connection to the peer behind the firewall, those configurations are named *candidates*, this information is given to the sender that process the information and tries to choose the best *candidate*. On the other side, TURN works as a relay, this option should be always stated as the last resort when there is no valid STUN *candidate* for connectivity. TURNs work by rerouting the traffic from one peer to the other.

All traffic in WebRTC is done over UDP and multiplexed over the same port except the case of TURN that might use TCP.

Media encoding in WebRTC is done through codecs implemented in the browser internals, those codecs are decided in the IETF working group and have been discussed for long time.

Defined codecs for audio are G711 and Opus. G711 is an International Telecommunication Union (ITU) standard audio codec that has been used in multiple real time applications such as SIP. In real-time media applications, Opus is also a good alternative for G711, Opus is a lossy audio compression format codec developed by the IETF and that is designed to work in real-time media applications on the Internet [?]. Opus can be easily adjusted for high and low encoding rates being a good candidate for the needs of WebRTC.

Along with the codecs, the audio engine for WebRTC also includes some interesting mechanisms such as Acoustic Echo Canceler (AEC) and Noise Reduction (NR) . The first mechanism is a software based signal processing component that removes, in real time, the acoustic echo resulting from the voice being played out coming into the microphone, with this, WebRTC solves the issue of the audio loops with the output and input sound devices of computers. NR is a component that removes background noise associated to real time audio communications. When both mechanisms are functioning the amount of rate required by the audio channel is reduced as the unnecessary noise is removed form the spectrum. AEC and NR mechanisms provide a smooth audio input for WebRTC protocol.

There has been a lot of discussion regarding video codec, two of the proposed codecs are H.264 and VP8. H.264 is a standard codec for video compression, this codec is widely used for recording and transmission of high definition video. Originally it was also selected due its high compatibility with existing devices and software, H.264 has made some controversy as it is patented and licensed by MPEG

LA and may add some royalty problem for WebRTC. VP8 is a video compression codec owned by Google released and released in May 2010, VP8 is supported by Chrome, Opera and Firefox by default and is the de facto codec for WebRTC by May 2013. Later on, Google announced a VP8 patent cross-license agreement to provide royalty-free license to allow developers to implement VP8 video in their web applications [?]. This video codec is adaptive and performs well in low bandwidth links at the same time as providing royalty-free implementation.

WebRTC is not only useful for sending media, it can also provide P2P data transfer. This feature is named *Data Channel* and provides real time data transfer, this can be used with multiple purposes, from real time IM service to gaming, but it is interesting as *Data Channel* allows generic data exchange in a bidirectional way between two peers [?]. Non-media data in WebRTC is handled by System Control Transmission Protocol (SCTP) encapsulated over Datagram Transport Layer Security (DTLS) [?] [?] [?].

The encapsulation of SCTP over DTLS on top of ICE/UDP provides a NAT traversal solution for data transfer that combines confidentiality, source authentication and integrity. This data transport service can operate in parallel with media transfer and is sent multiplexed over the same port. This feature of WebRTC is accessible from the JavaScript *PeerConnection* API by a combination of methods, functions and callbacks. From the developer perspective, all the previous statements regarding security and transport are handled in the browser internals providing a simple and reliable way of sending P2P secure data over WebRTC.

WebRTC provides Secure Real-time Transport Protocol (SRTP) to allow media to be secured.The key-management for SRTP is provided by DTLS-SRTP which is an in-band keying and security parameter negotiation mechanism [?]. Figure **??** illustrates the full protocol stack for WebRTC described in this chapter.

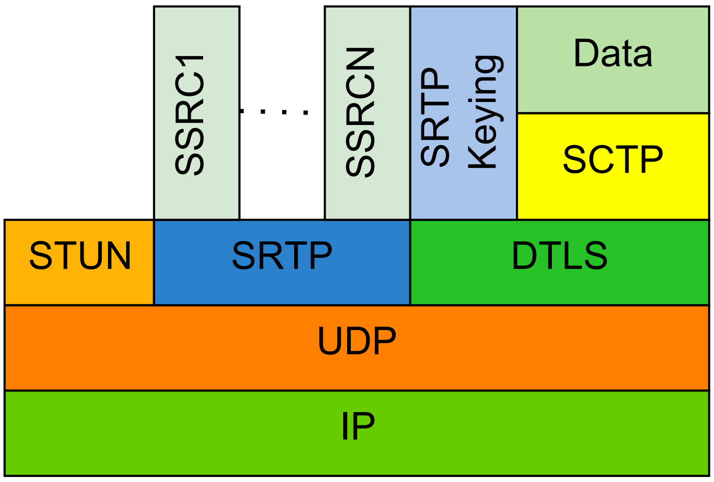


Figure 8: WebRTC protocol stack. Source [?].

Quality of Service (QoS) for WebRTC is also being discussed in the IETF and a draft is available with some proposals [?]. WebRTC uses DiffServ packet marking for QoS but this is not sufficient to help prevent congestion in some environments.

When using DiffServ, the problem arises from the Internet Service Providers (ISPs) as they might be using their own packet marking with different DiffServ code-points, those packets are not going to be interoperable between ISPs, there is an ongoing proposal to solve this problem by building consistent code-points [?]. Audio/video packets shall be marked as priority using DSCP mappings with audio being more important than video or data [?].

WebRTC also uses a Google congestion control algorithm that enables proper congestion control mechanisms for rate adaptation [?]. The aim of this algorithm is to provide performance and bandwidth sharing with other ongoing conferences and applications that share the same link.

2.3.6 Security concerns

To handle the signaling process in WebRTC we use a web server, peers exchange messages to each other through the web server in multiple different ways. By using this system WebRTC provides high flexibility for developers to allow multiple scenarios, on the other side, it also has some important security concerns [?]. Figure **??** represents the simple topology for a WebRTC call, web server handles the signaling messages to the peers and the media transport is done between them and provided by the browser.

Obviously, this system poses a range of new security and privacy challenges different from traditional VoIP systems. It has to avoid malicious calling when having a call established without user knowledge, considering that those APIs are able to bypass Firewalls and NAT, Denial of Services (DoS) attacks can also become a threat.

Actual browsers execute JavaScript scripts provided by the accessed web sites, this also includes malicious scripts, but in the case of WebRTC, this can produce some privacy issues. In a WebRTC environment, we consider the browser to be a trusted unit and the JavaScript provided by the server to be unknown as it can execute a variety of actions in the browser. At a minimum, it should not be possible for arbitrary sites to initiate calls to arbitrary locations without user apprehension [?]. To approach this issue, the user must make the decision to allow a call (and the access to its webcam media) with previous knowledge of who is requesting the access, where the media is going or both.

In web services, issues such as Cross-site scripting (XSS) provide high risk of privacy vulnerability. Those situations, shown in Figure **??**, are given when a third-party server provides JavaScript scripts to a different domain, this script cannot be trusted by the original domain that the user is accessing and could trigger browser actions that could harm the privacy. For example, in WebRTC, we could load a malicious script from a third-party entity that builds a WebRTC call to an undesired receiver without the user noticing this problem. Nowadays, browsers provide some degree of protection against XSS and do not let some scripting actions to be performed.

Other related vulnerabilities in WebRTC APIs is the possibility to establish media forwarding to a third peer, for example, once the user has accepted the access

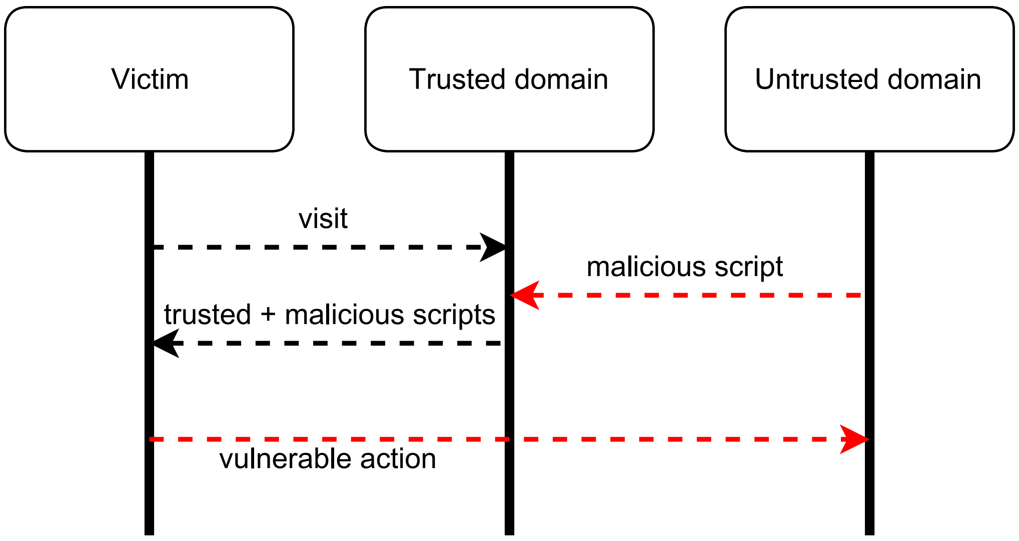


Figure 9: Example of cross-site scripting attack.

to the media, the provided JavaScript could build one *PeerConnection* to the receiver and an extra one to a remote peer that could store the call without the user noticing this behavior. Those problems are not only related to WebRTC and tend to happen in related protocols.

WebRTC calling procedure is done by the JavaScript provided by the server, this may be a problem as the user must trust an unknown authority server. Calling services commonly use Hypertext Transfer Protocol Secure (HTTPS) for authentication whose origin can be verified and users are verified cryptographically (DTLS-SRTP). Browser peers should be authorized before starting the media flow, this can be done by the *PeerConnection* itself using some Identity Provider (IdP) that supports OpenID or BrowserID to demonstrate their identity [?]. Usually this problem is not particularly important in a closed domain, cases where both peers are in the same social network and provide their profiles to the system, those are exchanged previous to the call, but it arises as a big issue when having federated calls from different domains such in Figure **??**.

If the web service is running over a trusted HTTPS certificate and has been authorized access to the media, *GetUserMedia* access becomes automatic after the first time under the same domain, otherwise, the user has to verify the access for each call. Once the media is acquired, the actual API builds the ICE candidates for media verification. Authentication and verification in WebRTC is an ongoing discussion in the working groups.

Security an privacy issues in WebRTC can be given in multiple layers of the protocol, the increment of trust for the provider gives some vulnerability issues that sometimes cannot be easily solved if the aim is to keep a flexible and open sourced real time protocol. Some use cases for WebRTC also incorporate some level of vulnerability as the JavaScript is going to be provided by a third-party, in the use case of media streaming, advertisement or call centers where service providers could pick data form the users and store them for further usage [?].

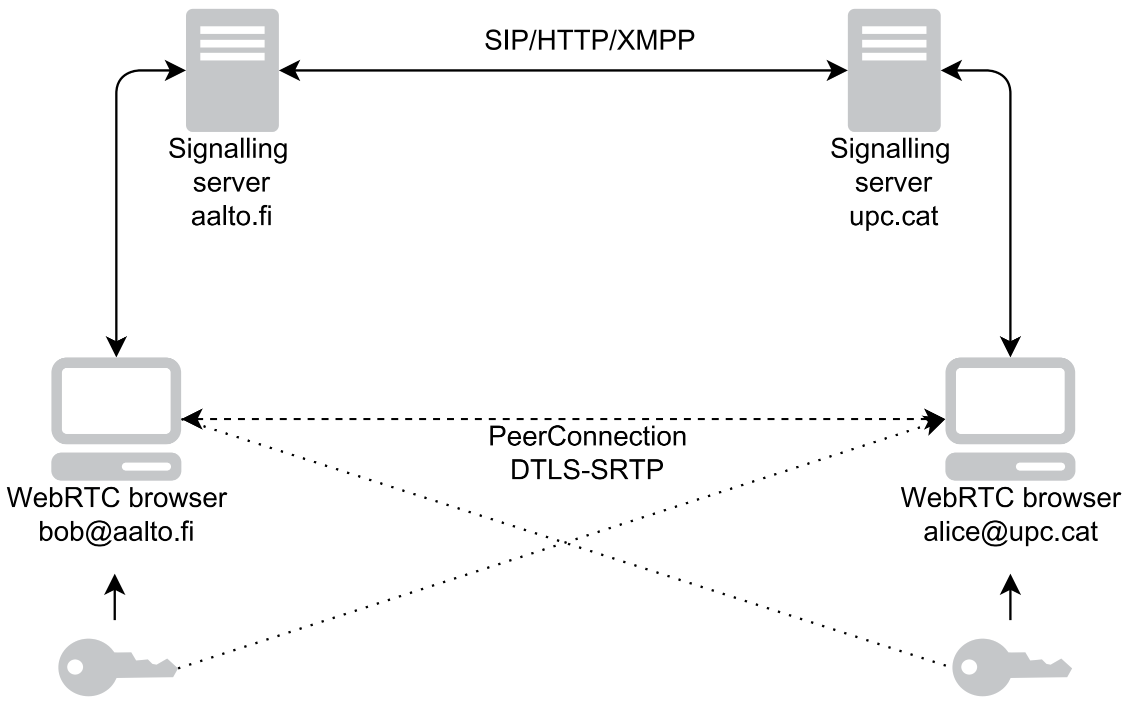

SIP/HTTP/XMPP
Signalling server aalto.fi
Signalling server upc.cat
PeerConnection DTLS-SRTP
WebRTC browser bob@aalto.fi
WebRTC browser alice@upc.cat

Figure 10: WebRTC cross-domain call with Identity Provider authentication.

2.4 Comparison between SIP, RTMFP and WebRTC

After describing various RTC technologies and all the similar alternatives for WebRTC, Table **??** is a summary of common features between SIP, RTMFP and WebRTC. In this Table **??**, common internal mechanisms are described for all of them.

RTFMP is a proprietary protocol which mean that might have its own mechanisms other than the standardized ones stated on the table to solve some of the issues.

All three protocols are designed to provide similar real time functionalities but in different ways, meanwhile SIP is a protocol that helped to develop some of the important mechanisms that are used in other technologies, is still not easily accessible by developers. On the other side, RTMFP provides a licensed alternative for real time communication having not standardized mechanisms and with compatibility issues.

From the mobile perspective, SIP is used in mobile technology and WebRTC has announced to be compatible with future versions of iOS and Android [?]. Furthermore, RTMFP has active support for Android but is still not able to extend its usage to iOS platforms.

All three protocols provide NAT traversal solutions but RTMFP is the only one that provides a proprietary solution that is not standardized, SIP and WebRTC use a conjunction of TURN, STUN and ICE mechanisms.

All of them are valid options, in this thesis we basically work with WebRTC and its related mechanisms.

	SIP	RTMFP	WebRTC
Plugin-enabled	No	Yes	No
Cross-domain	Yes	No	No
Licensed	No	Yes	No
Mobile	Yes	Partially	Yes
Audio	Yes	Yes	Yes
Video	Yes	Yes	Yes
Data	No	No	Yes
TURN	Yes	No	Yes
STUN	Yes	No	Yes
SDP	Yes	No	Yes
RTP	Yes	No	Yes
SRTP	Yes	No	Yes
UDP	Yes	Yes	Yes
TCP	No	No	Yes
SCTP	No	No	Yes
VP8	No	No	Yes
H.264	Yes	No	Yes
G711	Yes	No	Yes
Opus	No	No	Yes

Table 1: Feature comparison between SIP, RTMFP and WebRTC.

3 Topologies for real-time multimedia communication

In this chapter we discuss different possible topologies that can be used along in real time media communication.

A topology can be defined as the arrangement of the various nodes of a network together, those nodes can be connected through different links and configurations. Topologies may have different logics and paths to have optimal performance for each specific scenario. In WebRTC, we want to study how they perform in the most common use cases for real time multimedia communication.

Some challenges are common in all the topologies described in this chapter. For example, NAT traversal problems decide either if the call is established or not, this problem can be solved in WebRTC with the usage of TURN and STUN, but in some restrictive environments it might be impossible to succeed with the call establishment.

For some topologies that include the establishment of multiple *PeerConnections* resource usage can be a big problem. Considering that system capacity relies in how the OS architecture handles processes, CPU and memory usage of WebRTC might be seen as a constraint for those topologies. For example, in Unix based systems every tab of a browser is treated as a separate process meanwhile in other architectures this might be handled different. Media encoding usually consume most of those resources becoming a bottleneck for some scenarios.

3.1 Point-to-Point

The simplest possible topology is a permanent constant link between two peers, this model is widely used in telephony and provides reliable real time communication between users. In WebRTC, point-to-point topologies work only within people in the same domain opposite to many cross-domain communication alternatives such as SIP.

With point-to-point topology we can have traditional dedicated paths where the resources are reserved for each call. In small Local Area Networks (LAN) we use dedicated paths between two WebRTC users, this path can go through the switch or relay but it is unlikely that is going to change the routing. For WebRTC calls over the public internet, the link topology can change at any time trying to use the optimal path with less congestion, this is done in packet-switching technologies where the route is set up dynamically.

From usability perspective, different environments might require point-to-point topologies, direct calls between two users or real time communication for IM can be possible scenarios. Specific uses for point-to-point topologies can cover communication between doctor and patient in a medical web application that is cross-platform compatible and uses an WebRTC. Communication in other cases such as citizens and authorities could also succeed in a WebRTC application.

3.2 One-to-Many

One-to-many or star topologies are one of the most common network topologies for media streaming, this kind of topology consist on a central node that transmits streams to the rest of nodes connected to it. In the WebRTC example of Figure **??**, the central node might be also receiving real time data in difference of the traditional streaming scenarios providing P2P communication between the peers and the central node.

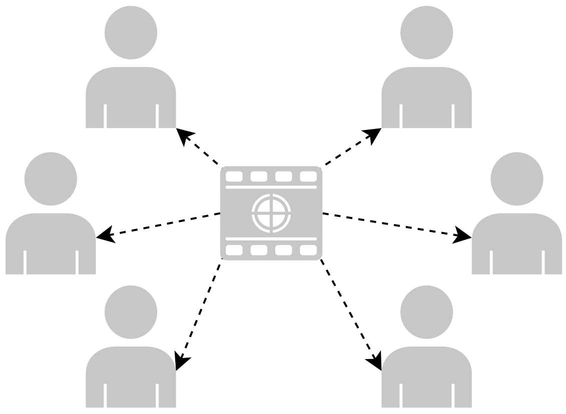


Figure 11: One-to-many topology for real time media.

Star scenarios are known as a type of multicast, one source sends the media to the different clients that connect to the origin. When using this topology, the common uses are related with video and audio streaming to multiple peers, TV media and streaming conferences are popular use cases.

Some of the problems are: high dependency of the central node and, in case of failure, the central node streaming could stop loosing all connectivity with the peers. On the other side, this topology is also good as it provides reliability in case of failure of one of the connected nodes because the rest of the network won't notice any difference on the response.

For example, we could have a major sport even being retransmitted to the viewers by using one-to-many. Other solutions could cover the use of WebRTC to have a CEO talking to the employees with an HMTL5 web application. Music bands also could take advantage of this scenario by being able to transmit his show to the audience with feedback in real time or having the members playing from different geographic areas. All the previous examples take advantage of WebRTC by having direct feedback from the connected nodes, actual media streaming technologies do not provide this kind of communication between the viewer and the origin.

In star topology we have a video, audio and data streaming connection from one source to multiple devices. This might cause a huge load on the source when having multiple *PeerConnection* running, central node performance can be a big constraint in this scenario. Observing other topologies, in most cases, media delay on the network is not as important as other options due to the one-way communication only. In most scenarios, video and audio is not required to be received on the source, so having the media delayed a couple of seconds is not going to affect the user experience in the call. Those scenarios are one-way only use cases.

From the client perspective, the *PeerConnection* stablished is easy to handle as in most cases no data is going to be sent back to the source, except the RTCP control messages.

3.3 Many-to-Many

Many-to-many topologies are also known as mesh, this style of topology is used in multiple VoIP systems for conferencing purposes. Conferencing systems are widely extended in enterprises for long-distance communication between employees and working groups, by this, the need of having those calls working with good response for all participants is very important.

In a full mesh topology all peers connect between them increasing the number of connections and used resources. The value of fully meshed networks rely on the number of subscribers, the amount of *PeerConnections* stablished in a mesh network shall be dependent on the amount of people in the conference. The number of *PeerConnections* can grow rapidly based on Equation **??**.

$$c = \frac{n(n-1)}{2}$$

c : Number of *PeerConnections*
n : Nodes in the mesh (1)

Equation **??** calculates the amount of WebRTC connections required for a *full mesh* topology.

