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

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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 browsers may lead 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 in other technologies are implemented in those browsers to handle the internals of WebRTC adding complexity to the system but hiding it from the application developer. 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 current 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 current 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

First of all, I would like to thank Prof. Jörg Ott for allowing me to join his research team and all the time he spend going through this thesis giving very valuable feedback. Huge thanks to my friend and supervisor Varun Singh for proposing this topic and all the time he spent teaching me and giving good ideas, at the same time as solutions, for this thesis. Varun has been a very important guidance for the development of the thesis and an accessible pool of knowledge.

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Lastly, thank to you reader. If you are reading this one line after the other, you at least have read one page of my thesis. Thank you.

Otaniemi, 15.8.2013

Albert Abelló Lozano

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## Abbreviations

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 rapidly changing, the dramatical increase of Internet communication between people is forcing technology to support mobile and real-time video experiences in different ways. Applications such as *Vime* and *Skype* provide media and real-time communication over the Internet.

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. Business structure and innovation is rapidly changing thanks to the existence of reliable, low cost and simple platforms for real-time communication.

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 [5]. At the same time, this is changing traditions and habits of communication between people and transforming personal relationships [6]. 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 such as gaming or using social networks.

Furthermore, it also helps businesses to have lightweight communication alternatives that will increase their efficiency whatever the size of the company.

Real-time communication is encouraging front end designers and device developers to turn their products into multi-functional and interoperable 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 (e.g DSL, 3G or EDGE) while affordable digital and video cameras have become integral parts of present day computers. Those two enablers along with the increased demand for richer applications with easy integration 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 instead of requiring extra software to be installed. With this simple approach, WebRTC aims to transform something that has been traditionally complex and expensive into 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 using a standardized, free technology.

Many web services already use RTC technology to allow communication (e.g., Google Hangouts and Adobe RTMFP) but 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 free to use and openly standardized.

## 1.1 Background

WebRTC is an effort to bring real-time communication APIs to JavaScript developers, allowing them to build RTC functionalities into web applications that enable different features, such as video calling platforms over the web.

Web application APIs are defined in 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 APIs works by combining two different technologies, HTML and JavaScript, HTML is the de facto markup 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 present version of WebRTC is formed between different API that integrate with each other to provide flexible RTC in the browser. However, the final goal is to allow developers to create plugin-free real-time web applications that are cross compatible with different browser vendors and operating systems.

The flexibility of WebRTC allow many uses of RTC thus foreseeing many new topologies that will emerge. Developing RTC technologies over the Internet has been always closely related to the topology distribution of the nodes that are being connected, performance issues in different topologies usually restrict the amount of nodes available for the session. From the simple point-to-point topology to the mesh composition, choosing the right option for each application is very important to deliver a great user experience.

## 1.2 History

WebRTC API is being drafted by the World Wide Web Consortium (W3C) alongside 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 made in a working group in May 2011 [7], and the official mailing list started in April 2011 [8]. 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 on 27th October 2011 [9]. This first W3C draft only specifies how to send media (audio and video) 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 got involved with the IETF in May 2011 [10]. 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 are the definition of the communication model, session management, security, NAT traversal solution, media formats, codec agreement and data transport [11]. In June 2011 Google publicly released the source code of their WebRTC API implementation [12].

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.

### 1.3 Challenges

WebRTC is a suite of protocols that share the available device resources with many other applications. Due to the little experience in WebRTC environments sharing the available resources, we find some lack of documentation or previous literature regarding congestion analysis compared with other technologies. During the development of the thesis we focus in the technical challenges of the protocol.

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 that could change in future versions of WebRTC, even though it should not affect the overall 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 technical problems. 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. During 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.

On the other side, we will also study the available topologies for RTC and which one of them fits in a WebRTC environment, trying to understand the limits of the protocol in different scenarios and topologies. All the structures that can be implemented using WebRTC might have some restrictions in performance and user experience that we need to study and understand, those limitations are important when designing WebRTC applications.

## 1.4 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.

## 1.5 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 being 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.

## 1.6 Structure

This thesis is structured as follows:

1. Introduction of real-time communication
2. Description of different protocols used for real-time communication and the APIs that are built within WebRTC
3. Definition of different possible topologies used for real-time communication
4. Analysis of the required metrics used to evaluate WebRTC performance
5. Environment setup used to evaluate WebRTC
6. Result and analysis of the different scenarios that can be given in WebRTC applications
7. Conclusions of the thesis

## 2 Real-time Communication

Real-time Communication is defined as any method of communication where users can exchange data packets (e.g media, text, etc.) with low latency in both direction. The purpose of RTC is widely seen as a way to communicate between people. This is done in a two-way scenario where both users are senders and receivers of media packets, live video is a one-way configuration with one unique source of data and one or multiple receivers. In the first RTC configuration, latency is very important in order to achieve good quality for bidirectional communication between both users whereas the live scenario can tolerate some latency in the link. In two-way communication data can be transmitted using multiple topologies, they are either peer-to-peer or using a centralized relay.

Some other ways of transmitting data include multicast or broadcast. In the development of this thesis we do not study multicast and broadcast streaming.



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

Figure 1 shows 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 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. The signaling channel is used to negotiate the codec agreement and NAT traversal methods that are used 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 starts to flow to the receiver, this stream may include media (audio or video) and different types of data (e.g binary, text, etc.). During the transmission we may also require some extra signaling messages to be exchanged in order to maintain the path or adapt the constraints to the present network conditions, at the same time, features might change during the session.

### 2.1 Session Initiation Protocol (SIP)

SIP is a protocol used to create, modify and terminate multimedia sessions. This protocol features real-time communication between different peers with multiple

optional extensions, those extensions allow the usage of instant messages or subscriptions to different events. SIP final Request for Comments (RFC) was published in June 2004, this document describes the original functionalities and mechanisms of SIP [1].

Other features of SIP is the ability to invite participants to already existing sessions in order to build multicast conferences. SIP also gives support for name redirection being a federated protocol regardless of the user network location.

The process of SIP includes the user location, availability, media capabilities, setup of the session and management of itself. On the other side, SIP can also be defined a suite of tools that are built together with other existing protocols such as Real-time Transport Protocol (RTP) , Real-time Transport Streaming Protocol (RTSP) , Session Description Protocol (SDP) and Media Gateway Control Protocol (MEGACO) .

SIP provides low level services to deliver messages between users, for example, it could deliver SDP messages to be negotiated between the endpoints in order to agree on the parameters of a session.