3.4 Multipoint Control Unit (MCU)

MCU is a device used to bridge streams in conferences, it multiplexes, mixes and encodes media of different sources to be sent over one gateway. MCU usage could be a good alternative when designing some WebRTC infrastructures, the ability to multiplex different streams into the same channel is going to directly affect on how the client performs when reproducing the video.

In real time media topologies, MCU is a common component, used as relay it helps end devices to handle less load for the sources by multiplexing all the streams of the call into the same channel, we can have multiple peers connected to the same MCU that can multiplex the media sent by all of them into one unique stream forwarded to all the participants of the call.

Some MCUs may have to encode and decode media on the fly, this is can be difficult in real time applications but can provide different encoding options to adapt the stream output to the link conditions.

Drawbacks on the MCU model affect the dependency of the end nodes from the MCU, if the MCU fails to give good latency and performance, the call quality is affected and receivers do not get the expected response. Load in the MCU can be very high when multiple conferences are being stablished, this requires abundant resources and good throughput.

3.5 Overlay

Overlaying media streams is the ability of a peer to forward media to a third party. Topologies that use overlay are those that require the media to be forwarded from one peer to the other, this kind of behavior is given in multiple peer topologies such as *spoke-gub* or *tree*, seen in Figure **??**.

Generally, in multiple peer scenarios, we can combine all of the following structures to build a topology that fit our requirements.

WebRTC does not provide native support for media overlay yet, but it is planned to implement those features in future versions of the API. Traditionally overlay has also been used for media streaming over the internet.

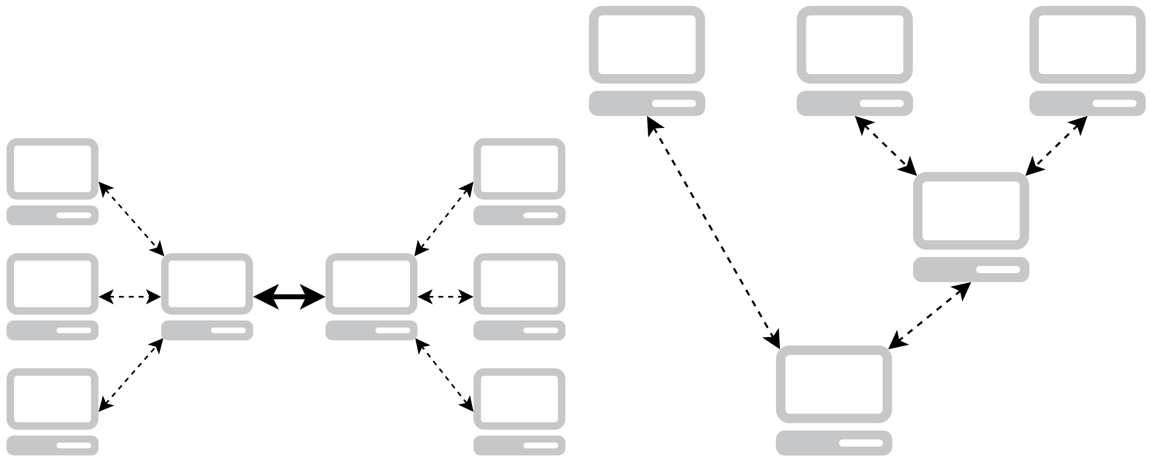

(a) Spoke-hub topology (b) Tree topology

Figure 12: Overlay topologies.

3.5.1 Spoke-hub

Spoke-hub distribution is a topology composed by nodes and arranged like a chariot wheel. Traffic moves along spokes that are connected to the hub at the center. This type of topology, represented in Figure **??**, is good for some scenarios as it requires less connections to perform a full mesh communication in the network.

This is a centralized model, we might have problems if the key nodes of the topology fail. It also relies in one or multiple trunk paths that can be crucial for the success of the streaming, those paths should provide good throughput and low delay.

In some technologies that rely in hub and spoke, the central nodes are usually picked from the end users, calculating the best response from the users the system is able to select the best candidate where the rest of nodes connect to. When this happens that node is handling and forwarding more data that in a standalone call, sometimes without knowledge.

This topology uses the concept of overlay previously described. Spoke-hub environments are also used for logistics in the world, for delivering products and goods around the globe, focusing in bridges over the continents, goods in Europe are distributed within an internal network and shipped to other continents from a centralized node.

3.5.2 Tree

Tree topology is based on a node hierarchy, the highest level of the tree consist of a single node that is connected to one or more nodes that forward the traffic to the other layers of the topology. Tree topologies are not constrained by the number of levels and can adapt to the required amount of end users as seen in Figure **??**.

This type of topologies are scalable and manageable. In case of failure it is relatively easy to identify the broken branch of the tree and repair that node.

On the other side, we can have connectivity problems if a node fails to keep the link up, all the layers under that node are going to be affected and the media forwarding will stop. Overlay is crucial for this topology that is widely used in media streaming, for real time communications, large tree topologies won't be the best candidates given the delay produced when forwarding the packets.

Topologies such as tree are not only used for media streaming but they can also be used to provide wireless coverage in difficult areas, acting as hotspots, each hop can extend the coverage of the wireless in remote areas.

4 Performance Metrics for WebRTC

This section describes metrics to monitor the performance of WebRTC topologies, WebRTC environments require a specific approach and metrics to define how the protocol behaves in different scenarios.

Some issues affect how WebRTC performs, these range from the resources available in the peers to the state of the link. In the following chapters we describe some of them that we use in our study cases.

There are two different type of metrics, from one side metrics related to network performance and on the other side those to the host. Some of them are going to be similar to rate and congestion but others might be more close to the performance of the actual API implementation on the browser.

4.1 Simple Feedback Loop

Traditionally, real time media communication always rely on inelastic traffic, this type of traffic has low tolerance to error as packets should always arrive in time for playback, this is more important than having 100% packet delivery rate. Elastic traffic is common in applications that do not require real time data to be sent and is tolerant to delay, having better data delivery performance compared to inelastic traffic.

With real time media applications, congestion directly affects in traffic delivery rate and performance, if the media generation rate is lower than the available channel capacity there is no need for rate adaption. However, losses and available capacity of the link can vary on time requiring adaptation from the sender side. This adaptation is done by analyzing the receiver feedback packets sent to the sender, this technique is called simple feedback loop.

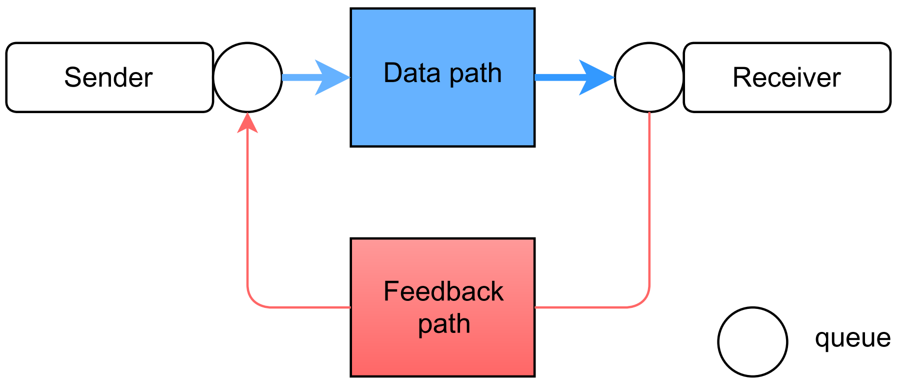


Figure 13: Multimedia feedback loop.

Figure **??** shows a simplified feedback model for multimedia communication, the feedback path carries messages with the link characteristics that help the sender to adjust its congestion mechanisms. Those feedback messages are sent periodically by the receiver and are very important to test the status of the link.

4.2 Network metrics

Metrics defined in this chapter are only related to those provided by the network, those metrics usually trigger the congestion mechanisms in WebRTC changing the behavior of the stream according to the constraints on the link.

Three global factors are considered when analyzing network links: *loss*, *bandwidth* and *delay*.

4.2.1 Loss

Loss rate indicate packet losses during the transmission over the path. Usually packet losses affect directly the performance of a call and can indicate how the link is behaving between the different peers, in WebRTC, packet loss is a direct indicator of the media quality of the ongoing WebRTC transmission.

There are two types of losses, bit-error losses and congestion. Bit-error losses appear randomly and affect some packets, those packets are automatically discarded as the data has been corrupted, this type of error occurs randomly and cannot be predicted easily. Over use of a link produces losses due to congestion of a path, those errors can be anticipated by using congestion mechanisms.

Some delayed packets should also be considered as losses, in real-time applications, as they are not useful anymore for the ongoing connection, those packets are not collected in the stats as losses and are directly discarded in the receiver. In WebRTC, loss rate directly affect to the ongoing transmission as the delay range that we can tolerate is very low before the quality of the call deteriorates, some data-driven WebRTC connections can tolerate some more delay. In overall, Loss Rate is used as a key metric to choose a better path route when having high loss. This indicator is manly attached to link quality.

WebRTC uses RTCP packets for control and monitoring on the ongoing stream [**?**]. In RTCP losses are reported in the feedback message, this metric does not include discarded packets by the protocol.

Losses are calculated in a period of time, with this we shall be able to see how much loss rate we have in a certain path.

$$\frac{PKT_{loss}(T) - PKT_{loss}(T-1)}{PKT_{received}(T) - PKT_{received}(T-1) + PKT_{loss}(T) - PKT_{loss}(T-1)} \tag{2}$$

Equation **??** calculates the estimated packet loss we have on the link. This operation is done periodically by our statistical API.

4.2.2 Round-Trip Time (RTT) and One-Way Delay (OWD)

Delay in a link can be measured form different perspectives, One-Way Delay (OWD) indicates the time it takes for a packet to move from one peer to the other peer, this time includes different delays that are produced along the link. OWD is calculated form the time taken to process the packet in both sides (building and encoding), the lower layer delay in the client (interface and intra-layering delay), queuing delay (from the multiple buffers in the path) and propagation delay (speed of light). The sum of all those delays compose the total one-way delay.

$$OWD = delay_{propagation} + delay_{queues} + delay_{serialization} + delay_{processing} \tag{3}$$

Considering the structure of WebRTC, one of the most important delays that we have to measure and study is the processing delay, as our applications are executed in a multiple layer structure. Running on top of the browser can affect the performance compared to other technologies that run directly over the OS. Delays in our case are symmetric as we are continuously sending and receiving media, low delay is important in order to reproduce the streams in the best quality possible and avoid uncomfortable communication.

OWD and RTT measurements are included in the standard RTCP specification, in order to calculate those metrics, timestamp from sender and receiver is needed in the reports [?]. Sender Report (ST) timestamp is saved in the sender, meanwhile, the receiver returns the same timestamp in the Receiver Report (RR) that goes back to the origin. With those, we are able to calculate the RTT using the following Equation **??**.

$$\begin{aligned}
&RTT = TS_{RR} - TS_{SR} - T_{Delay}\\
&TS_{RR}\text{: Local timestamp at reception of last Receiver Report}\\
&TS_{SR}\text{: Last Sender Report timestamp}\\
&T_{Delay}\text{: Receiver time period between SR reception and RR sending in the sender}
\end{aligned} \tag{4}$$

Calculating OWD requires both machines clock to be accurately synchronized and can be complicated, we try to accomplish this but usually OWD delay can be defined as $\frac{RTT}{2}$.

RTT and OWD are an early indicator of congestion in a WebRTC connection.

4.2.3 Throughput

Throughput is a key metric for testing the performance of WebRTC environments, this value is going to describe how much capacity of the link is taken by each *PeerConnection*. It is complex though, as there are still no fully functional QoS mechanisms implemented in WebRTC at this time. The throughput metric is going to provide bandwidth usage for video/audio in each direction, we can then use this value to get some quality metric in order to monitor the overall performance of the call. A sudden drop of the throughput usually mean that the bandwidth available for that *PeerConnection* has been drastically reduced, this produces artifacts and delays on the stream, or in the worst case, loss of communication between peers. In this scenario ICE tries to renegotiate new candidates in order to obtain an alternative link for the connection and reestablish the streaming with the best possible throughput.

Furthermore, throughput can be divided into sending rate (BR_S), receiver rate (BR_R) and goodput (GP). From the technical point of view, sender rate is defined as the amount of packets that are injected into the network by the sender, receiver rate is the speed at which packets arrive at the receiver and goodput is the result of discarding all the lost packets along the path, only packets that have been received are counted, goodput is a good metric to measure performance.

Typically, those metrics are calculated by extracting the information from the RTCP packets, in our case, we also rely on the Stats API included in WebRTC specification to obtain the amount of bytes and measurements to manually calculate rate by using JavaScript. Taking into account the last amount of bytes received, the actual amount and the time elapsed we are able to calculate an accurate value for the goodput.

$$\frac{\triangle Bytes_{received}}{\triangle Timestamp_{host}} \tag{5}$$

4.2.4 Inter-Arrival Time (IAT) and Jitter

Due to latency on the path, packets can arrive at different times, congestion causes the increase and decrease in Inter-Arrival Time (IAT) between packets. This is also known as packet Jitter.

Congestion mechanisms in WebRTC use an adaptive filter that continuously updates the rate control of the sender side by using an estimation of network parameters based on the timing of the received frame. The actual mechanisms implement rate control by using IAT but other delay effects such as jitter are not captured by this model [?].

In WebRTC, IAT is given by the Equation **??**, **??** and **??** [?].

$$\begin{aligned} &d(i) = t(i) - t(i-1) - (T(i) - T(i-1)) \\ &t\text{: arrival time} \\ &T\text{: timestamp time} \end{aligned} \tag{6}$$

Since the time t_s to send a frame of size L over a path with a capacity of C is approximately:

$$\begin{aligned} &t_s = \frac{L}{C} \\ &L\text{: Frame size} \\ &C\text{: Capacity of the link} \end{aligned} \tag{7}$$

The final model for the IAT in WebRTC can be simplified as:

$$d(i) = \frac{L(i) - L(i-1)}{C} + w(i) \tag{8}$$

Here, $w(i)$ is a sample from a stochastic process W which is given in function of C, the current cross traffic $X(i)$ and the send bit rate $R(i)$. If the channel congestion is high $w(i)$ will increase, otherwise $w(i)$ it is going to decrease decrease. Alternatively we can consider $w(i)$ equal to zero.

Jitter and IAT are usually calculated in the receiver and reported to the sender to have sender-driven rate control.

4.3 Host metrics

Host metrics are measurements made locally by the host that affect the behavior of real time media communication. Those metrics do not need to be directly related to the network and can give information about the host performance when using WebRTC.

They also provide information about timing and encoding for media in WebRTC sessions.

4.3.1 Resources

In WebRTC, we measure the local resource usage in the peers participating on a call, this metric is going to be very important for multiple peer topologies and resource demanding encoding.

CPU/RAM usage is stored for each call in order to calculate an average for each WebRTC scenario. This information is crucial for the success of WebRTC in some environments that are demanding large amount of resources.

4.3.2 Setup time

We also measure the setup time required for every topology, with this we can possibly determine an average setup time for WebRTC connections. The setup time is calculated using local timestamps when building the *PeerConnection*Êobject.

Once the media form the remote stream arrives, we can subtract both timestamps, start of the *PeerConnection*Êand stream arrival, to check the elapsed time, this is done with STUN and TURN to check the difference in each NAT scenario.

4.3.3 Call failure

Call failure is an important metric that might help to decide wherever to ship a production application based on WebRTC or not. This also fully depend on NAT situations and we are going to calculate the average failed calls with STUN and TURN NAT techniques.

This metric can define the success ratio for establishing WebRTC calls.

4.3.4 Encoding and decoding

WebRTC statistical API also provides real time information about the encoding and decoding bit rate of the video codec. This metric usually varies depending on the congestion mechanisms that change in different path conditions, it is a good indicator for congestion and can give some hints about the way the encoding is done in WebRTC.

4.4 Summary of metrics

Performance metrics for WebRTC help us to determine the behavior of the link just by using information provided by the RTCP packets and the statistical API of WebRTC, the goal of those metrics is to design better mechanisms to properly adapt the rate and response to the condition of the link.

Rate adaptation is a key metric used to deliver a good response in WebRTC calls and to closely study how rate perform in different topologies, some other metrics such as RTT, OWD and setup call directly affect the user experience on the call and should also be taken in consideration.

Based on the results we obtain for those indicators we can determine wether the congestion mechanisms are working as defined in the specs or should be improved for future versions of WebRTC.

5 Evaluation Environment

We have set up a testing environment to help us run benchmarks for WebRTC. Figure **??** describes the functional blocks used for a simple video call over WebRTC.

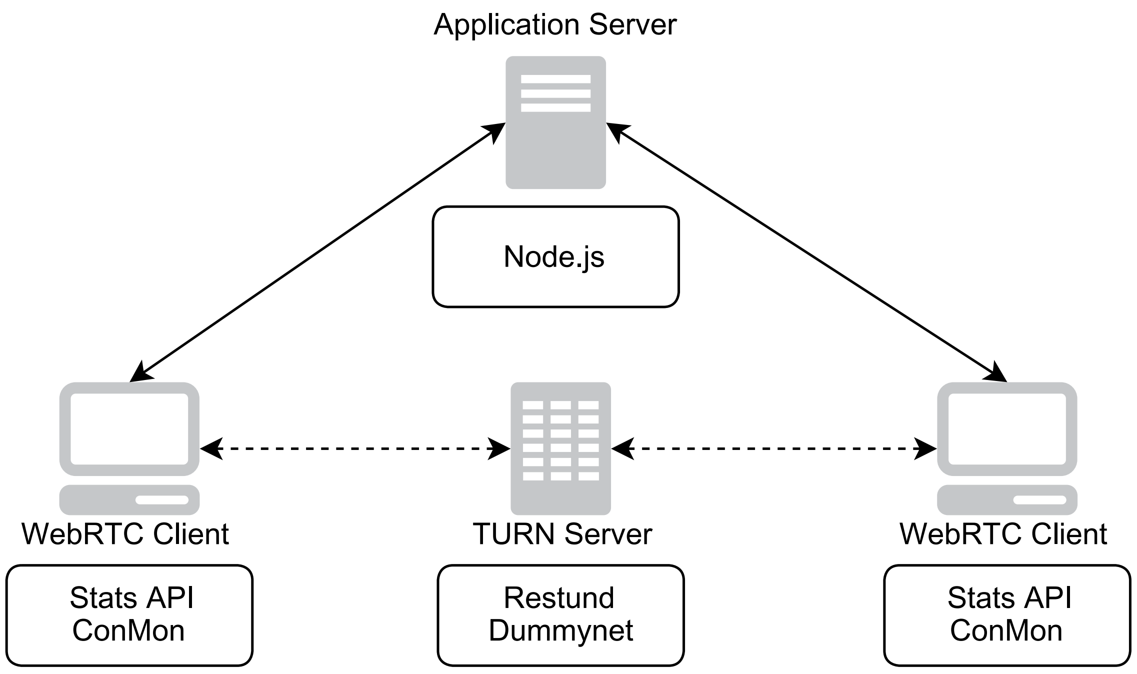


Figure 14: Description of simple testing environment topology for WebRTC.

5.1 WebRTC client

WebRTC clients are virtual machines that run a lightweight version of Ubuntu (Lubuntu[1]) with 2GB of RAM and one CPU. This light version removes the usage of 3D graphic acceleration providing better results in performance than compared with other distributions.

Clients run Chrome Dev version 27.01453.12 as WebRTC capable browser. To avoid unexpected results due to a bug in the *Pulse Audio* module of Ubuntu that controls audio input in WebRTC[2], calls are done with only video, the amount of audio transferred thank to the echo cancelation systems can be neglected.

5.1.1 Connection Monitor

Connection Monitor (*ConMon*) is a command line utility that relies on the transport layer and uses TCPDUMP[3] to sniff all the packets that to go a certain interface and port [**?**]. This utility is designed to specifically detect and capture RTP/UDP

[1]https://wiki.ubuntu.com/Lubuntu
[2]https://bugs.launchpad.net/ubuntu/+source/pulseaudio/+bug/1170313
[3]http://www.tcpdump.org/

packets, works on top of *libcap* library for the network layer. *ConMon* detects and saves the header but discards the payload of the packet keeping the information we need for calculating our performance indicators.

Typically we run the *PeerConnection* between two devices and start capturing those packets using *ConMon*. The *PeerConnection* carries real media so the environment for testing is going to be a precise approach to a real scenario of WebRTC usage.

ConMon captures are be saved into different files allowing us to plot separate stream rates and calculate other parameters such as delay doing some post processing. *ConMon* allows us to compare how precise are both way of analyzing WebRTC, as *ConMon* is working directly over the network interface and avoids all the processing that the browser is doing to send the stats to the JavaScript statistical API. Figure **??** represents one video stream from the same call as Figure **??** captured from the *ConMon* application.

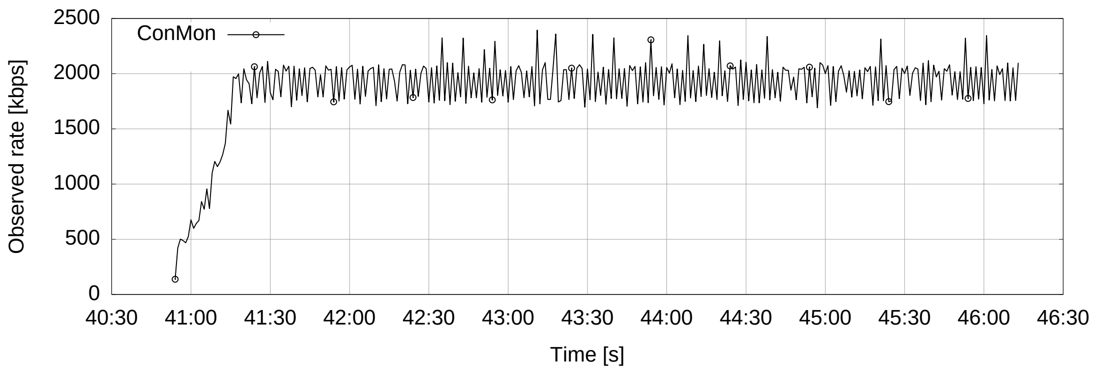


Figure 15: Point-to-point WebRTC video stream throughput graph using ConMon over public WiFi.

The capture from *ConMon* is very accurate as it analyzes all the packets that go through an interface, this data is processed and averaged for a each period of a second before being plotted. This process might output some fluctuations on the graph that could distort the reality in some cases..

Furthermore, *ConMon* is used to provide OWD and RTT calculations for our tests, in order to do this we must assure a proper synchronization between local clocks in all the peers. This is done by using the sequence number of all RTP packets captured and subtracting the timestamp stored from both sides, no RTCP data is used in this analysis.

5.1.2 Stats API

WebRTC statistical API provides a subsection of methods to help developers to access the lower layer network information, those methods return all different types of statistics and performance indicators that we use to build our own high level JavaScript *Stats API*. When using those statistics we process all the output data to obtain the metrics for WebRTC.

This system works in parallel with *ConMon*, both of them can provide similar results of some metrics and the comparison might be interesting to check the differences between the browser API and an interface layer capture.

The method used for *Stats API* is the *RTCStatsCallback* that returns a JSON object that has to be parsed and manipulated to get the correct indicators, this object returns as many arrays as streams available in a *PeerConnection*, two audio and video [?] objects per *PeerConnection* when having a point-to-point call. This data is provided by the lower layers of the network channel extracting the information from the RTCP packets that come multiplexed in the same network port [?].

RTCStatsCallback is the mechanism of WebRTC that allows the developer to access different metrics, as this is still in an ongoing discussion, the stats report object has not been totally defined and can slightly change in the following versions of the WebRTC API, methods involved in the *RTCStatsCallback* are available on the W3C editors draft [?].

We have built a high level *Stats API* that use those statistics from the *RTC-StatsCallback* to calculate the RTT, throughput, loss rate and encoding/decoding rate for the different streams that are being processed. Those stats are saved into a file or sent as a JSON object to a centralized monitoring system. Our JavaScript API grabs any *PeerConnection* passed through the variable and starts looping a periodical iteration to collect those stats and, either plot them or save them into an array for post-processing. Figure **??** represents an example of a captured call between two browsers in two different machines, Mac and Ubuntu, the call was made over Wifi network with no firewall in the middle but with unknown traffic on it. The measures were directly obtained from the *Stats API* we built and post-processed using *gnuplot*[4].

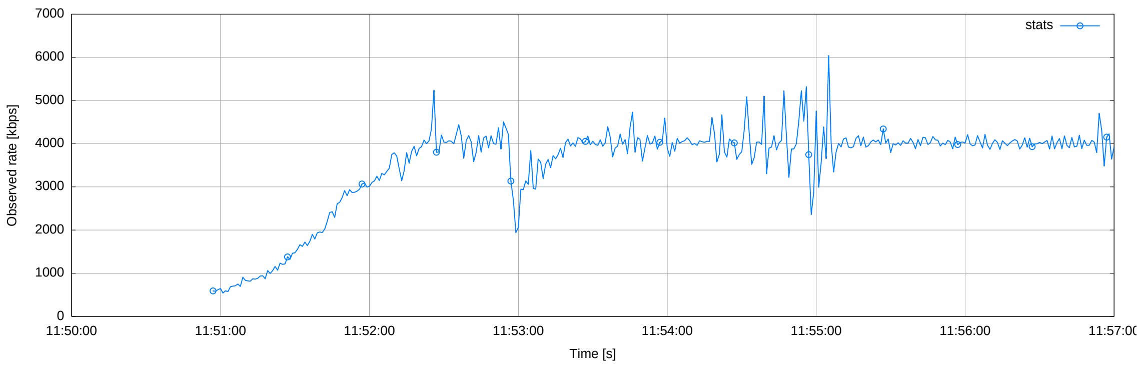


Figure 16: Point-to-point WebRTC video call total throughput graph using *Stats API* over public WiFi.

The previous Figure **??** plots the overall bandwidth of the call, this means that the input/output video and audio are measured together to check how much total bandwidth is being consumed over the duration of the call, as it is using RTCP packets to deliver the metrics to the *Stats API*, it takes a while to reach the average

[4]http://www.gnuplot.info/

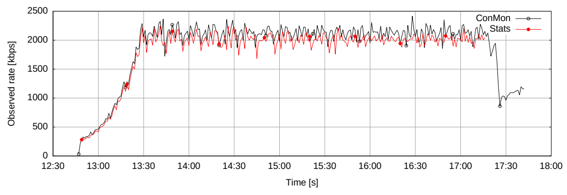
rate value until congestion mechanisms adapt the used rate to the network conditions. We can then plot all the different streams together to get an idea of how much bandwidth the *PeerConnection* is consuming.

5.1.3 Analysis of tools

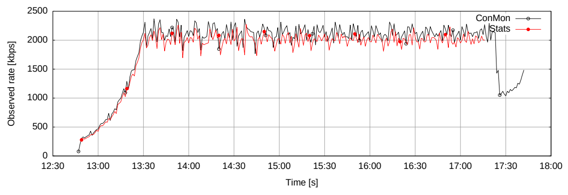
Stats API and *ConMon* measure the same metrics but from different layers of the operating system, this provides us some extra information in order to see how the our high level *Stats API* work and if it is reliable and accurate.

However, due to the period required to capture the data, this could produce strange outputs when plotting the results as the information regarding to the next data period could be stored in the previous one when processing. This is an accuracy problem that cannot be easily solved, when looking at the graph, it is important to observe if two peaks (positive and negative) get compensated by each other, this would mean that the data has not been allocated to the correct period when plotted. This accuracy error is a problem that can be observed when comparing both *ConMon* and *Stats API* capture in Figure **??**.

A second problem that we could face is the time it takes to the OS to process the stats form the RTCP packet and send them to the upper browser layer. Figure **??** and **??** plot two video streams being captured from Stats API and *ConMon*.



(a) Incomming stream.



(b) Outgoing stream.

Figure 17: P2P video stream comparison between *ConMon* and *Stats API*.

Figure **??** represents the incoming media stream from the other peer, we can see the little overhead that is not captured by the *Stats API* interface, as it just reads the bytes inside the payload of the packet. All the RTP headers are not considered when calculating the rate though *Stats API*. We can conclude that the real rate that WebRTC use is going to be defined by the result of *ConMon* instead of *Stats API*, but *Stats API* is going to be an accurate approach.

5.2 Automated testing

For our test scenario we have considered two options, manual and automated testing. The first test environment does not give as much accuracy due to the impossibility to iterate the test many times for the same configuration, if the second option is available the results can be averaged between all the iterations resulting in an accurate result.

In some environments, we won't be able to perform automated testing, when this happens the results won't be as accurate but they can provide a good approximation to the averaged value.

One of the main issues when building a test scenario is the media provided to the *GetUserMedia* input, this media must be as close to reality as possible without using a real webcam. Google Chrome provides a fake video flag that can be activated by adding *–use-fake-device-for-media-stream*[5] parameter, this video though, does not produce enough rate for our purposes.

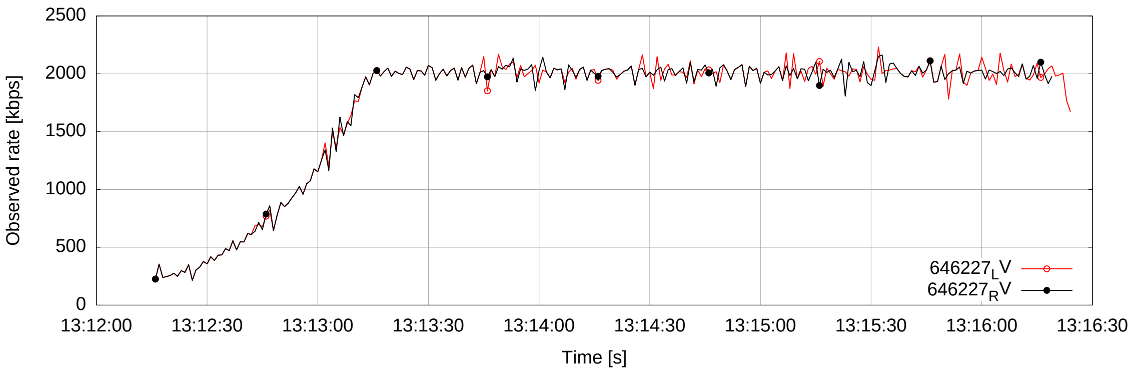


Figure 18: Video stream rate with SSRC 646227 captured using *Stats API* and webcam input.

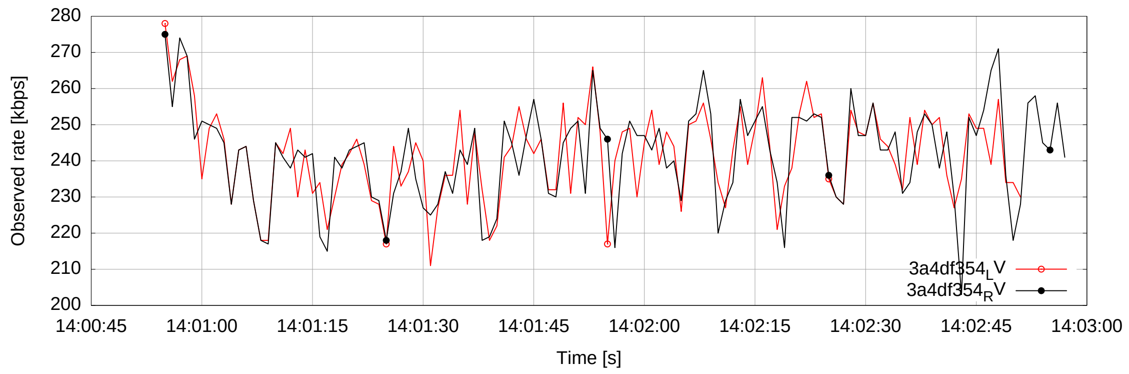


Figure 19: Video stream rate with SSRC 3a4df354 captured using *Stats API* and Chrome default fake video input.

Figure **??** represents the approximate bandwidth that a real video call uses when sending media to another peer, that capture shows the same stream captured from the origin and receiver *StatsAPI* perspective. The adequate rate rises to 2000 Kbps. On the other hand, Figure **??** represents the scenario but using the built-in fake video in both clients, the rate for this case drops to an average of 250 Kbps.

Both figures (**??** and **??**) print one unique stream, identified with the Synchronization Source identifier (SSRC) , but from the sender and receiver perspective, *LV* identifies the source capture and *RV* the receiver stream rate.

[5]http://peter.sh/experiments/chromium-command-line-switches/

Comparing global output from Figures **??** and **??**, we can see that the obtained rate is very different concluding that we cannot use *–use-fake-device-for-media-stream* flag for our testing environment. The reason is that Google Chrome uses a bitmap system to draw the figures and components that will be rendered in the video tag and sent over the *PeerConnection*, this means that the amount of encoding and bandwidth used will be low compared to a real webcam as the media sent over with fake video is minimum.

To address this issue of the media streaming for our automated devices, we have built a fake input device on the peers, the procedure is described in Appendix **??**.

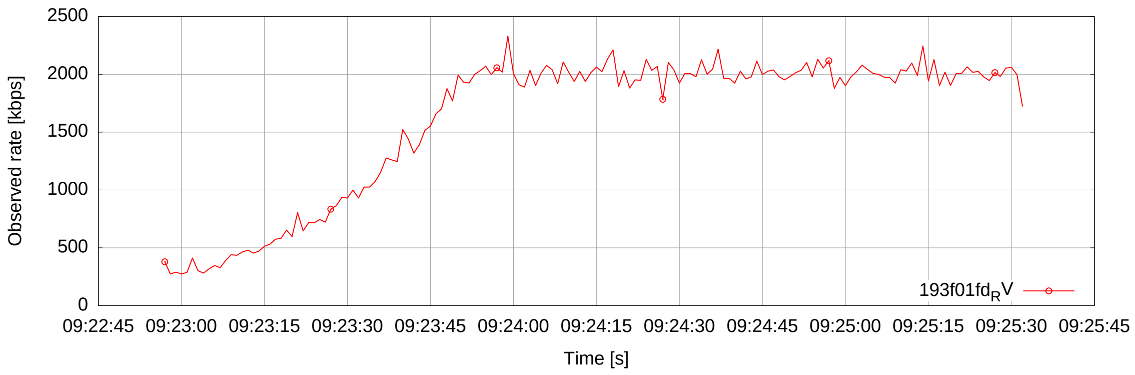


Figure 20: Video stream bandwidth using V4L2Loopback fake YUV file.

Figure **??** represents the bandwidth of a fake video stream measured by our *Stats API* using an YUV[6] video captured from a Logitech HD Pro C910 as source, resolution is 640x480 at a frame-rate of 30 fps.

Results can be compared between Figure **??** and **??**, both average rate output is approximately 2000 Kbps, which means that this procedure is a good approach to a real webcam. This setup allow us to run multiple tests without the need of a physical webcam.

5.3 TURN Server

Our TURN server is used to pipe all the media as a relay, allowing us to apply the network constraints required for the tests to a centralized node, this machine is a Ubuntu Server 12.04 LTS with a tuned kernel adapted to perform better with *Dummynet*.

The TURN daemon we use is called *Restund*, which has been proven to be reliable for our needs, this open source STUN/TURN server works with *MySQL* database authentication [?]. We have modified the source in order to have a hardcoded password making it easier for our needs.

[6]YUV is a color space that encodes video taking human perception into account, typically enabling transmission errors or compression artifacts to be more efficiently masked by the human perception than using RGB-representation.

To do so, we need to modify *db.c* file before compiling. Content of method *restund_get_ha1* has to be replaced with the following line of code, where XXX is username and YYY the password we use for the TURN configuration.

Listing 4: Forcing a hardcoded password in our TURN server

```
md5_printf(ha1, "\%s:\%s:\%s", "XXX", "myrealm", "YYY");
```

Furthermore, in order to force WebRTC to use TURN candidates we need to replace the WebRTC API server identification with our TURN machine by doing:

Listing 5: Configuring our TURN server in WebRTC

```
var pc_config = {
       "iceServers": [{url: "turn:XXX@192.168.1.106:3478",
          credential:"YYY"}]
};
```

The previous object is provided to the *PeerConnection* object enabling the use of TURN.

The IP address points to our TURN server and the desired port (3478 by default), now all candidates are obtained through our TURN. This does not mean that the connection will run through the relay as WebRTC will try to find the best path which may override TURN, to force the usage of TURN candidates we need to drop all candidates that do not force the use of the relay.

Listing 6: Dropping all candidates except relay

```
function onIceCandidate(event) {
       if ((event.candidate) &&
          (event.candidate.candidate.toLowerCase().indexOf('relay')) !==
          -1) {
             sendMessage({
                    type: 'candidate',
                    label: event.candidate.sdpMLineIndex,
                    id: event.candidate.sdpMid,
                    candidate: event.candidate.candidate
              },receiver,from);
       } else {
             console.log("End of candidates.");
       }
}
```

Function *onIceCandidate* is fired every time we get a new candidate form our STUN/TURN or WebRTC API, those candidates need to be forwarded to the other peer by using our own method *sendMessage* through *WebSockets* or similar polling methods. In this code, we are dropping all candidates except the ones containing the option *relay* on it, those are the candidates that force the *PeerConnection* to go through our TURN machine.

This part is important as it allow us to set the constraints in a middle point without affecting the WebRTC peers.

5.3.1 Dummynet

To check the performance of WebRTC we may modify the conditions of the network path to imitate some specific environments. This is achieved using *Dummynet*, a command line network simulator that allow us to add bandwidth limitations, delays, packet losses and other distortions to the ongoing link [?].

Dummynet is an standard tool for some Linux distributions and OSX [?]. In order to get appropriate results we need to apply the *Dummynet* rules in the TURN server, this machine will forward all the WebRTC traffic from one peer to the other being transparent for both ends.

The real goal of using TURN in WebRTC is to bypass some restrictive Firewalls that could block the connection, in our case, this works as a way to centralize the traffic flow through one unique path that we can monitor and modify. From the performance perspective, when not adding any rules to the TURN, the traffic and response of WebRTC is normal without the user noticing any difference.

Some problems arise when using *Dummynet* in our scenario, we will use *VirtualBox*[7] machines for some testing and for running TURN instance, read Appendix **??** for more information about *Dummynet* configuration for virtual machines.

5.4 Application Server

Our application server runs the Node.js instance to handle the WebRTC signaling part, this machine uses Ubuntu with a domain name specified as *dialogue.io*.

This app is a common group working application that allow people to chat and video call at the same time in their own private chat rooms, we have modified it to build an specific room for our tests, this instance simply allow two users that access the page to automatically call each other and start running the JavaScript code with the built-in *Stats API*.

Most of this application is coded with JavaScript and uses WebSocket protocol to handle the signaling messages from peer to peer.

5.5 Summary of tools

Using all the previous mentioned tools together we are able to measure how WebRTC performs in a real environment, some tools have been modified according to our requirements of bandwidth and security. To process the data obtained by all those tools we use some special scripts that measure and extract the information we require form the captures, some of them are explained in Appendix **??**.

[7]VirtualBox is an x86 virtualization software package.

6 Testing WebRTC

In this chapter we study how WebRTC performs in different network environment using topologies previously described in chapter **??**. All tests are deployed using a real working environment combining the tools mentioned in chapter **??**.

6.1 Network Performance Evaluation

We have performed different tests in a point-to-point scenario to study how the WebRTC applications handles different network constraints.

6.1.1 Non-constrained link

Firstly, we are going to proceed with a sample test in a non-constrained link to check the capacity of WebRTC in an open scenario.

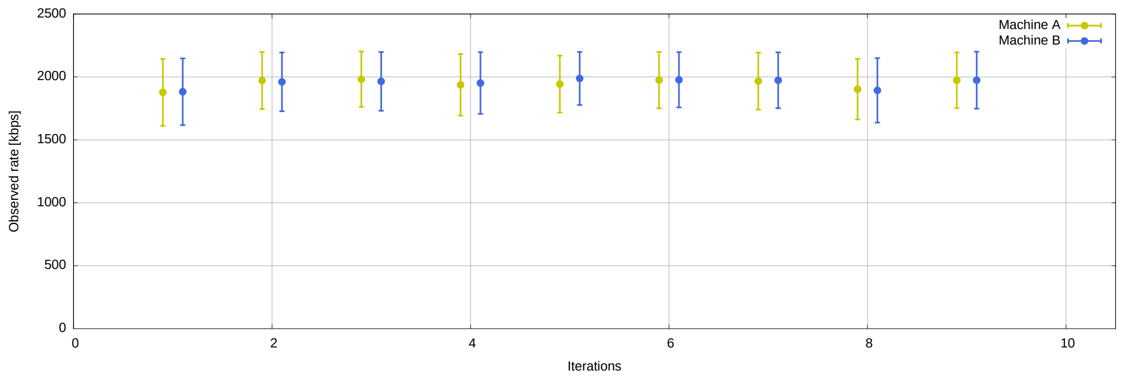


Figure 21: Rate average and deviation for non-constrained link.