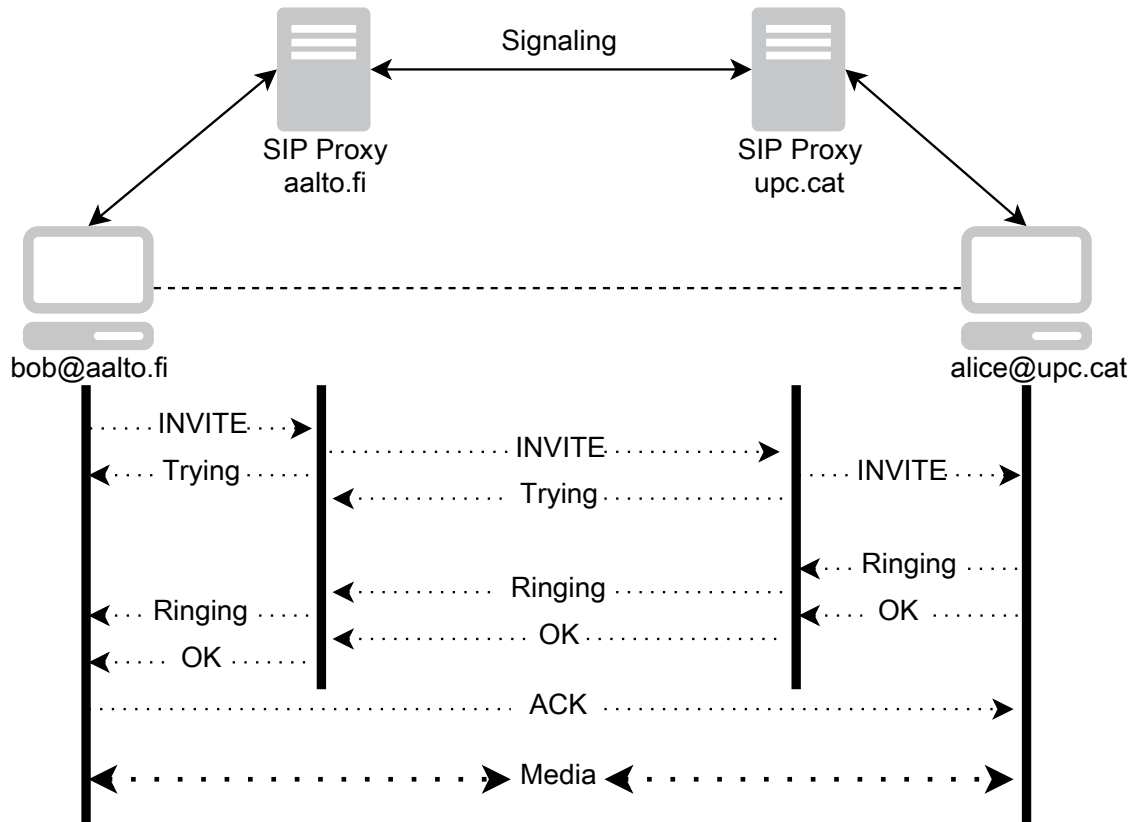


Figure 2: SIP session establishment example. Source [1].

Figure 2 shows a typical example of a SIP scenario with a message exchange between two endpoints. SIP can use SDP Offer/Answer model for the session description or capabilities negotiation between the end-points [13] and RTP for the media transport.



On the other side, SIP also relies on some elements called proxy servers that help to route requests to the final destination. Those proxies are represented in Figure 2.

SIP uses a wide range of methods that help the protocol to understand the type of message that is exchanging between peers. Figure 2 shows an example of message exchange between endpoints. INVITE request is an indicator to the receiver that someone is trying to contact, Trying indicates that the INVITE has been received and the proxy is trying to find the correct path to the destination, Ringing message represents hold for answer from the other peer. Finally, once the receiver chooses to answer, an OK message is generated to indicate that the media session can start.

The different media parameters are negotiated using the SDP bodies transmitted into the SIP methods previously mentioned. Those parameters are agreed between the peers in order to provide compatibility.

This protocol has been used for some time and improved due to many iterations and additions to itself. The knowledge raised from this technology helped to have a better understanding of the requirements when building real-time media protocols such as WebRTC.

## 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.

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 on the computer.

Adobe uses a proprietary programming language 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. Flash Player is available for most platforms, except iOS devices, and is present in about 98% of all Internet-enabled browser devices. This plugin allows developers to access media streams using external devices such as cameras and microphones to be used along with RTMFP.

This protocol is implemented by using Flash Player, Adobe Integrated Runtime (AIR) and Adobe Media Server (AMS) [14].

RTMFP uses Adobe Flash to provide media and data transfer between two end points over UDP [14]. RTMFP requires a plugin to be installed in order to be functional being a proprietary protocol. It also handles congestion control over the path and NAT transversal issues. One of the biggest differences is that, compared to SIP, RTMFP does not provide inter-domain connectivity and both peers must be in the same working domain to be able to communicate.

Media transfer is encrypted, this feature is provided in RTMFP by using proprietary algorithms and different encryption methods. RTMFP architecture allows

reconnection in case of connectivity issues and works multiplexing different media streams over the session. On the signaling part, Adobe uses an application server called Cirrus [15] (Figure 3) to handle the signaling between the different participants of a session. This service provides support to handle different topologies such as: end-to-end, many-to-many and multicast. Those structures rely on the use of overlay techniques in multicast and mesh scenarios.

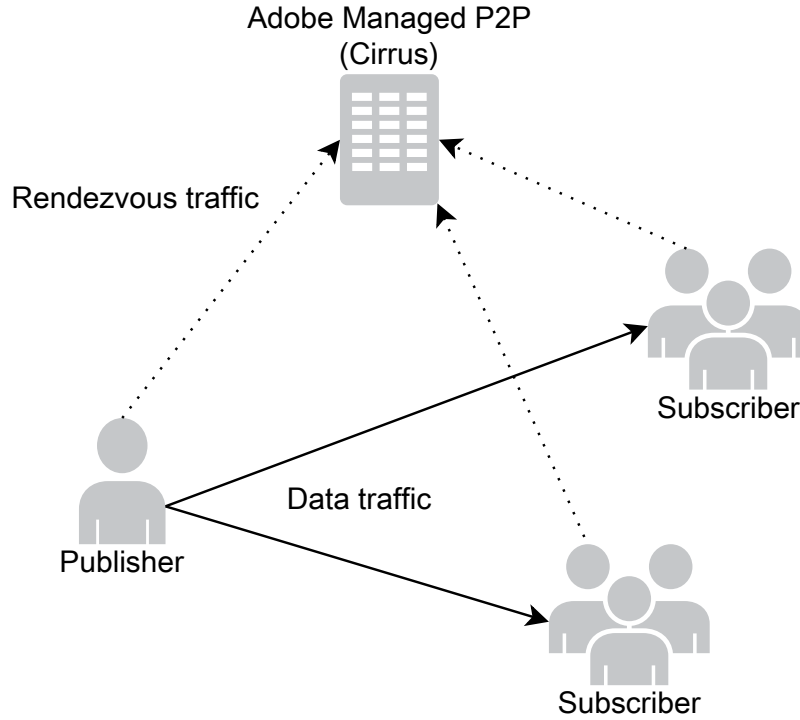


Figure 3: RTMFP architecture using Cirrus.