Figure **??** represents the average rate of every iteration of the test in a wired network without any link condition, the average rate obtained in the test is 1949.7±233 Kbit/s, we can conclude that a standard rate for a video call in a non-constrained link using WebRTC is approximately 2 Mbit/s. Furthermore, obtained delay is 5.1±1.5 ms and RTT is approximately 9.5 ms, those results can be taken as standard for a non-conditioned WebRTC call.

A summary of results is available in Table **??**, besides the network performance we are also tracking the call failure rate, considering all those calls go through a TURN server we might be able to approximate the success rate when establishing calls in WebRTC.

Setup time is evaluated with the time it takes since the creation of the *PeerConnection* object until the media stream from the other peer arrives, this value defines the time it takes for the user to start the communication, in an optimal environment it takes approximately 1.5 seconds to start the call. We also had zero packet losses and two calls that failed to succeed using TURN in the standard environment.

Delay values in Table **??** are averaged using all the *ConMon* data obtained for each stream in all iterations, thus it is an approximate delay value it might not be

	Machine A	Machine B	Overall
Rate (Kbit/s)	1947.61±232.75	1951.76±234.5	1949.7±233.62
RTT (ms)	9.49±2.11	9.64±2.71	9.57±2.41
OWD (ms)	4.84±1.5	5.4±1.53	5.12±1.52
Loss (%)	0.005	0.007	0.006
Setup time (ms)	1436.33±25	1447.44±22.71	1441.88±24.04

Table 2: P2P metrics output for WebRTC call with no link restriction.

representative of the exact delay occurred during the call. Considering the example in Figure **??** we can see that the delay can variate during the call, this means that the averaged OWD may not be the most appropriate metric to measure the time response in all the different link conditions. In order to evaluate the behavior of WebRTC in delay, we have two different approaches, the mean delay with deviation and delay distribution of all calls.

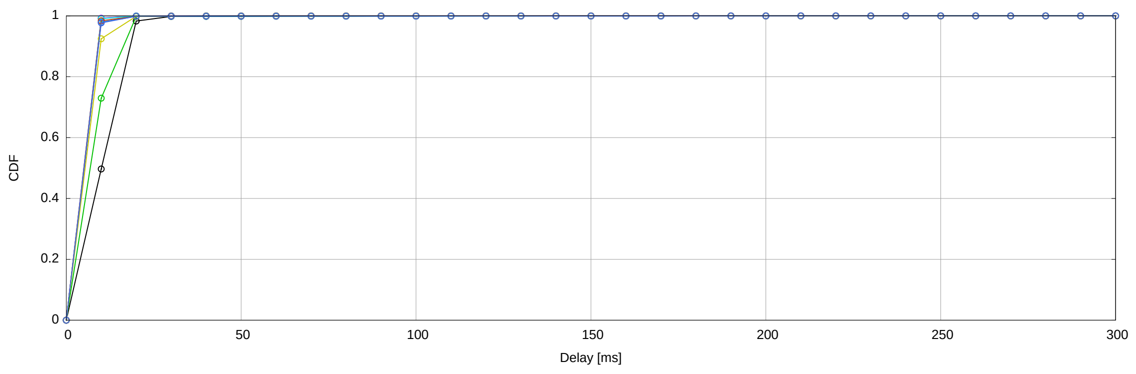


Figure 22: Delay distribution for each P2P iterations with no link constraints.

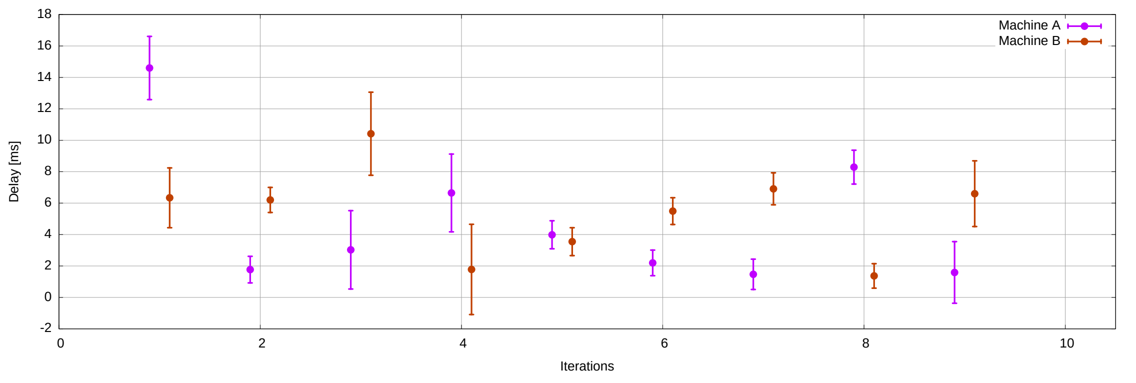


Figure 23: Mean and deviation for OWD in each P2P iteration with no link constraints.

Figure **??** represents the mean and deviation for the delay calculated in each iteration, this delay is calculated by using the arrival timestamp for each packet with the capture done in both peers with *ConMon*. We run a NTPD daemon to adjust the drift on the clock and sync both machines. This Figure **??** represents the delay output in a clearer way than the averaged result between all iterations, the difference in each iteration is small resulting in about 10ms difference between the best and worst case.

In Figure **??**, the distribution is given by the amount of packets whose delay is in a certain range of time, they are counted by batches of 10ms with an adaptable maximum range for each scenario. Most of the packets run with less than 25ms delay in all the iterations for the non-constrained test. The user experience with this small amount of delay that do not vary on time is barely negligible. Figures **??** and **??** try to evaluate the delay response for a specific scenario, this can be difficult as the real time response changes in all iterations depending on the actual of the link, we use the delay distribution and OWD mean to evaluate the delay response.

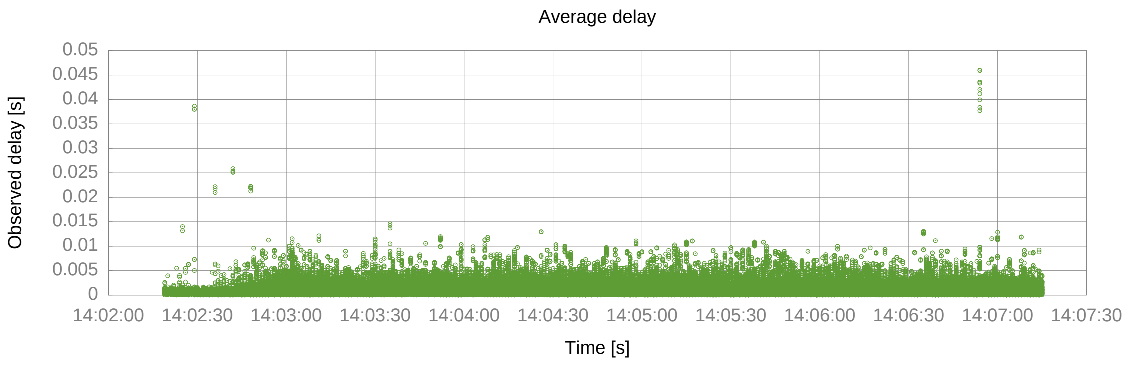


Figure 24: OWD response during the call for one video stream for a non-constrained link call.

However, with the captures performed in *ConMon* and *Stats API* we cal also study one specific stream delay response if we need more information regarding an specific iteration. Figure **??** represents the OWD response along the call of one specific video stream, is easy to see that there is some small variations that may affect the call but the global average is quite stable.

6.1.2 Lossy environments

This chapter evaluates the response of WebRTC calls in lossy environments, those situations can be easily reproduced in mobile environments with low coverage or when having packet drops in any link due to high congestion. Discarding packets in the peers for large delay also produces losses that can affect the call.

Losses in WebRTC directly affect the quality of the media that is sent over the path, when having heavy loss artifacts appear on the video and the user experience degrades exponentially. Google Chrome uses VP8 as de facto video codec, this codec is specifically designed to gracefully degrade the video quality to adapt it to the link conditions, by doing this the video aspect should not influence directly in the user experience.

We have tested the point-to-point topology with 1, 5, 10 and 20% of packet loss, according to the results in Table **??**, we can see a pretty good response in rate adaptation using the internal Google algorithm, rate adaptation response matches the expected output designed in the draft [**?**] in most cases.

Constraints	Metrics	Machine A	Machine B	Overall
1%	Rate (Kbit/s)	1913.59±252.11	1880.24±261.46	1986.91±256.78
	OWD (ms)	4.08±1.79	4.03±1.93	4.06±1.86
	Loss (%)	2.04	1.96	2
5%	Rate (Kbit/s)	1609.65±158.46	1527.74±178.52	1568.74±178.52
	OWD (ms)	3.72±1.82	3.26±1.76	3.49±1.79
	Loss (%)	9.72	9.82	9.77
10%	Rate (Kbit/s)	1166.7±145.96	1114.94±177.88	1140.82±161.92
	OWD (ms)	3.17±3.8	3.12±2.67	3.14±3.24
	Loss (%)	18.98	19.05	19.02
20%	Rate (Kbit/s)	333.34±65.99	295.46±57.98	314.4±61.98
	OWD (ms)	2.65±4	2.78±4.06	2.71±4.03
	Loss (%)	36.08	35.95	36.01

Table 3: Rate, OWD and loss averaged results for different packet loss constraints on the link.

Looking at Table **??** we can see that the packet loss in the transport layer, where *ConMon* is capturing the RTP packets, is higher than the configured in *Dummynet*. This happens due to the probabilistic model of *Dummynet* for packet drop, this mechanism is totally random being not a good approach to reality in some cases and could led to unexpected behavior [**?**]. However in wired environments, packet drops are usually due to queue overßows, queue management schemes, or routing problems. Radio links add noise and interference as other potential causes of drops. Those circumstances are not easily reproducible in testing environments.

The amount of packets lost in the upper WebRTC browser layer is slightly lower than the expected percentage of loss thank to Forward Error Correction (FEC) in WebRTC congestion mechanism on Chrome, this technique is used to control errors in data connection with noisy channels that led to packet losses. FEC is not a must feature to implement in WebRTC but Chrome carries it as default.

When using FEC, the sender encodes the message in a redundant way, by having this redundancy the receiver is able to detect a limited number of errors and autocorrect those errors without requiring retransmission.

On the other side, the sender calculates its rate based on the receive report that arrives from the receiver, if this report is not received within two times the maximum interval WebRTC congestion mechanism will consider that all packets during that period have been lost halving the rate in the sender.

Considering the congestion algorithm in WebRTC [**?**], the rate should not vary when having between 2-10% of packet losses. Table **??** proves that this mechanism is not working properly in some scenarios as we are noticing reduction of rate with the 10% barrier of packet losses, the mechanism should start modifying the rate above 10% of packet lost calculating a new sender available bandwidth ($\mathrm{A_s}$) using Equation **??** being p the packet loss ratio. This 10% value indicates the borderline where the mechanism starts working but it cannot be triggered in an accurate way.

$$A_s(i) = A_s(i-1) \times (1 - \frac{p}{2}) \tag{9}$$

If the packet loss is less than 2% the increase of bandwidth will be given by Equation **??**.

$$A_s(i) = 1.05 \times (A_s(i-1) + 1000) \tag{10}$$

We can conclude saying that the mechanism does work in most cases but should be improved to better adapt the borderline of the 10%. It is also important to mention that the rate adaption is also limited by the TCP Friendly Rate Control (TFRC) mechanism that sets a maximum level for the rate considering more metrics than only packet loss [**?**]. TFRC is triggered when the internal Google algorithm is not capable to properly adapt the rate.

6.1.3 Delay varying networks

Another interesting situation that are given in mobile environments and queued networks is delay, we have also tested the performance of WebRTC in those conditions. We have benchmarked tests in different one-way delays, 50, 100, 200 and 500ms. In our case, the RTT results should be multiplied by two.

Delay modeling for real time applications is difficult and can be done using the timestamp of the incoming packets, the incoming frame will be delayed if the arrival time difference is larger than the timestamp difference compared to its predecessor frame.

We have noticed that the system performs badly when having even small delays up to 100ms. The response of WebRTC is to reduce the bandwidth by discarding packets, this means that the congestion control systems that act in those environments are not working correctly. On the other hand, delay output does behave correctly having a continuous delay of the according time configured in the constraints, there are no sudden increases of delay and the deviation in delay fits in the standard limits.

Table **??** represents the bandwidth response to the delay conditions, it is interesting to see that the deviation with the biggest delay is smaller than expected. Only with 50ms the system will output a good quality call, when increasing delay the performance of the video will decrease. WebRTC uses VP8 codec which degrades gracefully the quality in packet loss and delay conditions but the response in this case should be better if the congestion mechanisms worked properly.

We can also observe that every iteration follows a different pattern even having an averaged result, Figure **??** show the test performed at 200ms and the iterations that fail to keep a constant rate making the amount of artifacts in the video affect

	Machine A	*Machine B*	*Overall*
50ms (Kbit/s)	1909.31±258.09	1917.81±251.62	1913.56±254.86
100ms (Kbit/s)	1516.07±263.43	1453.94±272.79	1485±268.11
200ms (Kbit/s)	503.71±116.45	617.92±142.69	560.82±129.57
500ms (Kbit/s)	303.58±59.22	207.77±32.48	255.67±45.85

Table 4: Summary of averaged bandwidth with different delay conditions.

the quality of the call. We can certainly confirm that the methods that WebRTC should use to control the congestion in the call are not working as they should.

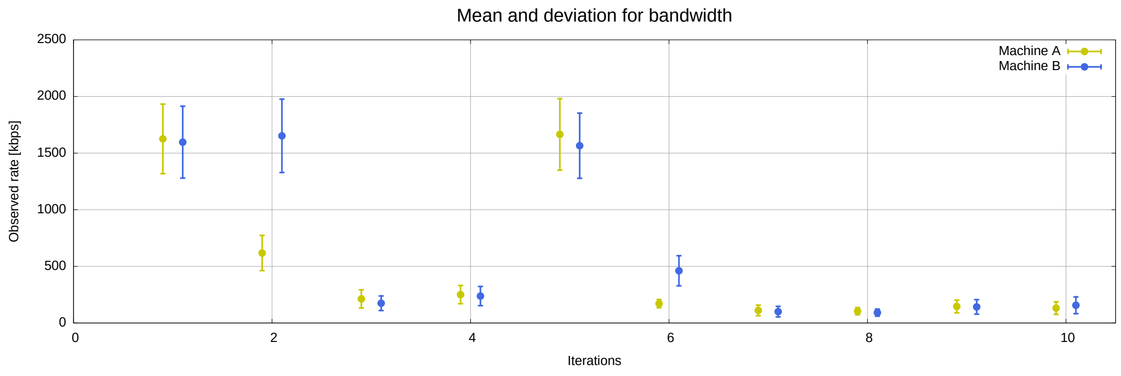


Figure 25: Mean and deviation for P2P 200 ms delay test.

The problem with WebRTC relies in the usage of RTP over UDP for packet transport as UDP does not carry congestion control mechanisms that TCP does, when having real time media adapting the encoding to accommodate the varying bandwidth is difficult and cannot be done rapidly.

Low latency networks will play a big role when WebRTC extends to mobile devices and the ability to react properly to delays and packet losses will be crucial for the success of WebRTC in those environments against its competitors.

6.1.4 Loss and delay

Regarding P2P scenario we also tested the possibility of having a combined lossy network with delay added to it, this kind of environment could be easily found in mobile applications in low coverage areas. We have set 10% packet loss with different delays such as 25ms, 50ms, 100ms and 200ms. In Table **??** we saw an average of over 1 Mbit/s of bandwidth usage in 10% loss environments, the result when adding delay to the constraint is an average of barely 60 Kbit/s. Those results differ due to the difficulty of WebRTC to handle congestion in those environments.

Table **??** describes the averaged bandwidth result with not much difference in each situation.If we study the way WebRTC calculates the rate in difficult situations we can see that the sender will establish its decision on the RTT, packet loss and

	Machine A	*Machine B*	*Overall*
25ms (Kbit/s)	72.59±18.54	70.69±18.09	71.78±18.32
50ms (Kbit/s)	59.7±16.84	60.36±18	60.03±17.42
100ms (Kbit/s)	63.3±19.29	64.82±20.95	64.06±20.12
200ms (Kbit/s)	66.89±20.12	65.66±19.63	66.27±19.87

Table 5: Averaged bandwidth with different delay conditions with 10% packet loss.

available bandwidth that is estimated from the receiving side using Equation **??** [**?**]. Obviously the real output differs form the expected by using the formula, the reason is that even the congestion mechanism on WebRTC calculates the rate using Equation **??**, the sender rate is always limited by the TCP Friendly Rate Control (TFRC) formula that is calculated using delay an packet loss ratio together [**?**].

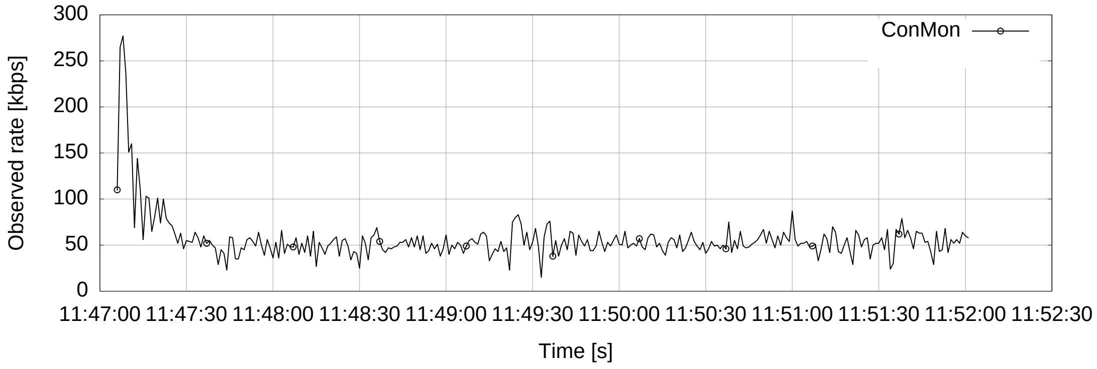


Figure 26: Remote stream bandwidth for 10% packet loss rate and 50ms delay.

Figure **??** is an example that illustrates how the rate is lowered after the beginning of the call even the bandwidth is available. This is due to the formulas and mechanisms previously described.

Carrying delay and losses in the same path will not be handled by the congestion mechanisms in WebRTC giving a low rate output for the stream.

Another interesting factor around this test is the setup time that increases to up 4.6 seconds with 200ms delay and 3 seconds with 50ms, obviously this increase will also affect mobile developers when establishing calls in delayed environments.

6.1.5 Bandwidth and queue variations

We have also performed a different set of tests modifying the bandwidth and queue length. For this part of the test we have chosen to run 500 Kbit/s, 1, 5 and 10 Mbit/s with different queue sizes ranging from 100 ms, 500 ms, 1s and 10 s. In total we have run 12 different tests with ten iterations each.

The queue size is set in slots to *Dummynet* considering each slot as standard ethernet packet of 1500 Bytes, to calculate this number we use Equation **??**.

$$\frac{Bandwidth(Bits)}{8 \times 1500} \times Queue(seconds) \tag{11}$$

We have seen a good response when having big queue sizes but larger deviation in bandwidth when reducing this queue size to 100 ms or 500 ms, this produced high delays over 20 ms for every call with different distribution curves. The delay that is given to the duration of the call is not stable and will affect the media flow, this increasing curve of delay distribution is given by the small queue size which produces bursty packets to arrive to the peer having different delay conditions.

When we tested the 5 Mbit/s case we got a high delay output even the bandwidth response adapted to the constraints, delay deviation is also high and will affect the time the packets arrive with large jittering.

We will study the result of the test performed at 1 Mbit/s limitation as the maximum standard bandwidth for WebRTC is approximately 2 Mbit/s, when having 1 Mbit/s limitation WebRTC will need to adapt the actual encoding rate and bandwidth control to that amount.