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

### 2.3 Web Real-Time Communication (WebRTC)

WebRTC is part of the HTML5 proposal, it is defined in the W3C [7] [3], and enables RTC capabilities between Internet browsers using simple JavaScript APIs, providing video, audio and data P2P without the need of plugins. This API is in the process of replacing the need for installing a plugin to enable P2P communications between browsers, WebRTC uses existing standardized protocols to perform RTC.

WebRTC provides interoperability between different browser vendors, this allow the APIs to be accessible by the developers assuring high degree of compatibility (Figure 4). Some of the major browsers that include some WebRTC APIs are: Google Chrome, Mozilla Firefox and Opera. Other providers, such as Internet Explorer, are in process of building prototypes for WebRTC [16].

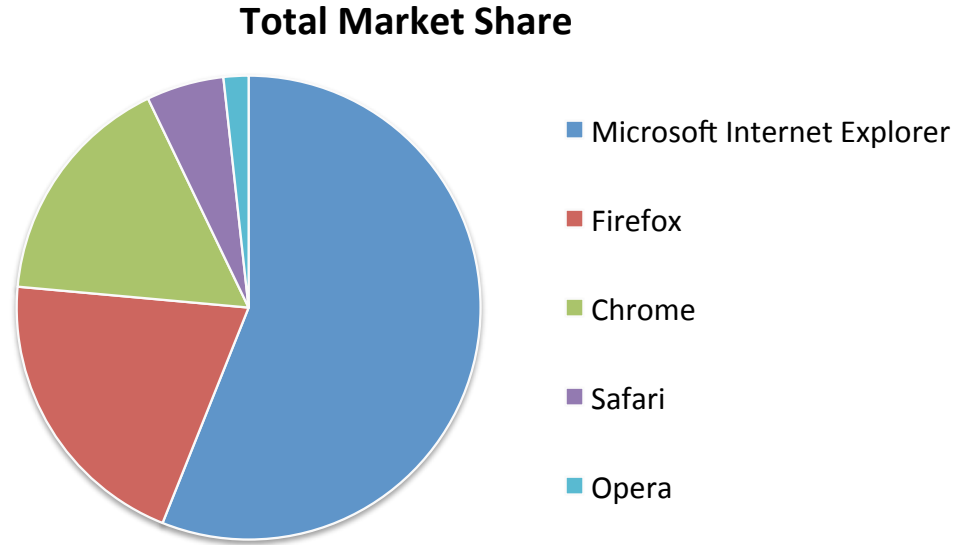


Figure 4: Market share of browser vendors by April 2013 [2].

With WebRTC, developers can provide applications for most of the desktop devices available, mobile devices will integrate WebRTC as part of their HTML5 package to also enable RTC soon [17].

WebRTC is composed of two important APIs that enable real-time features, `getUserMedia` and `PeerConnection`. Both of them are accessible by JavaScript on the browser.

### 2.3.1 `getUserMedia` API

WebRTC applications use the `getUserMedia` API to allow access to media streams from local devices (video cameras and microphones).

This API itself does not provide RTC, but provides the media to be used as simple HTML elements in any web application.

`getUserMedia` API allows developers to access local media devices using JavaScript code and generates media streams to be used either with the rest of the *PeerConnection* API or with the HTML5 video element for playback purposes [3]. `getUserMedia` is already interoperable between Google Chrome, Firefox and Opera [18].

`getUserMedia` proposal removes the need for using Adobe Flash to access the media device and also the plugin requirement.

Figure 5 illustrates how the browser access the media and delivers the output to JavaScript. We use the `getUserMedia` API to build WebRTC-enabled applications for RTC video conferencing. The video tag is an HTML5 Document Object Model (DOM) element that reproduces local and remote media streams.

In Figure 5 *MediaStream* is the object returned by the `getUserMedia` API methods, this object contains *MediaStreamTracks* that carry the actual video and audio media. The goal of using this architecture is to be able, in the near future, to include multiple sources of video and audio multiplexed over the same stream from differ-

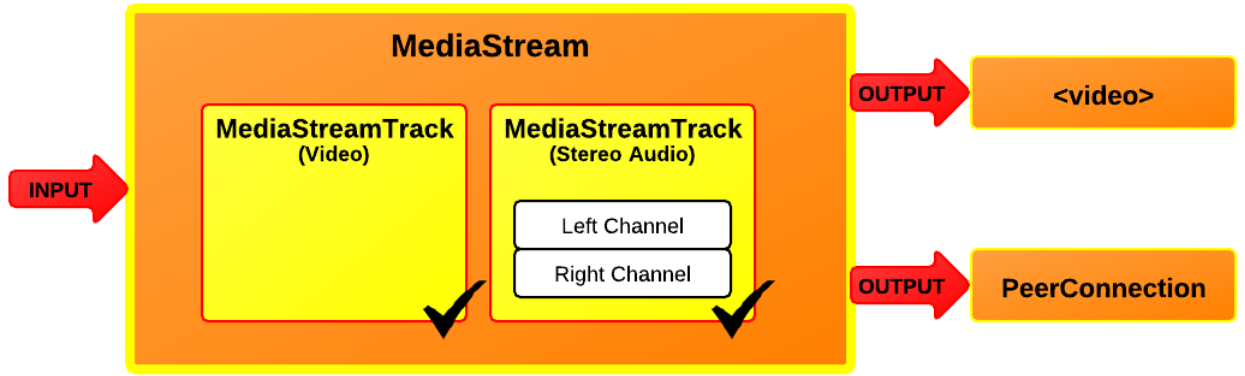


Figure 5: Media Stream object description. Source [3].

ent devices. Different alternatives include the implementation of overlay topologies by forwarding media from one peer to the other by including different *MediaStreamTracks* into the same *MediaStream*. Furthermore, *MediaStreams* handle the synchronization between all the *MediaStreamTracks* included for proper playback at the application level, by this it assures that audio and video will be always synchronized.

*getUserMedia* API works using a JavaScript fallback method, this method returns a *MediaStream* object to the application that is played in the HTML web application or used in the *PeerConnection* API. A sample example of this method can be seen in Listing 1.

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

---

```

getUserMedia(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 =
        createObjectURL(stream);
}

```

---

In Listing 1, we are using the video and audio media from our devices to be played in an HTML video element identified as *local-video*.

*getUserMedia* API also allow developers to set some specific constraints for the media acquisition. This helps applications to better adapt the stream to their requirements, those *cameraConstraints()* are provided by a JavaScript Object Notation (JSON) library.

### 2.3.2 PeerConnection API

WebRTC uses a separate API to provide the networking support to transfer media and data to the other peers, this API is named *PeerConnection* [19]. *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.

Figure 6 describes the topology used in WebRTC for a bi-directional media session, with the messages being sent either by WebSockets or by HTTP long polling. 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.

SDP is widely used in SIP to provide media and NAT reversal negotiation between two different endpoints prior to establish data transmission. This protocol is used in WebRTC in a modified version that allow the usage of multiple media descriptions over a single set of Interactive Connectivity Establishment (ICE) . Usually, in other conditions, different media types will be described using different media descriptions.