Figure **??** represents the bandwidth and mean plotted for all the different tests performed in the 1 Mbit/s case. We can see that the response varies in small amount of bandwidth but with large deviation, when having 500ms and 1s queue size (**??**) we have much more deviation in means of packets being buffered in the relay. Otherwise, when the queue size reduces to 100ms (**??**) the deviation gets smaller but delay response is worst.

We can compare Figure **??** delay distribution results for the best case (**??**) and worst case (**??**). The delay response with large queue is better due to the rapid increase of packets that carry small delay, for the 100ms queue the curve is smoother having packets ranging in all values of delay between 0 and 130ms.

Delay experience with small queue sizes will be worst in the sense of the call flow, we might experience sudden delay situations that WebRTC won't be able to handle, when having larger queue sizes we son't notice the delay variations as much as with the previous example. Having a curvy increase in delay distribution figure will result in sudden delay variations in the call. The conclusion is that WebRTC is able to adapt to low capacity networks using its codec mechanism at the same time as it should improve the congestion control systems to adapt to different buffer sizes and queuing conditions.

The congestion mechanisms in WebRTC will stabilize the rate until the amount of delay triggers the rate change to fit the new queue state requirements. Figure **??** and **??** show the bandwidth and delay for the same stream and how the rate adapts once the queues are full increasing the delay on the packets, rate is lowered and queues get empty giving producing low delay.

Studying the way the sender takes decisions about rate constraints we can observe that the available bandwidth estimates calculated by the receiving side are only reliable when the size of the queues along the channel are large enough [?] . When having short queues along the path the maximum usage of the bandwidth cannot be estimated if there is no packet loss in the link, as in this case the packet loss

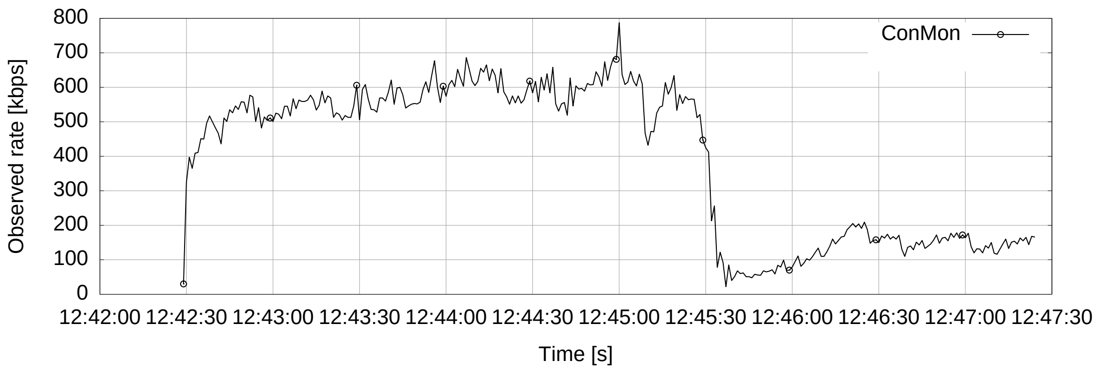


Figure 27: Remote stream bandwidth for 1 Mbit/s and 500ms queue size.

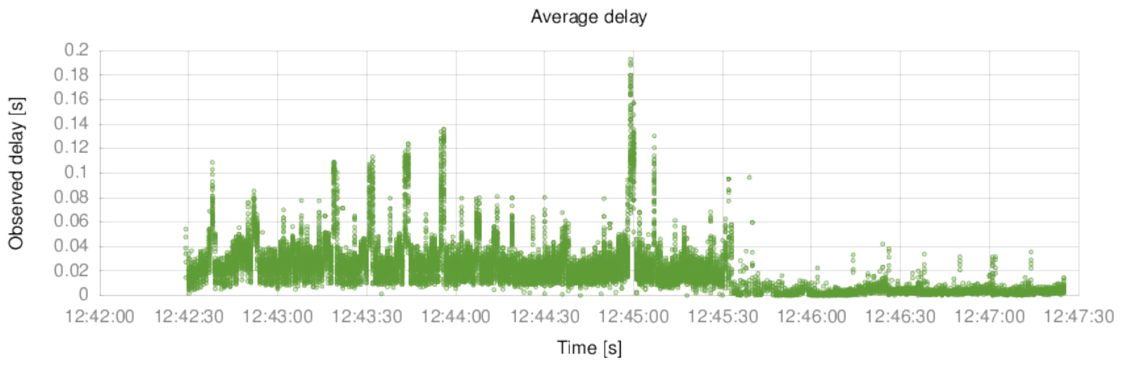
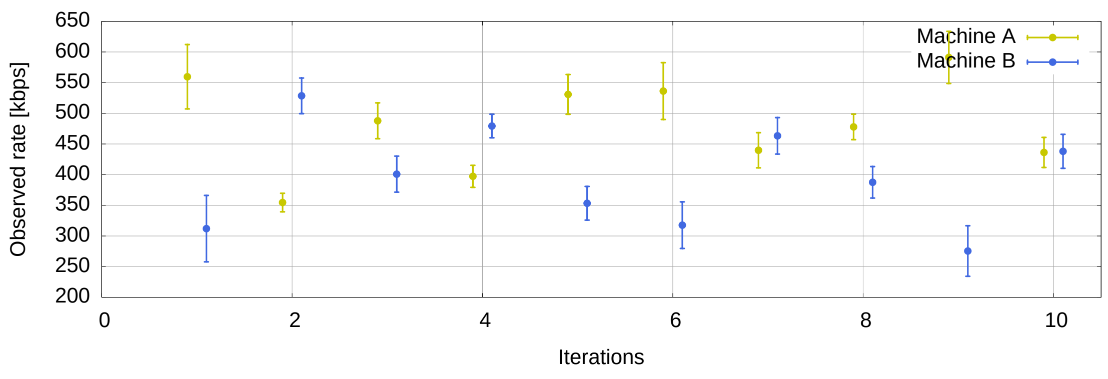


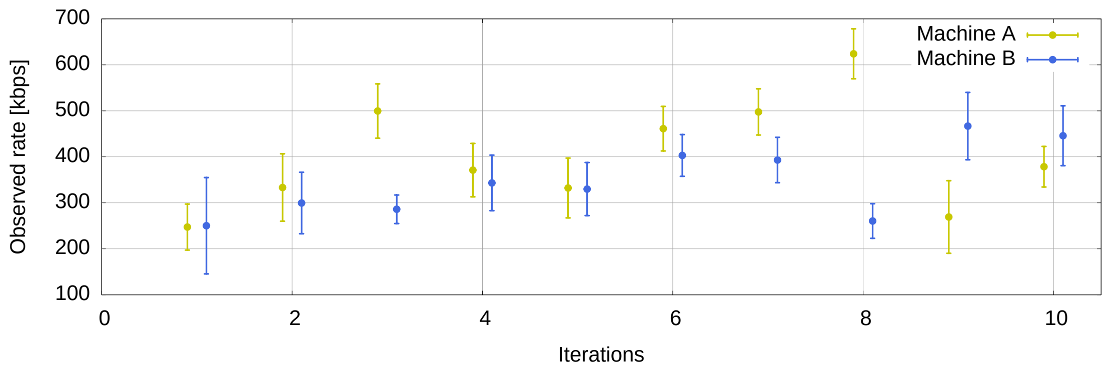
Figure 28: Stream delay for 1 Mbit/s and 500ms queue size.

is negligible, the connection is not able to use the maximum amount of bandwidth available in the link.

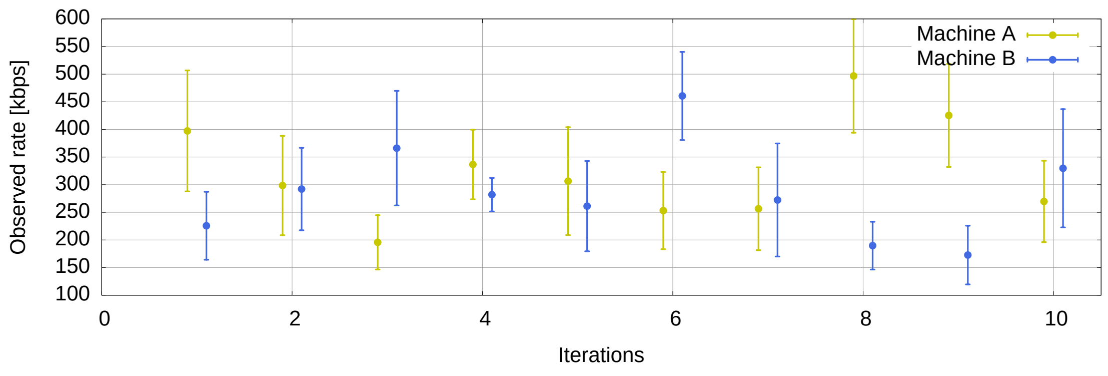
Throughput	Queue	Bandwidht (Kbit/s)	Delay (ms)	Packet Loss (%)
500 Kbit/s	100ms	Lucus Radebe	blah	blah
	500ms	Michael Duberry	blah	blah
	1s	Dominic Matteo	blah	blah
	10s	Didier Domi	blah	blah
1 Mbit/s	100ms	Lucus Radebe	blah	blah
	500ms	Michael Duberry	blah	blah
	1s	Dominic Matteo	blah	blah
	10s	Didier Domi	blah	blah
5 Mbit/s	100ms	Lucus Radebe	blah	blah
	500ms	Michael Duberry	blah	blah
	1s	Dominic Matteo	blah	blah
	10s	Didier Domi	blah	blah



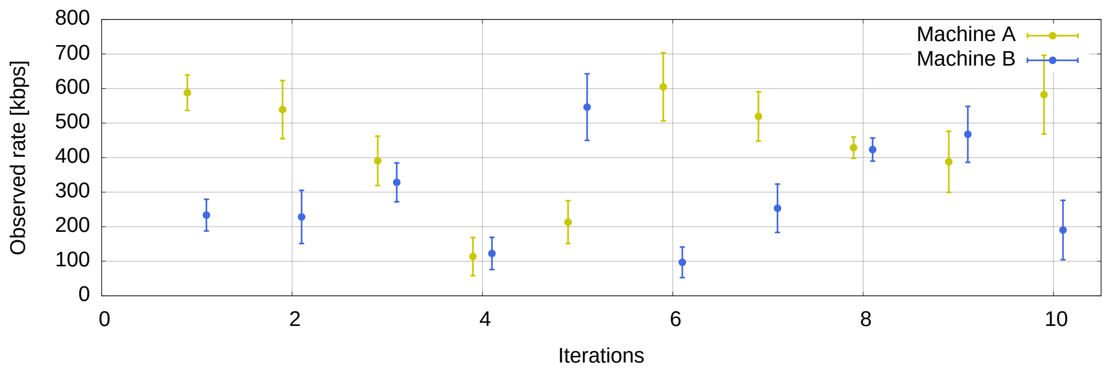
(a) 1 Mbit/s and 10s queue size.



(b) 1 Mbit/s and 1s queue size.

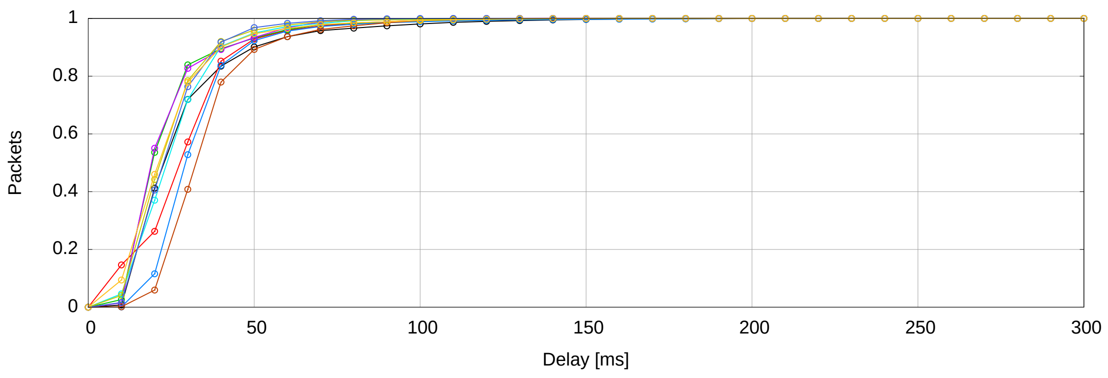


(c) 1 Mbit/s and 500ms queue size.

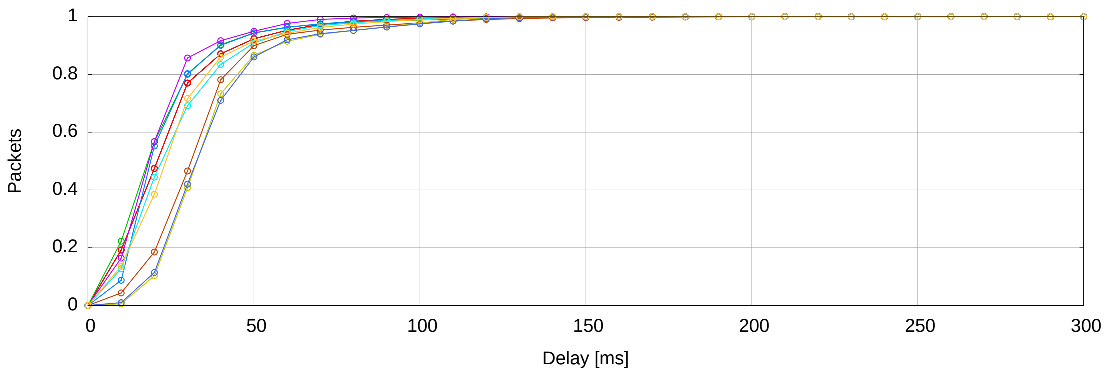


(d) 1 Mbit/s and 100ms queue size.

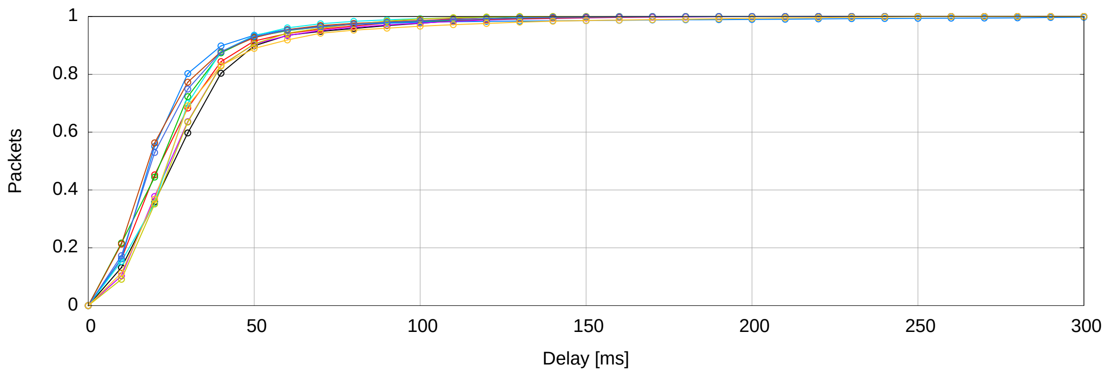
Figure 29: Bandwidth and mean for 1 Mbit/s with multiple queue sizes



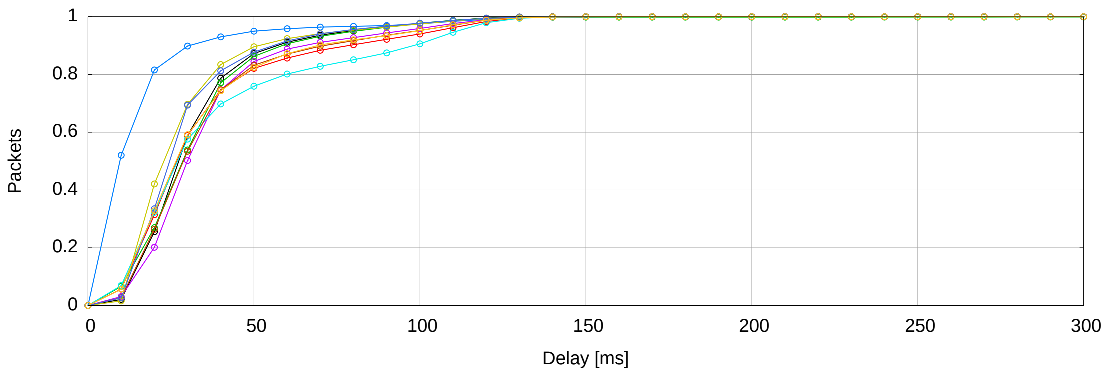
(a) 1 Mbit/s and 10s queue size.



(b) 1 Mbit/s and 1s queue size.



(c) 1 Mbit/s and 500ms queue size.



(d) 1 Mbit/s and 100ms queue size.

Figure 30: Delay distribution for 1 Mbit/s with multiple queue sizes

6.2 Loaded network

Similar to the previous test, in this case we will be measuring the performance of WebRTC in a loaded network using a tool named *Iperf*. This tool will allow us to emulate traffic between our two peers loading the network according to our needs with UDP or TCP packets. The configuration we will use is the one shown in Figure **??** with the clients running *Dummynet* instead of the relay. This scenario is chosen due its widely usage in real devices, having video calls meanwhile manipulating large amounts of online data is something that might happen when using WebRTC.

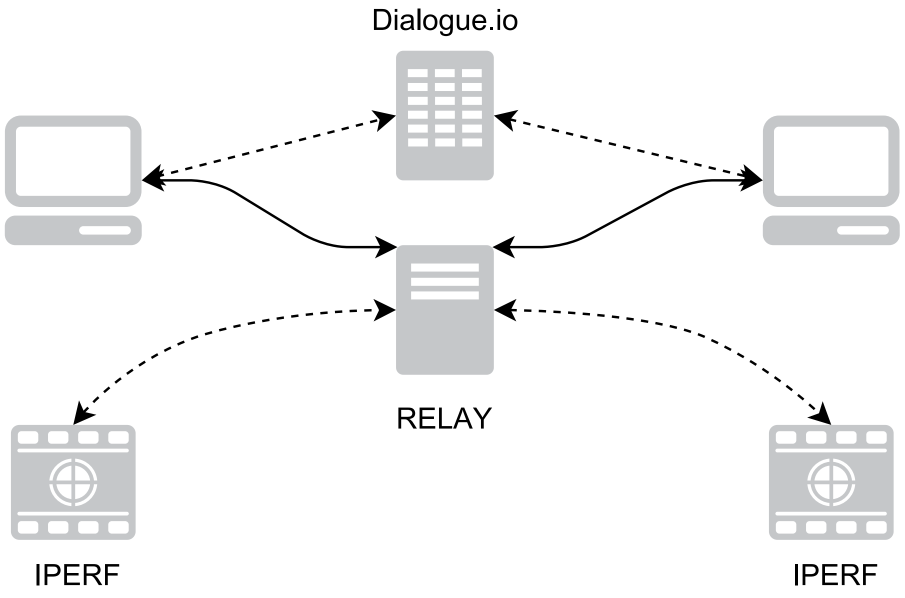


Figure 31: Topology for traffic flooded path using *Iperf*.

In this scenario we are interested in measuring also the behavior of real bandwidth setups for different environments, we will be testing the link with 100/10 Mbit/s and 20/4 Mbit/s limitations, the second one could be defined as the standard for HSPA networks. The data that will be sent to the other peer will be either 10 Mbit/s of TCP and UDP traffic or 2 Mbit/s.

First we will run the server as daemon on the recipient of the packets by executing:

```
# iperf -s -D
```

The next step will rely on the usage of UDP or TCP, *Iperf* sends TCP packets by default, to do so we will run:

```
# iperf -c XXXX -t 300 {-u} -b 10m/2m
```

In the previous command, *-t* is the amount of time the test length, *-c* is the feature that configures the remote server to send the packets to, *-u* is going to be used to sent UDP datagrams instead of TCP and *-b* will define the amount of Mbit/s to be sent to the remote server. In this case every test is run three times.

Table **??** summarizes the results of the 10 Mbit/s TCP packet test without *Dummynet* constraints in the link.

	Machine A	*Machine B*	*Overall*
CPU (%)	81.06±5.2	82.15±5.23	81.16±5.22
Memory (%)	35.65±0.43	34.27±0.39	34.96±0.41
Bandwidth (Kbit/s)	990.11±202.62	1250.13±264.38	1120.12±233.506
Setup time (ms)	1533.66±11.3	1577.66±41.89	1555±32.59
RTT (ms)	25.61±16.02	24.76±14.11	25.19±15.07
Delay (ms)	81.61±11.42	83.99±11.42	81.61±11.42

Table 6: IPERF 10 Mbit/s TCP test without link constraints.