This new feature is described as Bundle [20] and can be used along with the existing SDP Offer/Answer mechanism to negotiate the different media ("m=" lines) on the session.

By using Bundled SDP, WebRTC multiplexes all the traffic using a single port, 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 [21]. *PeerConnection* API provides signaling and NAT transversal techniques, this part is very important to guarantee a high degree of success when establishing calls in different environments.

*PeerConnection* P2P session establishment system works in a constrained environment designed to provide some degree of legacy for other SDP based technologies. Figure 6 shows how a WebRTC simple P2P scenario works, the server used for signaling is a web server. WebRTC scenarios do not work easily in a federated environment such as SIP.

On the other side, signaling is not standardized in WebRTC and has to be provided in the application level by the developer.

---

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);
}

```

Figure 6 does not show relay servers that provide NAT transversal solutions described in Section 2.3.5 . When developing a WebRTC application, those servers must be provided into the WebRTC *PeerConnection* configuration when starting a new call as seen in Listing 2.

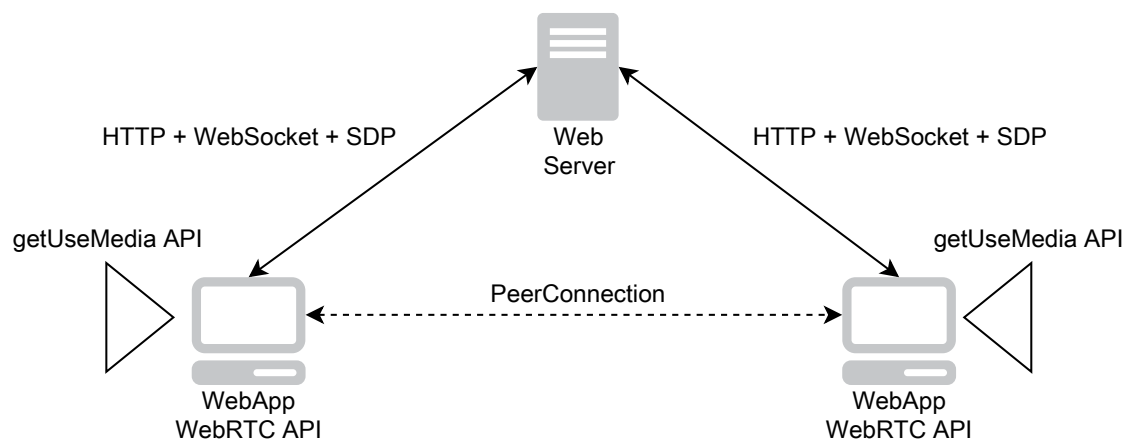


Figure 6: WebRTC simple topology for P2P communication.

Listing 2 represents a simple example of 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 2.3.1. When building the new *PeerConnection* object we need to pass the JSON object *server* with the stun configuration for the NAT transversal process: `var pc_config = {"iceServers": [{ "url": "stun:XXXX" } ]};`.

### 2.3.3 Control and Monitoring

Control and monitoring is an important part of all RTC protocols, this part is usually handled by the JavaScript API.

Media constraints are defined as a set of parameters that limit the media quality when processed from the devices such as microphone or webcam, those parameters are usually related to video size or frame rate in WebRTC. However, in WebRTC we can also adapt the maximum link rate through a special set of constraints.

Those parameters are implemented through the Statistics Model and Constraints defined in the W3C draft [19], these set of methods are part of the actual *PeerConnection* API defined in section 2.3.2. 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 from the remote side. We focus on those remote stats to study the status of the path and to obtain the desired metrics for monitoring [22].

To access the statistical data retrieved from the control messages we need to use the *getStats()* method of the *PeerConnection* object defined in the draft [19], this method allow the application 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*.

Within constraints, developers are able to change media capture configuration by setting parameters such as 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 in Pixels for Width and Height. Frames per Second in
minFrameRate
var constraints = {
    "audio": true,
    "video": {
        "mandatory": {
            "minWidth": "300",
            "maxWidth": "640",
            "minHeight": "200",
            "maxHeight": "480",
            "minFrameRate": "30"
        },
        "optional": []
    }
}
//Bandwidth in kbps
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 about the complexity level of the API, how much is available to be accessed by the developers and which configurations or mechanisms should be automated in the browser. After long discussion, WebRTC is now using Javascript Session Establishment Protocol (JSEP) [23], this API is a low level API that gives the developers control of the signaling plane allowing each application to be used in specific environments.

The media processing is done in the browser internals but most of the signaling is handled in the JavaScript plane by using JSEP methods and functions. Figure 7 shows 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.

Furthermore, JSEP also handles the state management of the session by building the specific SDP message that is forwarded to the other peer. NAT traversal mechanisms are activated in JSEP also, those mechanisms are described in chapter 2.3.5.

Another 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 [23].

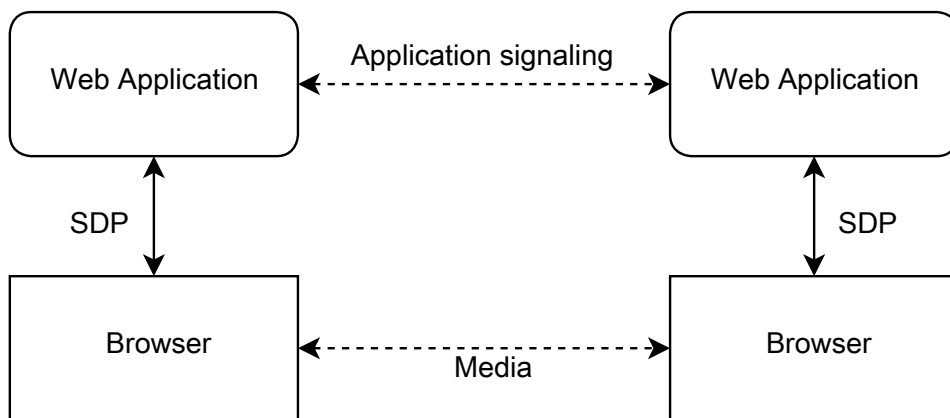


Figure 7: JSEP signaling model.

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 is that there are the multiple possibilities to adapt WebRTC to the requirements of each specific application disregarding its advantages or disadvantages without being sensitive to pick one specific design at a time.

### 2.3.5 Internals of WebRTC

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

One of WebRTC main issues is NAT transversal difficulties, this problem usually affects all RTC related technologies. Interactive Connectivity Establishment (ICE) is a technique that helps WebRTC to decide which is the best way to bypass NATs and firewalls, ICE is widely used in media communications and has proven to be reliable when choosing the best option to enable connectivity in restrictive environments [24]. The enablers that work together with ICE for Real-time protocols are Simple Transversal Utilities for NAT (STUN) and Traversal Using Relays around NAT (TURN) [25] [26].

TURN and STUN servers are usually placed outside the local network of the clients and help them to find the way to communicate with each other by discovering new open paths, the final decision is taken by the ICE mechanism. STUN server function is to discover the available IP addresses and ports that allow direct connectivity to a target machine placed behind a firewall or NAT, those interfaces are named *candidates*, this information is provided to the sender that processes it in order 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 connection to no other *candidate* was established. 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. In case of TURN the traffic can be sent over TCP also.

Media encoding in WebRTC is done through codecs implemented inside the browser. Mandatory-to-Implement codecs for audio are G.711 and Opus. G.711 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 G.711, 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 [27]. Opus can be easily adjusted for high and low encoding rates, applications can use additional codecs.

Along with the codecs, the audio engine for WebRTC also includes some features such as Acoustic Echo Cancellation (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 (local loop), 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 working properly, the rate required by the audio channel is reduced as the unnecessary noise is removed from the spectrum. AEC and NR mechanisms provide a smooth audio input for WebRTC protocol.