The bandwidth rate in the call is affected by the traffic of TCP packets along the path, at the same time we are getting higher delays. Call behavior in this environment changes in every iteration being unpredictable, Figure **??** represents the bandwidth mean and deviation of every iteration, we can easily observe that in the worst case we are getting three times less rate than the optimum case, ranging from 1.5 Mbit/s to under 400 Kbit/s in the worst iteration.

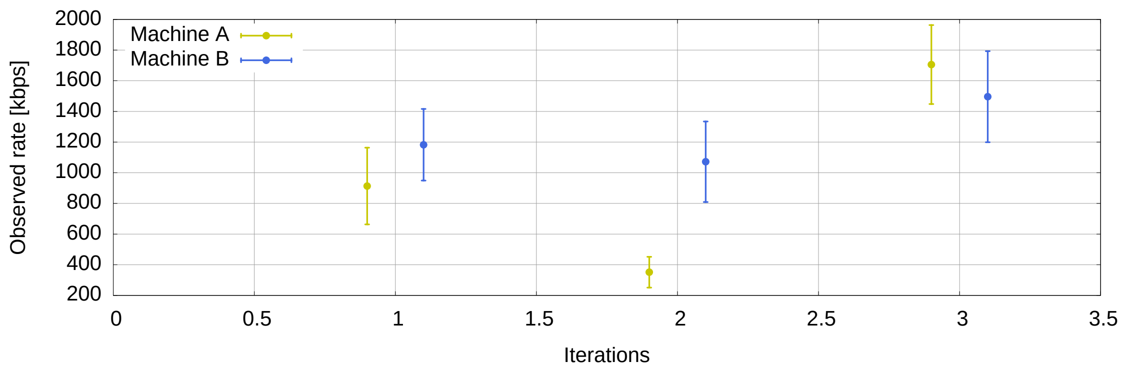


Figure 32: Bandwidth mean and deviation for 10 Mbit/s TCP *Iperf* test without link constraints.

We can observe some interesting behavior in all three iterations when looking at Figure **??** total delay distribution, the response varies from all three tests being all of them bad, a lot of sudden delay changes will appear during the call making real time communication difficult. The delay deviation is small but the tolerance for TCP flooded networks is low int WebRTC.

Now we will test the behavior when sending those 10 Mbit/s with UDP and TCP in a constrained link of 100/10 (downlink/uplink), in this test *Dummynet* scripts have been executed on the client side instead of in the Relay. Table **??** shows

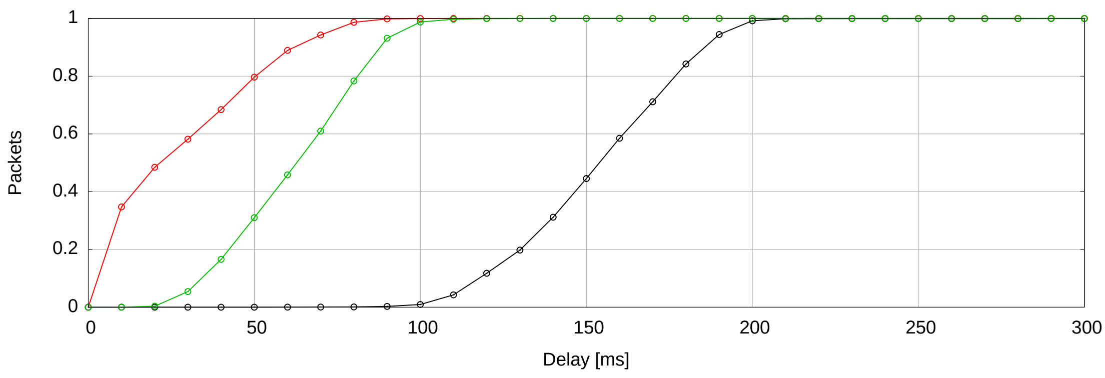
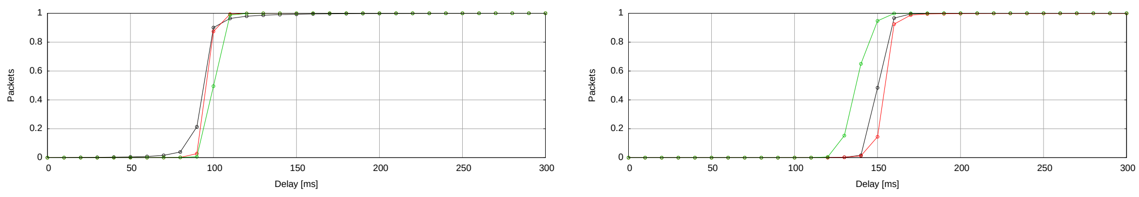


Figure 33: Total delay distribution for 10 Mbit/s TCP *Iperf* test without link constraints.



(a) Delay distribution response for UDP test (b) Delay distribution response for TCP test

Figure 34: 10 Mbit/s UDP/TCP *Iperf* test with 100/10 link condition.

the different bandwidth responses between TCP and UDP traffic, in both cases the link constraint have been the same but the result varies. We will se an increase of rate with TCP flooded packets but also an increase of delay, this delay might be produced due the need of processing more packets with TCP than the simple mechanism of UDP.

	Machine A	*Machine B*	*Overall*
Bandwidth UDP (Kbit/s)	159.41±28.69	149.04±25.76	159.23±27.23
Delay UDP (ms)	98.07±3.14	98.85±2.75	96.85±2.94
Bandwidth TCP (Kbit/s)	208.97±20.64	194.41±18.9	201.69±19.77
Delay TCP (ms)	146.9±4.38	147.92±4	147.41±4.23

Table 7: IPERF 10 Mbit/s TCP and UDP test with constrained 100/10 Mbit/s link.

Delay distribution response in a constrained environment (Figure **??** and **??**) is smoother compared to Figure **??**, the absolute amount of delay is larger but the distribution curve is better for WebRTC needs as it does not have any sudden increase of delay. Delay response when having constraints will output a better delay distribution but with higher RTT in the link.

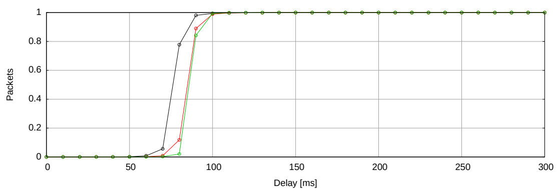
When testing the 2 Mbit/s TCP and UDP flows with 20/4 Mbit/s constraints results are surprisingly close to the version without constraints, we are testing this configuration due to its similitudes to HSDPA networks that carry a similar averaged bandwidth. Unstable bandwidth is also noticed in this test but values for the rate are much higher and delay distribution graphs are similar to Figure **??**. We are using 2 Mbit/s flows to imitate the encoding rate for an online streaming 1280x720 HD video.[8]

Table **??** describes the output we had in terms of rate and delay for the 2 Mbit/s test in a HSDPA type network. Rate adaptation is good even having an small uplink capacity of 4 Mbit/s, the way the rate is adapted to this link confirms that in this kind of not delayed or lossy low latency networks WebRTC could perform properly with simultaneous ongoing traffic.

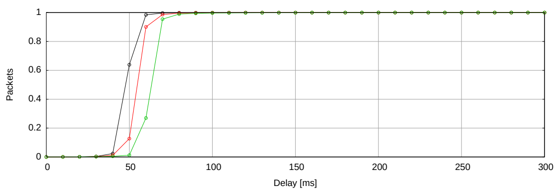
	Machine A	*Machine B*	*Overall*
Bandwidth UDP (Kbit/s)	683.81±259.38	749.66±249.69	716.74±254.53
Delay UDP (ms)	56.34±2.83	54.31±2.64	55.32±2.74
Bandwidth TCP (Kbit/s)	760.94±238.44	1174.95±235.12	967.94±236.78
Delay TCP (ms)	85.18±2.3	80.04±2.26	82.61±2.28

Table 8: IPERF 2 Mbit/s TCP and UDP test with constrained 20/4 Mbit/s link.

From the delay distribution point of view (Figure **??**), the output is similar in both tests being TCP slightly better (**??**) with less absolute delay and with an acceptable variation.



(a) Delay distribution response for UDP test



(b) Delay distribution response for TCP test

Figure 35: 2 Mbit/s UDP and TCP *Iperf* test with 20/4 link condition.

In general, the response of WebRTC congestion mechanisms with ongoing link traffic should be better as this environment will be common for all users. The bandwidth mechanism produces an acceptable call rate but should produce delays smaller than one second which are acceptable from the usability perspective, the delay distribution for the standard case with an ongoing traffic of 10 Mbit/s is not as good as expected but it might be due to the high capacity on the path and the way *Iperf* simulates the traffic.

[8]http://www.adobe.com/devnet/adobe-media-server/articles/dynstream$_l$*ive/popup.html*

6.3 Parallel calls

In this part of the test we will be checking how WebRTC handles multiple parallel calls with different peers, this is not to me mixed with mesh style of topology as it will be running using different tabs or processes through the same TURN path. Figure **??** represents the topology used for the test.

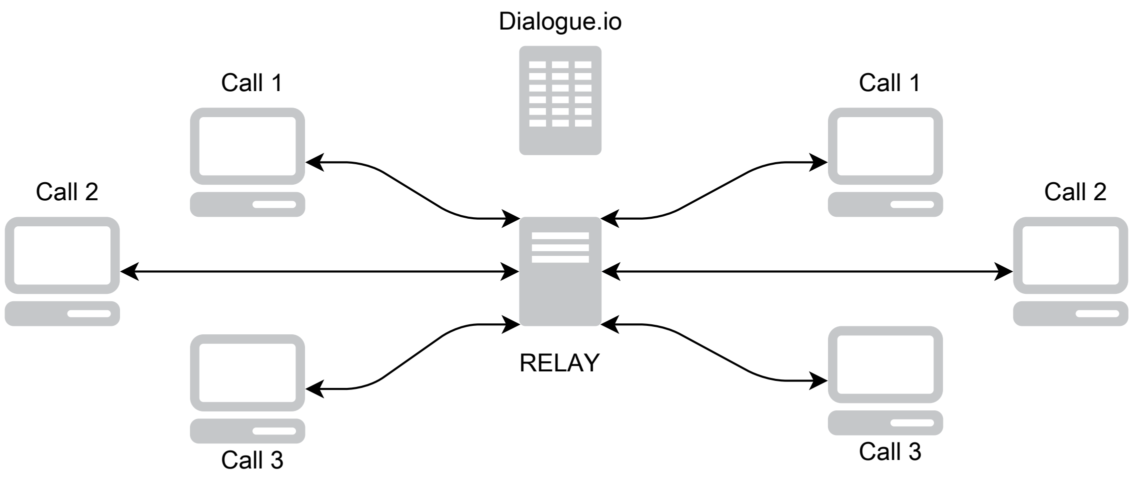


Figure 36: Topology for three different parallel calls using the same link.

We will run a combined batch of tests using 2 and 3 simultaneous calls without *Dummynet* or with 20 Mbit/s and 10 Mbit/s bandwidth limitation for the link. The case without any constraint will run with the standard 100 Mbit/s of the ethernet link capacity. For the test we have focused in running the calls in the same machine but in different processes.

This kind of environment will be given in local networks or it could be compared with mesh topologies handling multiple peer connections, from the resources perspective it will be interesting to observe the CPU and memory consumption as every PeerConnection will be working in a different process, the machine used carries 1 CPU and 2 Gb of RAM.

	CPU (%)	*Memory (%)*
Three calls	99.25±2.41	44.99±0.5
Two calls 20 Mbit/s	95.67±3.51	46.16±0.37
Two calls 10 Mbit/s	86.83±5.03	44.91±0.32
Two calls	81.6±6.48	42.61±0.35

Table 9: Memory and CPU consumption rates for parallel calls in different link conditions.

Table **??** describes the resource comparison between two and three simultaneous calls. CPU usage is critical when handling three peer connections or when the network condition forces the congestion mechanism to continuously adapt the bandwidth and encoding. In this test, each call is placed in a different process which should improve the results as the OS will handle them better than in a single thread. When the CPU load gets to its maximum the performance of WebRTC for encoding/decoding and transmission is deprecated, in this kind of topologies having high CPU performance increases the call quality.

	Machine A	*Machine B*	*Overall*
Three calls	768.04±180.93	850.1±223.84	809.07±202.38
Two calls 20 Mbit/s	432.56±141.32	531.13±169.82	481.85±155.56
Two calls 10 Mbit/s	178.83±60.05	141.83±42.02	160.24±51.04
Two calls	392.08±181.9	545.94±259.27	469.01±221.09

Table 10: Bandwidth rates for parallel calls in different link conditions.

The bandwidth represented in Table **??** is the one used per each machine with the calls together, we will study the worst case. The bandwidth for three calls is to use an average of 800 Kbit/s with different responses in every case, the deviation is approximately ±202 Kbit/s.

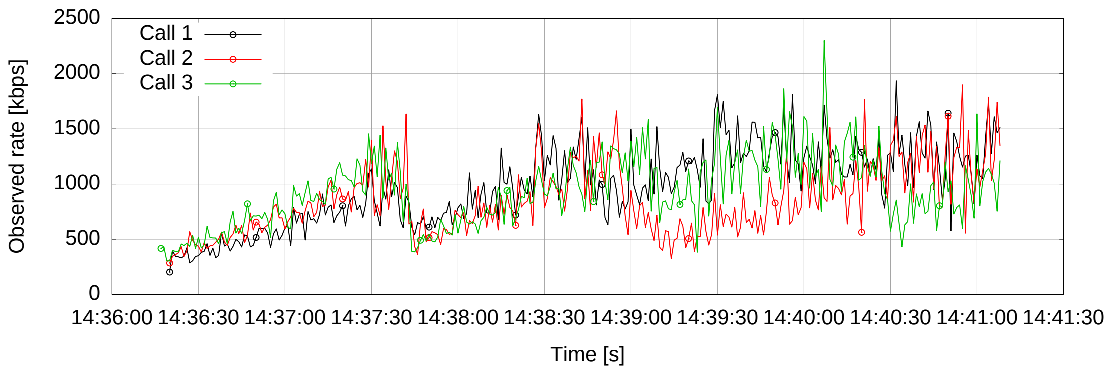
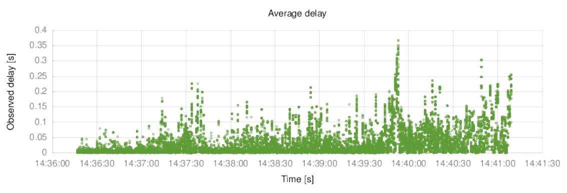
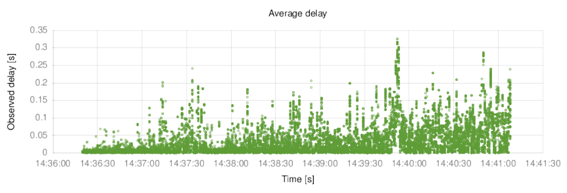


Figure 37: Bandwidth representation for all remote streams in a synchronous three peer parallel call for first iteration.

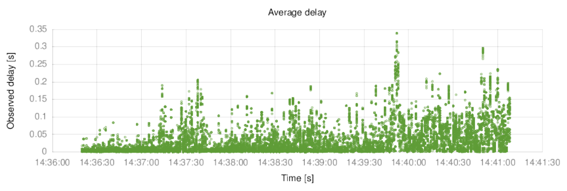
Furthermore, it is interesting to see the global rate on the call as the bandwidth averaged is not following a stable value, Figure **??** represents the bandwidth during all call for the remote video stream of the three peers. We can see how the rate mechanism tries to use the maximum available rate for the actual video encoding but fails to reach the 2 Mbit/s as multiple calls are running and behaving in the same way, this decision is taken also considering the delay that limits the maximum available rate for the call. Figure represents the delay on the same streams during the call, we can observe those peaks of delay during the same period as the bandwidth rise, the result of this is the sudden drop of bandwidth, this mechanism is triggered multiple times.



(a) Remote stream for call 1



(b) Remote stream for call 2



(c) Remote stream for call 3

Figure 38: Delay representation for all remote streams in a three peer parallel call.

We can compare this scenario with the one in Figure **??**, the difference relays in the channel condition, in the example of Figure **??** the channel condition set high restrictions on the path making the rate drop and keep stable as the condition didn't change after that moment. In Figure **??**, path condition changes after the drop as it becomes available again as all the three peer calls behaved the same way.

In general the delay response in all the calls is bad, Figure **??** plots the delay distribution of the three simultaneous calls, the slow increase of delay makes the call lag large and variable, probably the user experience for all the three calls will be bad.

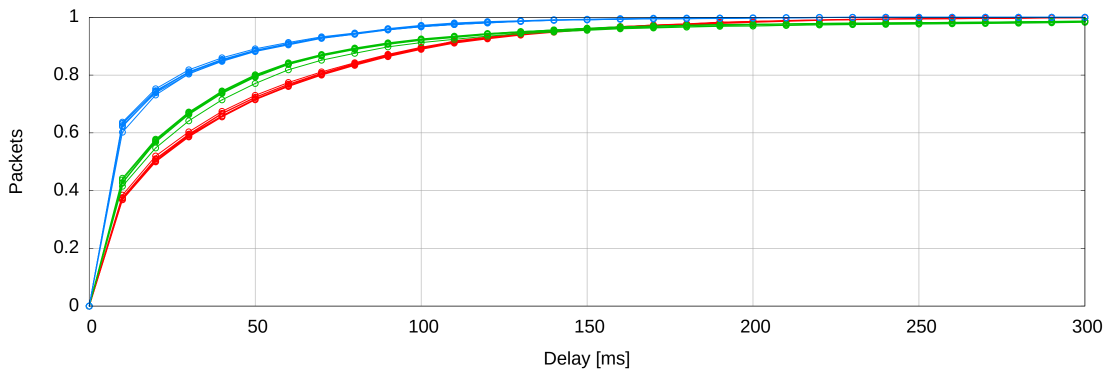


Figure 39: Total delay distribution for three parallel calls.

The problem relies on the separate treatment of every peer connection, they have no acknowledge of having different similar processes going so the rate change cannot be constrained by the other ongoing calls. Notice that for this test all calls started at the same time, for this purpose we ran a second set of tests starting every call delayed by 15 seconds to check the behavior of the system.

After analyzing the captures of the new asynchronous call we can plot the averaged bandwidth to be approximately 1154 Kbit/s with ±250 Kbit/s of deviation.

Overall averaged results are significantly higher than in the previous case but we should have a close look at Figure that represents the bandwidth behavior of the three calls during all the duration for the first iteration we can observe that the average is high but one of the calls rate is very low.

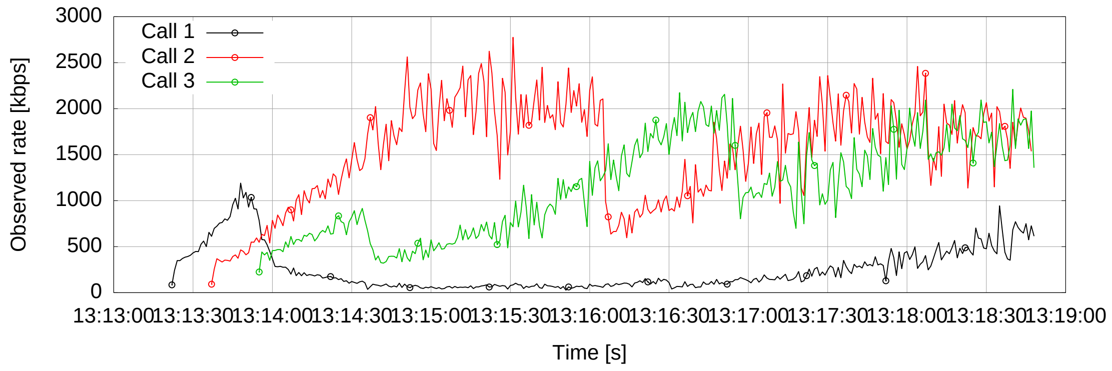


Figure 40: Bandwidth representation for all remote streams in an asynchronous three peer parallel call for iteration one.

In this case, the first call that started to increase its rate suddenly drops it to approximately 100 Kbit/s and stays there along the call meanwhile the other two parallel calls try to obtain the maximum available bandwidth of the path. This environment is more approximated to the a real scenario as users won't start calls exactly at the same time but they will probably do that randomly, some calls quality will be degraded with barely no quality and others will have sudden drop of bandwidth affecting the interaction with the user.

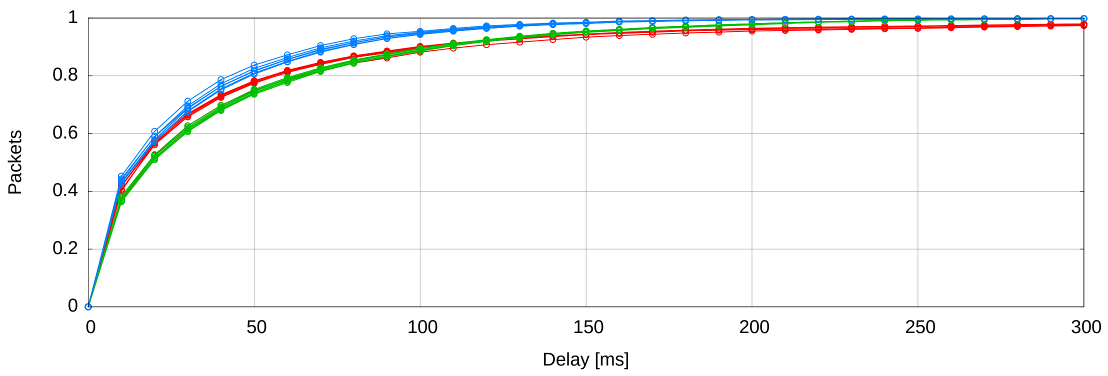


Figure 41: Total delay distribution for three asynchronous parallel calls.

Figure **??** represents the delay distribution for the asynchronous test, similar to the previous example (Figure **??**) but with a worst delay response. Delay will affect to the user experience having sudden cuts of communications of milliseconds that will occur randomly, they are not large but they are random and unexpected.

WebRTC should be able to identify parallel connections of the same type and balance the bandwidth usage, this is difficult as the transport level uses already existing RTP technology over UDP, the conclusion is that WebRTC will not be reliable for multiple parallel calls in an average computer.

6.4 Mesh topology

A common setup in real-time communications is video conferencing, this way of calling people is widely used in virtual meetings. Until this moment there were multiple available options in the market, with the arrival of WebRTC this feature extends to the web application world, with multiple options and features to be enabled with it. We will try to determine if WebRTC is mature enough to handle multiple peers at the same time and session, the way this is done varies depending on the technology, we will study pure peer-to-peer mesh networks.

The most common option for this kind of environments is to use an MCU to perform the relying of the media through a unique connection by multiplexing the streams in a single one. There are some MCU available in the market for WebRTC but the API is still not evolved enough to allow multiplexing of streams over the same Peer Connection, in the following updates of the API this should be enabled allowing developers to perform real media multiplexing. Some vendors offer MCUs that require extra plugins to be installed, this is due to the impossibility to multiplex multiple media streams over the same Peer Connection, this is required when using Google Hangouts product.

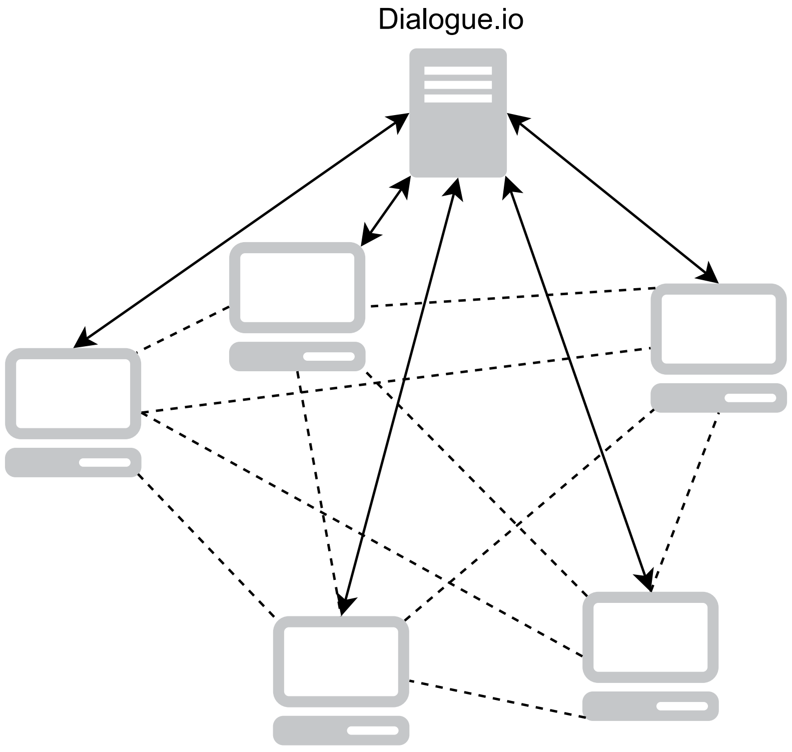


Figure 42: Mesh topology for WebRTC.

Our topology, shown in Figure **??**, consist in a mesh network of different virtual machines that connect by using our centralized *dialogue.io* application for signaling, media is sent over the peers directly, this will produce a big load in the performance for those clients as the amount of peer connections will be large, each of them obliged to encode and decode media, different from the previous example of parallel calls in this case all the peer connections will be running in the same process making the resource management a key point for the performance of WebRTC. We have increased the amount of CPUs to two in order to get proper results.

Three peers are used for the first test and it will increased by one peer every test, the output for the first test is shown in Table **??**. It is important to consider the amount of time required to set up the call, the worst result for the setup time in this scenario has been of 2206 ms to set up the three whole mesh.

	Machine A	*Machine B*	*Machine C*
CPU (%)	88.5±4.77	89.49±4.46	91.65±4.23
Memory (%)	49.25±0.33	50.52±0.28	55.52±0.29
Bandwidth (Kbit/s)	333.38±115.13	344.48±95.43	410.77±115.97

Table 11: CPU, memory and bandwidth results for three peer mesh scenario without relay.

CPU and memory usage is nearly double than in the previous scenario, considering we have doubled the CPU capacity, being this very consuming taking into consideration that is the only application running on the test machine.

Bandwidth will be an important constraint when having multiple peer connections, in a single P2P call we have two streams that are being sent/received, in mesh this amount is greater. Figure **??** represents the bandwidth mean and deviation for the three peer mesh call without relay, for every machine we will have two remote streams plotted with different bandwidth rates for them, is interesting to see how they are related in being one always greater than the other, this means that one stream will handle better quality than its other peer.

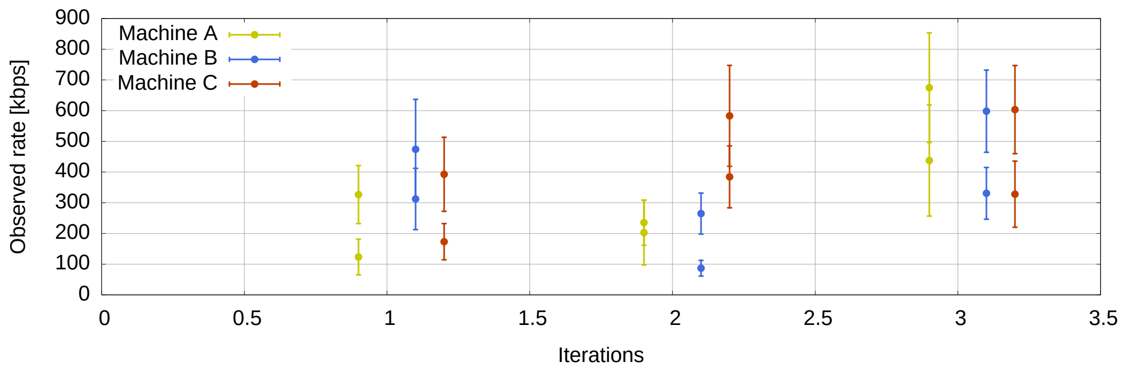
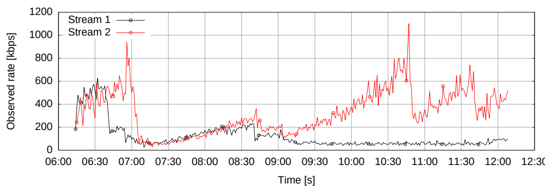
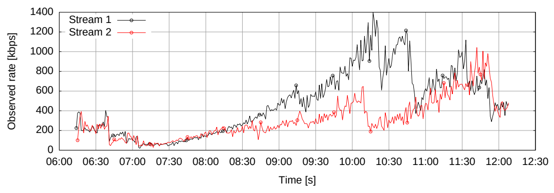


Figure 43: Bandwidth average and deviation for three peers mesh call.

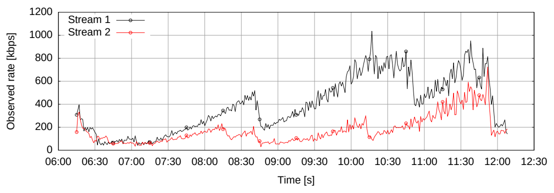
The total averaged bandwidth is approximately 362 Kbit/s, as it can be observed this value does not agree with the real plot for every peer (**??**), this is due to the disturbances of the call and the continuous rate adaptation of the congestion mechanism. For better accuracy and understanding of what happened during the call we should look directly to the continuous rate of the incoming streams in Figure **??**.



(a) Remote streams for machine A



(b) Remote streams for machine B

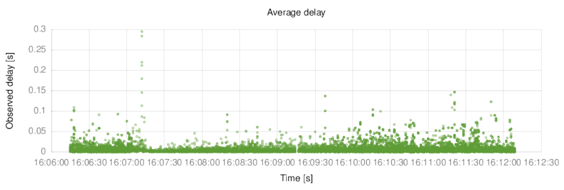


(c) Remote streams for machine C

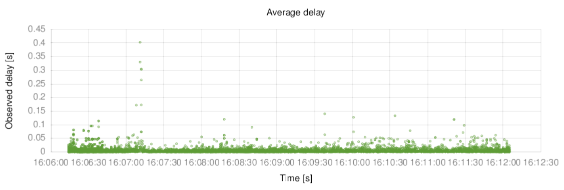
Figure 44: Bandwidth plot during all the call for the incoming streams of each peer for the first iteration.

Each machine will have a pair of incoming streams, the rate calculated for the streams is based on the same mechanisms as the previous tests, this figure should look similar to Figure **??** as they both are running multiple peer connections in the same test, we can see how the rate is being recalculated and how one of the streams is always being deprecated to a very low rate compared to the rest at a certain point when the other increase their throughput. This response of the rate adaptation will be negative to the call.

Furthermore, a different behavior is observed in the delay of the call (Figure **??**), compared to the parallel calls (**??**) which had a very bad response to OWD in this specific case the output for the same four streams in delay is much better than the prior test.



(a) Delay for Figure **??** stream 1



(b) Delay for Figure **??** stream 2

Figure 45: Delay output for Figure **??** incoming streams.

The global delay distribution response is also good compared to previous tests, we can compare the obtained one in Figure **??** with the three parallel calls in Figure **??** and **??**. Considering the curve of the delay for the mesh networks we can say that the delay won't be affecting significantly the user experience during the call duration, we won't have good quality regarding the video and rate but there won't be any sudden delays expected in the communication. This means that from the perspective of a non relayed mesh call we can have three peers with relative acceptable bandwidth in most of the streams with an acceptable delay, the only drawback observed is the amount of used resources by the process, considering that the browser was the only application running on the test machine increasing the amount of processes will probably affect the behavior of the call.

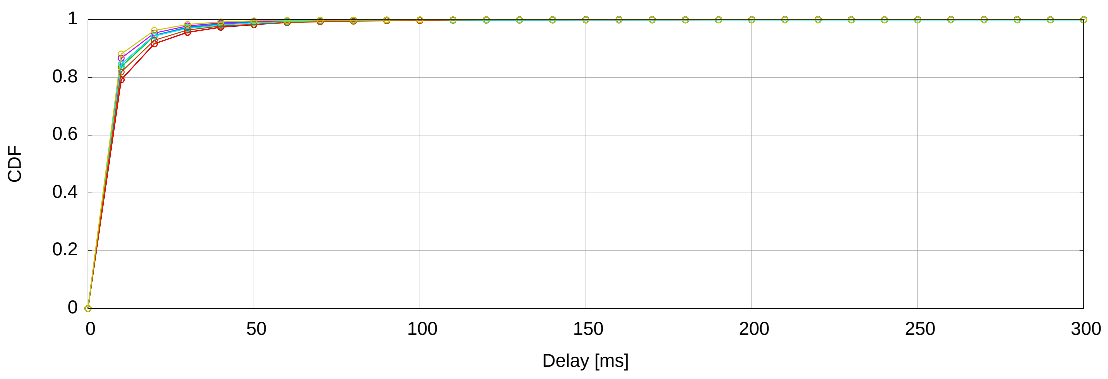


Figure 46: Total delay distribution for three peer mesh call without relay.

Similar to the previous scenario we have run a test using the TURN as relay for the media to check the results, the bandwidth and CPU results are approximately the same, Figure **??** represents the averaged delay result for the three iterations in both tests. Results are slightly better with the TURN, this might be due to the fixed path for routing the packets that produce smaller deviation, the averaged delay is similar in both cases.

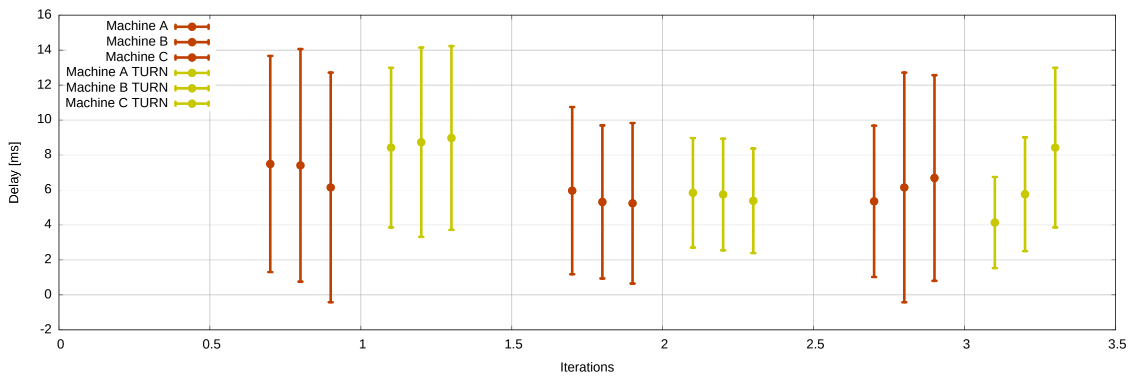


Figure 47: Averaged delay and deviation for TURN and non relayed mesh call for all iterations.

For environments such as mesh, the usage of MCU should be considered as a good alternative to the pure P2P option, this could drastically reduce the resource consumption and bandwidth distribution on the different peer connections when using multiple peer topologies.

6.5 Wireless scenario

To check the performance of WebRTC in a wireless scenario we have stablished a simple call between two peers that handle video and audio in an open WiFi network. This network does not carry any UDP packet filter or Firewall, the connection is performed without the need of STUN or TURN, we could easily say it is a straight forward host peer-to-peer connection. The aim of this test is to observe how the captures differ between origin and receiver between *StatsAPI* and *ConMon*.

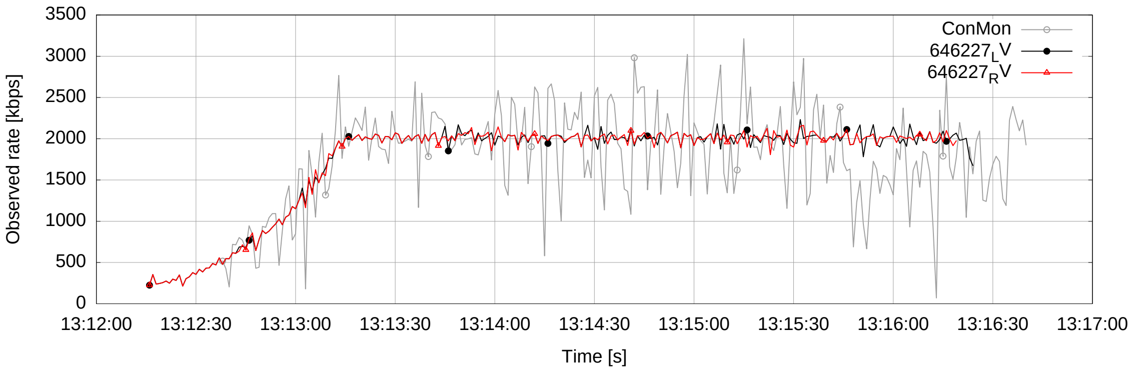


Figure 48: Point-to-point video stream plot using StatsAPI and ConMon data over WiFi.

Figure **??** represents the throughput rate on the same video stream, the three lines are the comparison between local video stream on the source, remote video stream at the receiver and *ConMon* capture of the remote video stream at the receiver. The same stream has been measured from three different perspectives, this will help us to understand the difference of rate that the overhead of the RTP is consuming and the disruption caused by the WiFi network.

Notice that red and black colors represent the Local Video (LV) and Remote Video (RV) from with same SSRC captured using *Stats API*, the grey line illustrates the capture performed using *ConMon* of the same stream. It is easy to observe that both *StatsAPI* captures are similar, some offset is given due to the processing time between the network layer and the browser API that returns all values. Besides this, the capture is smooth and the outgoing and incoming rate of the origin client and input of the receiver are similar. Capture in the network layer using *ConMon* is more abrupt as all packets are captured over the WiFi and the period used when plotting affects the moment when value is stored, when having two peaks with opposite they should balance each other, meaning that the transmission in most of the period is stable and the peaks when plotting are a result of post-processing accuracy. Call duration in this test has been around five minutes. Some areas, mostly between 13.15.30 and 13.16.00, show a strange oscillation of the rate that could be produced by the WiFi, this throughput distortion is balanced on the WebRTC layer as the rate delivered to the API does not change.

When we try to measure the quality of the call, one important indicator is the delay of the stream, to calculate the delay we can either use the RTT measured by our *Stats API* or the captures performed on the network layer by *ConMon*. The *ConMon* procedure will give us higher accuracy by subtracting the timestamps from both captures of the same stream in each peer, to obtain a good result in this procedure we might have to reduce the internal clock drift of both peers by using systems such as Network Time Protocol daemon (NTPd) .

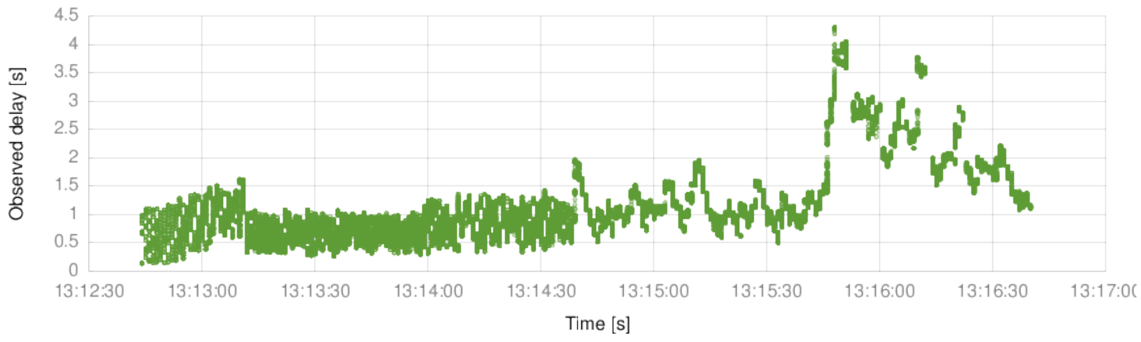


Figure 49: Delay calculated on the same stream captured using ConMon in both ends over WiFi.

Figure **??** represents the delay of the stream plotted in **??**. We can see that the quality of the call is affected by the network distortion at Figure **??**, this variation of the rate delivers a high delay of more than 4 seconds during some periods of the call and an average of approximately 1 second globally, the media received in those periods will not render correctly and the user experience of the call is going to be worst than at the beginning of the call.

WebRTC haven't performed as expected on this scenario providing big delays and low quality media when the status of the link is not optimal.