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 [28]. Non-media data in WebRTC is transferred using System Control Transmission Protocol (SCTP) encapsulated over Datagram Transport Layer Security (DTLS) [29] [30] [28].

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.

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 [4]. Figure 8 illustrates the full protocol stack for WebRTC described in this chapter.

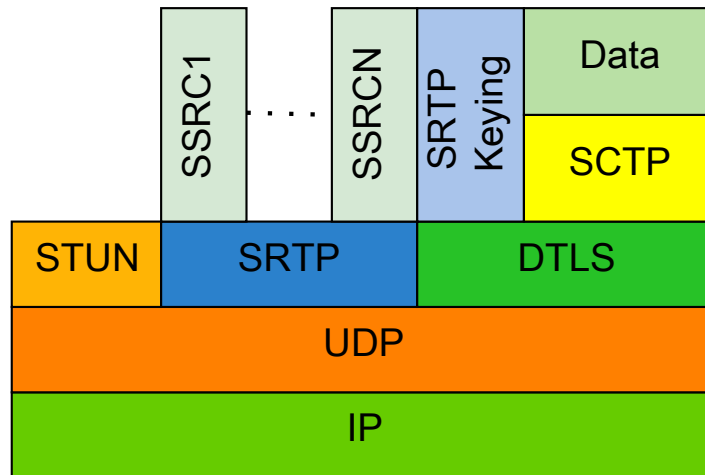


Figure 8: WebRTC protocol stack. Source [4].

Quality of Service (QoS) for WebRTC is also being discussed in the IETF and a draft is available with some proposals [31]. WebRTC uses DiffServ packet marking for QoS but this is not sufficient to help prevent congestion in some environments. When using DiffServ, problems that arise are originated on the Internet Service Providers (ISPs) as they might be using their own packet marking with different

DiffServ code-points, those packets are not interoperable between ISPs, there is an ongoing proposal to solve this problem by building consistent code-points [32]. Otherwise, clients might also be sending too much data for the specified path reducing the effectiveness of DiffServ. Each specific application will mark Audio/video packets with the designed priority using DSCP mappings [31].

Officially there is no congestion control mechanisms for WebRTC, the only mechanism actively used are circuit breakers for RTP [33].

Furthermore, Chrome specifically uses a Google congestion control algorithm that enables congestion control mechanisms for rate adaptation [34]. The aim of this algorithm is to provide performance and bandwidth sharing with other ongoing conferences and applications that share the same link. This algorithm is defined in Section 6.

### 2.3.6 Security concerns

To handle the signaling process WebRTC uses a web application server, peers exchange messages with 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 [35]. Figure 6 presents a simple topology for a WebRTC call, the web application 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. Considering that WebRTC APIs are able to bypass Firewalls and NAT, Denial of Services (DoS) attacks can also become a threat. On the other side, malicious JavaScripts could also perform calling to unknown devices.

Browsers execute JavaScript scripts provided by the web applications, this may include malicious scripts, that in the case of WebRTC could lead to 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 could execute a variety of actions in that browser. At a minimum, it should not be possible for arbitrary sites to initiate calls to arbitrary locations without user apprehension [36]. 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 [37]. Those situations are given when a third-party server provides JavaScript scripts to a different domain to the one accessed. This script cannot be trusted by the original accessed domain as it could trigger browser actions that might harm privacy. For example, in WebRTC, the user could load a malicious script from a third-party entity in order to automatically build a WebRTC call to an undesired receiver without the user noticing this situation. 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 include the possibility to establish media forwarding to a third peer, for example, once the user has accepted the access 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.

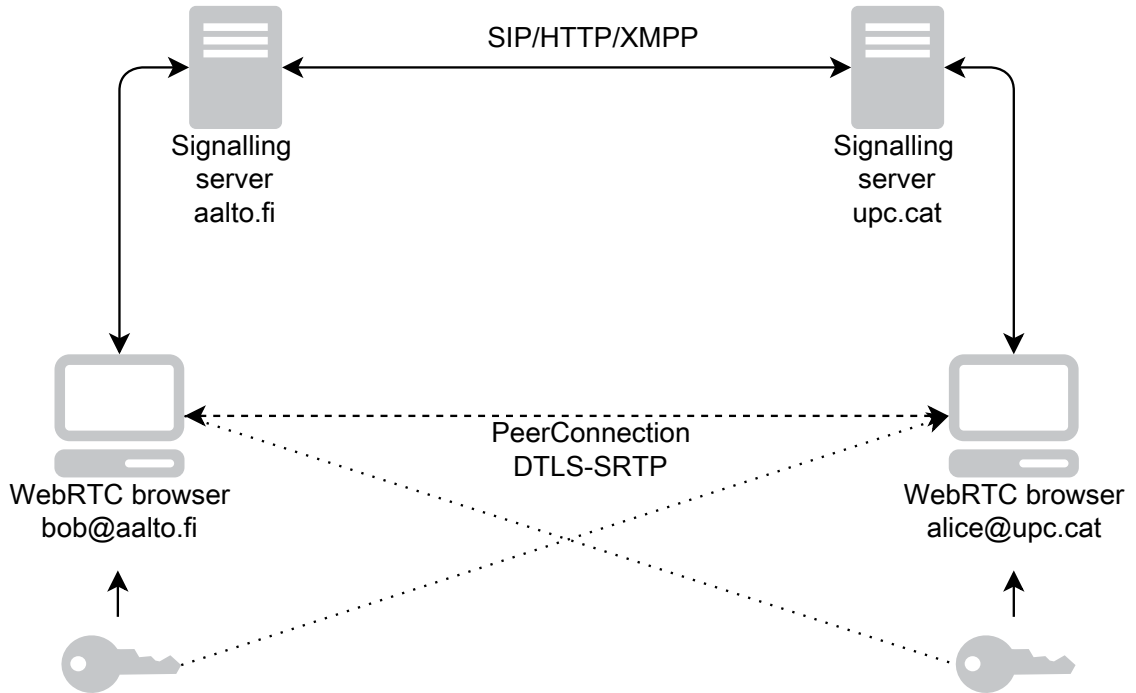


Figure 9: WebRTC cross-domain call with Identity Provider authentication.

Calling procedure is done using the JavaScript provided by the server, this may be a problem as the user must trust an unknown authority provider. WebRTC calling services usually rely on Hypertext Transfer Protocol Secure (HTTPS) for authentication where the 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 [38]. 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 9.

If the web service is running over a trusted secure certificate and has 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 API builds the ICE candidates for media verification. Authentication and verification in WebRTC is an ongoing discussion in the working groups.

Security and 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 from the users and store them for further usage [35].

## 2.4 Comparison between SIP, RTMFP and WebRTC

After describing various RTC technologies and two important alternatives for WebRTC, Table 1 is a summary of common features between SIP, RTMFP and WebRTC. In this Table 1, common internal mechanisms are described for all of them.

	SIP	RTMFP	WebRTC
Plugin-enabled	No	Yes	No
Cross-domain	Yes	No	Maybe
Audio	Yes	Yes	Yes
Video	Yes	Yes	Yes
Data	Yes	No	Yes
NAT Traversal	Yes	Yes	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	Yes	No	Yes
SCTP	Yes	No	Yes

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

RTFMP is a proprietary protocol which means that it might have its own mechanisms other than the standardized ones stated on Table 1 to solve some of the issues.

All the protocols explained in this section are designed to provide similar real time features but in different ways, meanwhile SIP is a protocol that helped to develop some of the important technologies, such as RTP and SRTP, that are used in other technologies, is still not easily accessible for web developers. On the other side, RTMFP provides a licensed alternative for real time communication having some mechanisms not standardized and with compatibility issues between devices.

From the mobile perspective, SIP is used in mobile technology and WebRTC has announced to be compatible with future versions of iOS and Android [17]. 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 for NAT traversal 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.

## 2.5 Summary

We can conclude that real-time media protocols have been developed over quite a long period. However, a standardized open source solution has not been provided yet. WebRTC and SIP share a basement of principles that work together, we could state that WebRTC could not exist as it is now without the previous knowledge provided by SIP. Besides, the usability of SIP in web applications is still very complex and prohibitive for most web developers.

Furthermore, RTMFP has shown more ubiquitous availability on user's devices than any specific web browser. The main problem with RTMFP approach is that the protocol for end-to-end media path is proprietary, so interoperating with existing VoIP solutions can be inefficient and developers rely on vendor's plugins to take care of any platform incompatibilities.

WebRTC solution may not be perfect but is a good start to provide interoperable real-time solution for web applications.

### 3 Topologies for real-time multimedia communication

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

A topology is defined as the arrangement of the various nodes of a network together. Nodes can be connected through different links and configurations.

Network topologies are determined by network protocols as opposed to by physical cables, wires or devices. We can think of a topology as a virtual network shape or structure. This shape does not usually match the physical layout but focuses on the logical map of the network. For example, computers in a building can be arranged physically in a circle, but it would be highly unlikely to find them defined as a ring topology.

Topologies are composed by multiple devices used to extend cable connections, concentrate interfaces, convert data formats and security. Commonly, routers are used to concentrate multiple connections and provide connectivity to a Wide Area Network . They have the responsibility of routing data packets from the source to the destination within the Local Area Network or WAN.

However, routers may also occur to provide some issues as they usually activate security mechanisms that affect WebRTC. 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, described in Section 2.3.5. However, in some restrictive environments it might be impossible to succeed with the call establishment.

The usage of NAT traversal mechanisms in WebRTC is crucial and at the same time increases the complexity of the browser internals (Section 2.3.5). STUN and TURN servers must be reachable from the browser in order to provide the best *candidates* that are evaluated by the ICE mechanism.

All the mentioned mechanisms provide high level of success but might fail in very restrictive environments. To solve some of the issues, WebRTC allows UDP and TCP (using TURN) packet transport, this is done in order to enable connectivity even in very restrictive environments that may have UDP packet drop mechanisms.

For some topologies that include the establishment of multiple *PeerConnections*, resource usage can be a big problem (e.g. mesh, one-to-many or tree). Considering that system capacity relies on 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 differently. Media encoding usually consume most of those resources becoming a bottleneck for some scenarios.

In this section we describe the following topologies for WebRTC: Point-to-point, one-to-many, many-to-many, usage of MCU and overlay networks. Those are some of the most common used topologies. However, some of them are still not possible to implement with the current WebRTC API.

### 3.1 Point-to-Point

The simplest topology is a communication between two nodes or endpoints. This model is widely used in telephony and provides simple real time communication between users. In WebRTC, point-to-point topologies work only within people in the same domain in contrast to cross-domain communication alternatives such as SIP. Scenarios such as the one shown in Figure 2 are difficult to design in a WebRTC application, on the other side, Figure 6 presents the most common WebRTC point-to-point scenario.

There can be two different types of point-to-point topologies: dedicated or switched. Dedicated topologies appears to the user as a permanently associated communication channel between two endpoints. Within many switched telecommunications system it is possible to establish a permanent circuit. The resources in such connection can be released when no longer needed.

On the other side, switched networks are the most common network topology for WebRTC. It is defined as a point-to-point circuit that can be set up a dynamic path thanks to packet-switching technologies and dropped when no longer needed.

Uses for point-to-point topologies in real-time communications are usually related to bidirectional communications between two endpoints. Other uses can also be given in data transmission between devices. Traditional radio links also rely on point-to-point topologies.

### 3.2 One-to-Many

One-to-many or star topologies are one of the most common network topologies for synchronous and asynchronous media streaming (e.g Windows Media Server or RTMFP). This kind of topology consists of a central node that transmits streams to the rest of nodes connected to it. In the WebRTC example of Figure 10, 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.

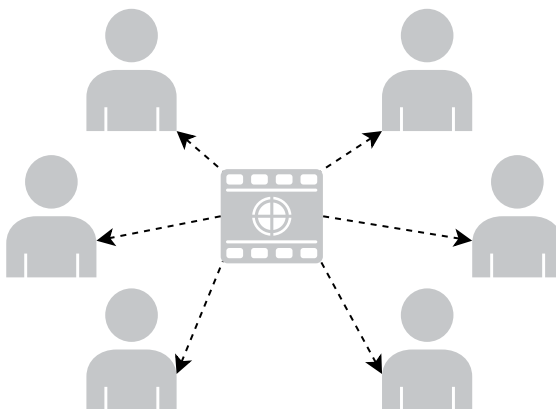


Figure 10: Multi-unicast topology for real time media.

This central node provides a connection point for all nodes. This type of topology reduces the change of network failure by connecting all the clients to a central device.



When this happens each client can fail individually without affecting the rest of the network. However, this might also produce high dependency on the central node. If this node fails, all the connectivity is lost. All the peripheral nodes may communicate with all others by transmitting to, and receiving from, the central device. This behavior can be given when using rebroadcasting.

One-to-many topologies can provide some advantages. Better performance, as the packets are not processed by an excessive number of nodes before being delivered. Resiliency against endpoint failures, this isolation also prevents non-centralized failure from affecting the rest of nodes. Star topologies also make it easier to detect and localize faulty nodes. Furthermore, there are no disruptions when adding or removing endpoints on the network.

On the other hand, this topology relies on high dependence of the central node that sends the traffic. If this node fails, the network becomes inoperable. The central node will also act as a bottleneck, calculating its capacity is required when seizing the network.

In a WebRTC star topology we have a video, audio and data provided from one source to multiple endpoints. This might cause a huge load on the source when having multiple *PeerConnection* running, central node bottleneck behavior can be a constraint in this scenario. Compared with other topologies, media delay on the network is not a good indicator for this topology due to the one-way communication only. In some scenarios we may not require data to be received on the source. Those scenarios are one-way only use cases.