6.6 Interoperability

One of the main goals of WebRTC is that it should be interoperable between different browser vendors. In this chapter, we will analyze if it is possible for all browsers to be interoperable.

Actually, three different browsers carry native implementations for WebRTC: Google Chrome, Mozilla Firefox and Opera. Of those, two of them include the *GetUserMedia* and *PeerConnection* APIs: Mozilla Firefox and Google Chrome, they should be interoperable between them. Google Chrome has included WebRTC *PeerConnection* API for long time, but Mozilla Firefox came up with the *PeerConnection* API much later [?]. Making both browsers compatible required work from the vendors, because the SDP messages provided by them are different but rely on the IETF standard. Both engines should be able to understand those signaling messages and respond with the adequate SDP negotiation.

However, there are some differences in the SDP signaling messages, in the Firefox implementation, the implementation provides the STUN/TURN candidates bundled on the message meanwhile the API from Chrome provides the candidates separately by default, at the same time, the signaling messages from Chrome include some extra information about the streams that is not given in the Firefox version [?].

Furthermore, both browsers must implement cross-compatible codecs to send the stream and be able to handle the proper incoming RTP packets. Even though, most compatibility issues are solved, there are still some ongoing problems with the actual JavaScript method calls in both browsers [?].

One of the goals when checking the interoperability of the browsers, is to evaluate the enabled congestion mechanisms already implemented in both APIs. They should be able to manage the different environments in the same way as they rely on top of the same standards. At the time of this document, not all congestion control mechanisms are included in Mozilla Firefox.

For this test we have executed a point-to-point call between a Firefox and Chrome in different machines. We are more interested in seeing the behavior of the call rate and other metrics between the peers than the possible overall averaged result.

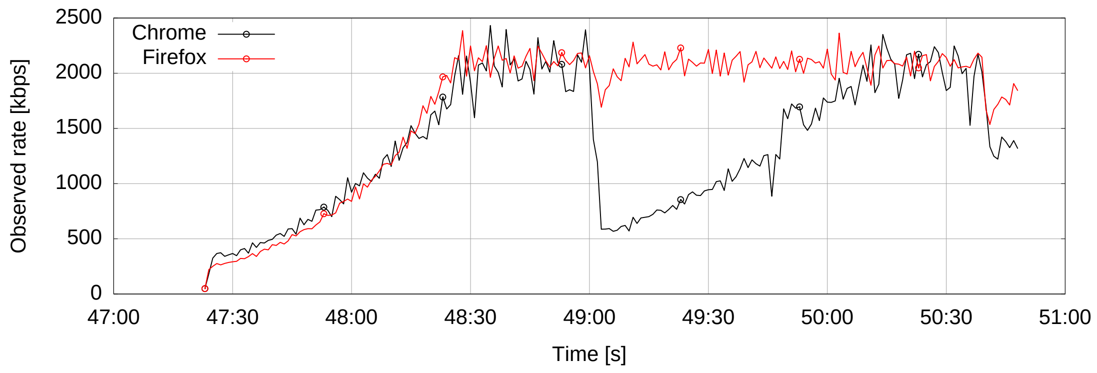


Figure 50: Remote stream rate for a point-to-point call between Mozilla Firefox and Google Chrome.

Figure **??** illustrates the rate given by the streams of both vendors captured at each receiver. We can observe that congestion mechanism is not triggered for the Firefox stream when some congestion occur on the path, Chrome applies some rate adaptation meanwhile Firefox maintains the same stream rate over the link.

We should consider that WebRTC should RTCP message with Receiver Estimated Maximum Bitrate (REMB) for rate adaptation [?] [?]. This mechanism provides an extra field in the WebRTC RTCP messages with the estimated total available bandwidth for the session.

To check the behavior of the RTCP feedback mechanism in WebRTC, we have captured different samples using a network sniffer. However, it is important to advise that some of the congestion mechanisms are still in ongoing discussion in the working groups and have not reached full consensus.

Firstly, we obtained a capture from a call between two Chrome browsers, after analyzing the data, we can observe that both peers where exchanging *Sender Reports* control packets. In WebRTC, the RTT measurement of the RTCP packets is made by using the same RTP media flow timing, this avoids the usage of *Receiver Reports* in most point-to-point or single *PeerConnection* topologies. WebRTC multiplexes the RTP and RTCP packets in the same port to avoid extra usage of network resources [?].

Those messages carry important data required by the WebRTC internals to build the *Stats API*. The following fields (**??**) are extracted from a RTCP packet sent by a Google Chrome browser in our test. We can see that most of the metrics are available in the packet, other required metrics which are not fully standardized as REMB are not able to be decoded by our packet sniffer [?].

Listing 7: RTCP message exchange between Chrome and Firefox

```
Real-time Transport Control Protocol (Sender Report)
    10.. .... = Version: RFC 1889 Version (2)
    ..0. .... = Padding: False
    ...0 0001 = Reception report count: 1
    Packet type: Sender Report (200)
    Length: 12 (52 bytes)
    Sender SSRC: 0xdf1a474d (3743041357)
    Timestamp, MSW: 3026830625 (0xb469c521)
    Timestamp, LSW: 653452974 (0x26f2e6ae)
    [MSW and LSW as NTP timestamp: Dec 1, 1995 18:17:05.152143000 UTC]
    RTP timestamp: 973093429
    Sender's packet count: 4236773172
    Sender's octet count: 3803919253
    Source 1
        Identifier: 0x9046a1ac (2420548012)
        SSRC contents
            Fraction lost: 102 / 256
            Cumulative number of packets lost: 4832630
        Extended highest sequence number received: 1896484047
            Sequence number cycles count: 28938
            Highest sequence number received: 3279
        Interarrival jitter: 1420789294
        Last SR timestamp: 2622976836 (0x9c577344)
        Delay since last SR timestamp: 2032955356 (31020436 milliseconds)

Real-time Transport Control Protocol (Receiver Report)
    10.. .... = Version: RFC 1889 Version (2)
    ..0. .... = Padding: False
    ...0 0001 = Reception report count: 1
    Packet type: Receiver Report (201)
    Length: 7 (32 bytes)
    Sender SSRC: 0xf46245fb (4100081147)
    Source 1
        Identifier: 0x7f3343fb (2134066171)
        SSRC contents
            Fraction lost: 217 / 256
            Cumulative number of packets lost: 3472040
        Extended highest sequence number received: 4229724701
            Sequence number cycles count: 64540
            Highest sequence number received: 31261
        Interarrival jitter: 1181356458
        Last SR timestamp: 2930563385 (0xaeacd939)
        Delay since last SR timestamp: 3634988546 (55465523 milliseconds)
```

Furthermore, some other features are required to be answered by the WebRTC RTCP engine, but might not be asked by the *PeerConnection* if they are not necessary or implemented in one of the peers. This is why features such as REMB may still not provided the RTCP *Sender Report*, if this field is not available, the congestion mechanism of Chrome calculates the estimated rate by using the mechanisms provided in the RMCAT Internet-Draft [?], those mechanisms use the information extracted from the RTCP report (**??**).

During the capture with different vendors we have observed a different behavior in the RTCP mechanisms between Chrome and Firefox. Meanwhile Chrome continuously provide the *Sender Report* metrics in the RTP/RTCP channel, Firefox is only reporting *Receiver Reports* back to the source, this message exchange procedure is seen in the previous description(**??**). No metrics from the outgoing stream of Firefox are being sent to Chrome, this forces Chrome not to provide any feedback control messages to Firefox that would trigger congestion control mechanisms. This behavior may affect the rate adaptation mechanisms in Firefox which could lead to an output similar to Figure **??**.

We can state that congestion mechanisms on Firefox are still not available and this reject any rate adaptation in Firefox, in conclusion, using Firefox in multiple scenarios would lead to unexpected rate response and poor call quality. Apart from the congestion control mechanisms, Firefox and Chrome APIs are fully interoperable to perform calls even though some features, such as statistics, are still not available.

6.7 Summary of results

After all the performed tests we can conclude that WebRTC is a young protocol for real time communication that still has long way to go in terms of congestion control to adapt to all topologies, still is a reliable and interesting method that uses all existing technologies previously developed for other projects there is a lot of work to do in terms of scalability and environment adaption.

Considering the existing WebRTC congestion mechanisms we can conclude that is still not ready for low tolerance networks or to be used simultaneously in different calls, the resource consumption of the existing WebRTC API is still something to care if we want to integrate WebRTC in mobile devices.

In multiple peer connections environments the usage of an MCU is something that should be considered in order to reduce load and increase performance, the actual WebRTC API does not allow to natively use multiplexing of streams over the same peer connection but the roadmap of the standardization protocol includes this feature for future versions of WebRTC.

Delay response for some low latency environments is not suitable for production applications and scenarios should be considered by developers prior deciding to use WebRTC instead of other competitors.

7 Conclusion

The end.

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A Setting up fake devices in Google Chrome

To address the issue in the video that is transferred from our automated devices we have built a fake input device on the virtual machines that will be fed with a RAW YUV video of different resolutions and quality. This device will be added by using a hacked version of the *V4L2Loopback* which derives from the *V4L* driver for Linux, the modified version of the *V4L2Loopback* builds two extra devices as Chrome is unable to read from the same reading/writing device for security reasons, one of them will be used to fed the video and the other one to read it [?].

Differences between standard driver and modified version:

- Need to write a non-null value into the the bus information of the device, this is required as Chrome input needs to be named as a real device. When using Firefox this is not required but works as well.

```
strlcpy(cap->bus_info, "virtual", sizeof(cap->bus_info));
```

- Our driver will pair devices when they are generated, this will create one read device and one capture device. Everything written into */dev/video0* will be read from */dev/video1*.

```
cap->capabilities |= V4L2_CAP_VIDEO_OUTPUT | V4L2_CAP_VIDEO_CAPTURE;
```

We used the code provided by Patrik Hglund [?] for the *V4L2Loopback* hacked version.

```
# make && sudo make install
# sudo modprobe v4l2loopback devices=2
```

Now we should be able to see both devices in our system, next step is feeding the */dev/video1* with a YUV file. In order to do this we will use the *V4l2 File Player* [?], this player executes on top of *Gstreamer* but adds a loop functionality to the file allowing long calls to succeed. Sample videos can be obtained from a Network Systems Lab.[9]

```
# sudo apt-get install gstreamer0.10-plugins-bad libgstreamer0.10-dev
# make
# v4l2_file_player foreman_cif_short.yuv 352 288 /dev/video1 >& /dev/null
```

We can now open Google Chrome and check if the fake device is correctly working in any application that uses GetUserMedia API.

[9]http://nsl.cs.sfu.ca/wiki/index.php/Video_Library_and_Tools

B Modifying Dummynet for bandwidth requirments

Dummynet is the tool used to add constraints and simulate network conditions in our tests.

Besides this, *Dummynet* has been natively developed for *FreeBSD* platforms and the setup for *Linux* environments is sometimes not fully compatible. Our system runs with Ubuntu Server 12.10 with a 3.5.0 kernel version on top of VirtualBox, this system requires to modify some variables and code in order to achieve good test results.

The accuracy of an emulator is given by the level of detail in the model of the system and how closely the hardware and software can reproduce the timing computed by the model [?]. Considering that we are using standard Ubuntu images for our virtual machines we will need to modify the internal timer resolution of the kernel in order to get a closer approximation to reality, the default timer in a Linux kernel 2.6.13 and above is 250Hz [?], this value must be changed to 1000Hz in all machines that we intend to run *Dummynet*. The change of timing for the kernel requires a full recompile of itself. This change will reduce the timing error from 4ms (default) to 1ms. This change requires the kernel to be recompiled and might take some hours to complete.

Once the kernel timing is done we will need to compile the *Dummynet* code, the version we are using in our tests is 20120812, that can be obtained form the *Dummynet* project site [?].

We should try the code first and check if we are able to set queues to our defined pipes, this part is the one that might crash due to system incompatibilities with FreeBSD and old kernel versions of Linux. If we are unable we should then modify the following code in the *./ipfw/dummynet.c* and *./ipfw/glue.c*.

```
Index: ipfw/dummynet.c
===================================================================

if (fs->flags & DN_QSIZE_BYTES) {
        size_t len;
        long limit;

        len = sizeof(limit);
        limit = XXX;
        if (sysctlbyname("net.inet.ip.dummynet.pipe_byte_limit", &limit,
            &len, NULL, 0) == -1)
                limit = 1024*1024;
        if (fs->qsize > limit)
                errx(EX_DATAERR, "queue size must be < \%ldB", limit);
} else {
        size_t len;
        long limit;

        len = sizeof(limit);
        limit = XXX;
        if (sysctlbyname("net.inet.ip.dummynet.pipe_slot_limit", &limit,
            &len, NULL, 0) == -1)
                limit = 100;
        if (fs->qsize > limit)
                errx(EX_DATAERR, "2 <= queue size <= \%ld", limit);
}
```

The problem arises from a the misassumption of *sizeof(long) == 4* in 64-bit architectures which is false. By changing those two files we are modifying the system in order to accept higher values than 100 for the queue length.

```
Index: ipfw/glue.c
===================================================================

char filename[256];   /* full filename */
char *varp;
int ret = 0;          /* return value */
long d;

if (name == NULL) /* XXX set errno */
        return -1;


        fprintf(stderr, "\%s fopen error reading filename \%s\n",
            __FUNCTION__, filename);
        return -1;
}
if (fscanf(fp, "\%ld", &d) != 1) {
        ret = -1;
} else if (*oldlenp == sizeof(int)) {
        int dst = d;
        memcpy(oldp, &dst, *oldlenp);
} else if (*oldlenp == sizeof(long)) {
        memcpy(oldp, &d, *oldlenp);
} else {
        fprintf(stderr, "unknown paramerer len \%d\n",
        (int)*oldlenp);
}
fclose(fp);


        fprintf(stderr, "\%s fopen error writing filename \%s\n",
            __FUNCTION__, filename);
        return -1;
 }
if (newlen == sizeof(int)) {
        if (fprintf(fp, "\%d", *(int *)newp) < 1)
                ret = -1;
```

```
} else if (newlen == sizeof(long)) {
        if (fprintf(fp, "\%ld", *(long *)newp) < 1)
                ret = -1;
} else {
        fprintf(stderr, "unknown paramerer len \%d\n",
                (int)newlen);
}
fclose(fp);
```

When doing this we are making the file compatible with systems that have compatibility problems with the *sysctlbyname* function, XXX should be the value of the queue maximum length in slots and Bytes. Slots are defined considering a maximum MTU size of 1500 Bytes.

By default, maximum queue size is set to 100 slots, this amount of slots is not designed for bandwidth demanding tests such as 10Mbit/s or similar. In order to modify this we will need to set a higher value according to the maximum we require. Once this is set we need to recompile *Dummynet* form the root directory of the download source code and follow the install instructions in the README file attached to the code.

Even we have allowed *Dummynet* to accept more than 100 slots we won't be able to configure them into the pipe even the shell does not complain with error. The next step is to modify the module variables set in the */sys/module/ipfw_mod/parameters* folder, this folder simulates the *sysctl* global variables that we would have running *FreeBSD* instead of Linux.

We need to modify the files *pipe_byte_limit* and *pipe_slot_limit* according to the values set in the *dummynet.c* previously modified.

Last convenient step is to add ipfw_mod to the end of */etc/modules* file so *Dummynet* module will be loaded even time the system starts.

We can now set large queues according to our needs.

C Scripts for testing WebRTC

Listing 8: Script for testing WebRTC with 15 iterations

```
#!/bin/bash
#

#First argument will define the name of the test, second the video to use
#and third may define the IPERF configuration if used

#We also will use this as test example modifying some parameters for the
#other examples such as parallel and mesh


echo "" > 1to1.log
#Exporting variables required for the test
echo "Exporting variables"
PATH="$PATH:/home/lubuntu/MThesis/v4l2_file_player/"
PASSWORD=lubuntu

#Timers for the call duration and break time after the call
REST_TIMEOUT=30
TIMEOUT=300

INIT_TIME=$(date +"%m-%d-%Y_%T")

#Define folders to sabe files
backup_files="/home/lubuntu/MThesis/ConMon/rtp/rtp_*"
mkdir results/$INIT_TIME"_"$1
dest_folder="/home/lubuntu/results/"$INIT_TIME"_"$1
echo "Starting $INIT_TIME"
counter=0

#Loop the test 15 times to avoid call failures
while [ $counter -le 14 ]
do
        actual_time=$(date +"%m-%d-%Y_%T")
        echo "Iteration - $counter"
        #Clean all ongoing processes from previous iterations
        echo "Cleaning processes"
        echo $PASSWORD | sudo -S killall conmon >> 1to1.log 2>&1
        killall v4l2_file_player >> 1to1.log 2>&1
        killall chrome >> 1to1.log 2>&1
        sleep $REST_TIMEOUT

        #Set virtual device for Webcam
        echo "Setting dummy devices"
        echo $PASSWORD | sudo -S modprobe v4l2loopback devices=2 >>
            1to1.log 2>&1

        cd MThesis/ConMon
        #Start ConMon and configure 192.168.1.106 which is the turn relay
            for the media
        echo $PASSWORD | sudo -S ./conmon eth3 "udp and host 192.168.1.106"
            --turn >> 1to1.log 2>&1 &
        cd ../..

        #Load fake video into virtual device
        echo "Loading video"
        v4l2_file_player /home/lubuntu/MThesis/v4l2_file_player/$2 352 288
            /dev/video1 >> 1to1.log 2>&1 &
        #If third argument available then we run the IPERF
        if [ $# -eq 3 ]
        then
                        iperf -c 192.168.1.106 -t 300 -i 5 -b $3 >> 1to1.log
                            2>&1 &
        fi

        #Load browser pointing the test site with the n= parameter that
            will define the StatsAPI filename
        #We need to ignore the certificate errors to load the page with an
            untrusted certificate
        DISPLAY=:0 google-chrome --ignore-certificate-errors
            https://192.168.1.100:8088/?n=$1"_"$counter >> /dev/null 2>&1 &

        #Script for capturing CPU and Memory usage for every test
        ./memCPU.sh $dest_folder $counter >> 1to1.log 2>&1 &
        memCPUPID=$!

        sleep $TIMEOUT
        echo $PASSWORD | sudo -S killall conmon >> 1to1.log 2>&1
        kill $memCPUPID
        dir_file=$1"_"$counter
        mkdir $dest_folder/$dir_file
        mv $backup_files $dest_folder/$dir_file
        (( counter++ ))
done

sleep 30
echo "Finishing test..."
echo $PASSWORD | sudo -S killall conmon >> 1to1.log 2>&1
killall v4l2_file_player >> 1to1.log 2>&1
killall chrome >> 1to1.log 2>&1
```

Listing 9: Measue and store CPU and Memory usage

```
#!/bin/bash

#Script used to measure periodically the status of the CPU and memory
PREV_TOTAL=0
PREV_IDLE=0

#Runs until the script is killed by another process
while true;
do
        CPU=(`cat /proc/stat | grep '^cpu '`) # Get the total CPU
            statistics.
        unset CPU[0]                        # Discard the "cpu" prefix.
        IDLE=${CPU[4]}                      # Get the idle CPU time.
        timeStamp=$(date +%s)
        # Calculate the total CPU time.
        TOTAL=0
        for VALUE in "${CPU[@]}"; do
                let "TOTAL=$TOTAL+$VALUE"
        done

        # Calculate the CPU usage since we last checked.
        let "DIFF_IDLE=$IDLE-$PREV_IDLE"
        let "DIFF_TOTAL=$TOTAL-$PREV_TOTAL"
        let "DIFF_USAGE=(1000*($DIFF_TOTAL-$DIFF_IDLE)/$DIFF_TOTAL+5)/10"

        # Remember the total and idle CPU times for the next check.
        PREV_TOTAL="$TOTAL"
         PREV_IDLE="$IDLE"

        #Save the amount of used memory in Mb
        total=$(free |grep Mem | awk '$3 ~ /[0-9.]+/ { print $2"" }')
        used=$(free |grep Mem | awk '$3 ~ /[0-9.]+/ { print $3"" }')
        free=$(free |grep Mem | awk '$3 ~ /[0-9.]+/ { print $4"" }')

        #Calculate the percentage
        usedmem=`expr $used \* 100 / $total`

        #Export all the data to the defined iteration in argument 2 and
            folder 1
        echo  $timeStamp"    "$DIFF_USAGE" "$usedmem"    "$total"
            "$used" "$free >> $1/log_performance_$2.txt

        # Wait before checking again one second
        sleep 1
done
```