Common uses for star scenarios are related with video and audio streaming to multiple peers, TV media and conferencing. Live streaming is a common use for internet TV. For example, media providers could use this topology when sending media streams from the source to multiple endpoints. This could allow their subscribers to join any channel when desired.

Other WebRTC solutions could cover communication between employees with an HTML5 web application. Music bands also could take advantage of this scenario by being able to transmit their 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, current media streaming technologies do not provide this kind of communication between the viewer and the origin.

### 3.3 Many-to-Many

Many-to-many topologies are also known as full mesh. In this type of topology, each network node is interconnected with every other node. Each endpoint sends and receives data from each other node of the network.

In a full mesh topology all peers connect between themselves increasing the number of connections and used resources. The value of fully meshed networks depend on the number of subscribers, the number of *PeerConnections* established in a mesh network is dependent on the amount of people in the conference. The number of *PeerConnections* in a full mesh topology can grow rapidly proportionally

to the square of amount of the nodes.

Full mesh topologies provide some advantages. They can expand and be modified without disrupting other nodes. Furthermore, data can be transmitted from different devices simultaneously. On the other hand, the overall cost of this network is high compared with other topologies. Maintenance of this topology is also very difficult and traffic load is high.

This topology is used in wireless and backbone networks. WebRTC uses are related to video conferencing. Conferencing systems are widely extended in enterprises for long-distance communication between employees and working groups. It can also be used to transfer data between peers as a file delivery network.

### 3.4 Multipoint Conferencing Unit (MCU)

Multipoint Conferencing Unit (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 WebRTC video conferencing applications.

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.

MCUs receive the streams from the clients and multiplex them over one unique channel, this provides good scalability from the client perspective because it is only building one connection even though there are multiple peers on the conference.

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

Drawbacks of the MCU model are related to the dependency of the end nodes on 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 depend on the number of participants per conference and the number of conferences going through the MCU. A scalable implementation is needed when handling multiple conferences. A different alternative is the usage of cascaded MCUs to provide more scalability.

### 3.5 Overlay

Overlaying media streams is the ability of a peer to forward data to a third party. Systems that implement overlay are those that require the media to be forwarded from one peer to the other. Multiple topologies can be constructed using overlaying. Some examples are *hub-spoke* or *tree*, seen in Figure 11.

We can combine all of the following topologies to build an infrastructure that fits our requirements.

In WebRTC we can use overlay to forward media streams from one node to another. This feature can be very useful when building scalable applications. We can design multiple layers of nodes that transmit the media relaying to each other. Other alternatives would be the usage of overlaying in mesh or one-to-many topologies. In those environments, nodes could forward the stream to provide reliability on the network. However, WebRTC does not provide native support for media overlay yet, but it is planned to implement those features in future versions of the API.

Overlay scenarios have some disadvantages. Topologies that rely on this technique usually are slow distributing the data, this produces long latency. Furthermore, overlay can also provide duplicate packets at certain points.

Overlaying is widely used in audio and video conferencing, multi-party games and content distribution protocols. Considering WebRTC it could also be implemented in group communication applications.

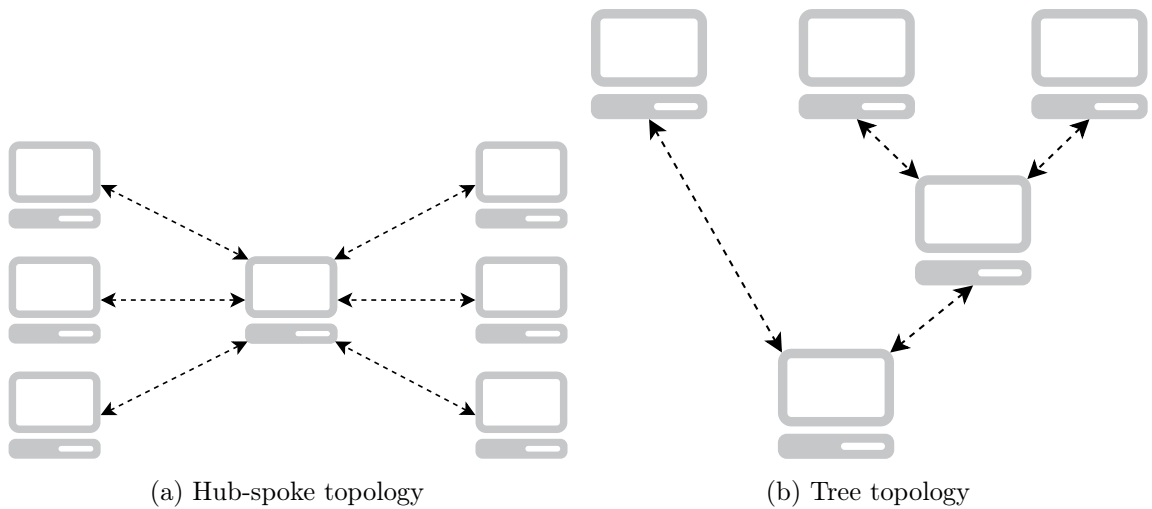


Figure 11: Overlay topologies.

### 3.5.1 Hub-spoke

Hub-spoke distribution is a topology composed of multiple nodes and arranged like a chariot wheel. Traffic flows along spokes that are connected to the hub at the center. This type of topology, represented in Figure 11a, behaves well in some scenarios as it requires less connections to perform a full communication in the network.

On this type of topology, hubs are designed to forward traffic from one node to another. However, those devices are not always dedicated machines. In some specific protocols it is possible that hubs are common endpoint devices selected to act as dynamic hubs. For example, in Skype, user computers can act as hubs forwarding calls to other nodes. This happens when the response of those devices is very good, then the protocol selects the best candidate and triggers it to operate as a hub in the network. From then this endpoint becomes a node that forwards data to other peers.

Hub-spoke model has some advantages. The amount of paths to connect all nodes is significantly less than other topologies. Having less amount of paths also makes the topology to be more efficient in terms of resources. Lastly, new spokes can be created easily without affecting the rest of nodes in the network much.

However, due to a centralized hub model, the reliability of the network is based on this central node. Previous statements of dynamic hubs can help to reduce the dependency of central nodes. Resources have to be seized optimally to avoid overload of the hub.

### 3.5.2 Tree

Tree topology is based on a node hierarchy, the highest level of the tree consists 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 number of end users as seen in Figure 11b.

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. The expansion of this type of network is easy.

On the other hand, 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 from different layers.

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.

## 3.6 Summary of topologies

Different topologies can be used for real-time communication. This section has described the ones that can adapt to WebRTC in different use cases. However, some of them are still not possible to implement due to technology constraints.

During this thesis we focus our study in point-to-point, many-to-many and MCU topologies. Those two topologies can adapt to WebRTC application requirements.

## 4 Performance Metrics for WebRTC

This section describes metrics to monitor the performance of WebRTC in the different topologies described above.

WebRTC behavior can be measured by using performance indicators. In the following sections we are going to describe the metrics used to evaluate the response of WebRTC in multiple scenarios.

WebRTC performance can differ in different environments. When using real-time communication there are many indicators for performance, we use the following metrics for this thesis. There are two different type of metrics, metrics related to network performance and those measuring the status of the endpoints.

The usage of metrics for congestion control is crucial. Congestion control mechanisms are used to adapt the sender characteristics to the path conditions. This path condition is evaluated using metrics. Once the metrics are processed, congestion control mechanisms react and adapt the rate to the path status.

### 4.1 Simple Feedback Loop

Traditionally, real time media communication uses inelastic traffic, this type of traffic has some tolerance to error as packets can be discarded but media playback won't be seriously affected, this does not require having 100% packet delivery rate. Elastic traffic is common in applications that do not require real time data to be sent, it requires high packet delivery rate but is tolerant to delays.

Figure 12 shows a simplified feedback model for multimedia communication. The feedback provides messages with path condition information that helps the sender to adjust its congestion mechanisms. Those feedback messages are sent periodically by the receiver and are very important to report the status of the path.

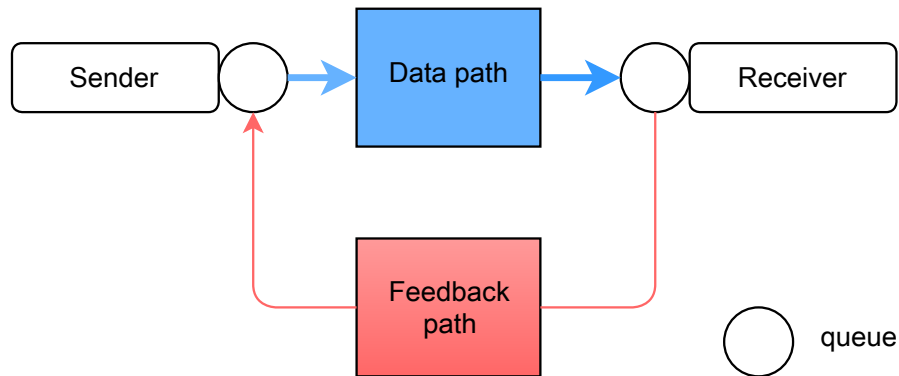


Figure 12: Multimedia feedback loop.

### 4.2 Network metrics

Metrics defined in this section are related to the status of the network, those metrics usually trigger the congestion control mechanisms in WebRTC changing the behavior

of the stream according to the constraints on the link [39] [34].

With real time applications, congestion affects protocol performance, if the media rate is lower than the available channel capacity there is no need for rate adaptation. However, losses and available capacity of the path can vary over time requiring adaptation from the sender side to match the new constraints. This adaptation is done by analyzing the receiver feedback packets sent to the sender using the simple feedback loop.

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

#### 4.2.1 Losses

Loss rate indicates packet losses during the transmission over the path. Usually packet losses directly affect the quality of a call. In WebRTC, packet loss is an indicator of the playback quality in the ongoing WebRTC transmission. In the case of huge packet loss in real-time media, the quality of the stream can be degraded producing artifacts on the playback.

When monitoring WebRTC calls we use packet loss as an indicator to adapt the media constraints. This indicator can be processed to be used along with the *GetUserMedia* constraints described in Section 2.3.3 to adequate the media acquisition to the path condition. For example, an increase of packet loss could trigger some JavaScript mechanism that monitors the call quality and takes decisions based on the reported metrics. Those decisions may affect the video size and frames per second used to acquire the media. This allows a better adaptation for the ongoing session with the conditions on the path.

WebRTC uses RTCP packets for control and monitoring of the ongoing stream [40]. In RTCP losses are reported in the feedback message. Packet losses can also appear when being discarded by the protocol itself, this might happen due to heavy delay on the path.

Losses are calculated in a period of time, Equation 1 calculates how much loss rate we have in a certain path. We subtract the packet loss given in a period and divide it between the difference of packets loss and received on that same period.

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

Equation 1 calculates the estimated packet loss we have on the link. This operation is done periodically by the statistical API described in Section 2.3.3.

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

The delay on a path can be measured in two different forms. One-Way Delay (OWD) indicates the time it takes for a packet to move from one peer to the other peer, this time includes multiple delays that are produced along the link. OWD is calculated from 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 queues on the path) and propagation delay (speed of light). The sum of all those delays compose the total one-way delay.

$$OWD = delay_{propagation} + delay_{queues} + delay_{serialization} + delay_{processing} \quad (2)$$

In WebRTC applications, one of the most important delays that we have to measure is the processing delay. WebRTC applications usually run 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. In the case of native mobile WebRTC applications, this delay is significantly lower than in browser applications.

Round-Trip Time (RTT) measurements are included in the standard RTCP specification, in order to calculate this metric, timestamp from sender and receiver is needed in the reports [40]. Sender Report (SR) 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 3.

$$RTT = TS_{RR} - TS_{SR} - T_{Delay}$$

$TS_{RR}$ : Local timestamp at reception of last Receiver Report

$TS_{SR}$ : Last Sender Report timestamp

$T_{Delay}$ : Receiver time period between SR reception and RR sending in the sender (3)

We can calculate the RTT using the data given by the *Stats API* in WebRTC. This metric is important to be monitored as it indicates the time delay along the path and the playback quality on the endpoints.

Calculating OWD requires both machines' clocks to be accurately synchronized and can be complicated. Assuming that delay is symmetric we can define OWD as  $\frac{RTT}{2}$ .

### 4.2.3 Throughput

Throughput is a key metric for measuring the performance of WebRTC applications. This indicator describes the capacity of the path taken by each *PeerConnection*.

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 number of packets that are injected into the network by the sender. Receiver rate is the ratio at which packets arrive at the receiver. Goodput is the result of discarding all the lost packets from the total number along the path, only packets that have been received are counted, goodput is a good metric to measure performance.

Typically, throughput is calculated by extracting the information from RTCP packets, in our case, we also rely on the Stats API included in WebRTC specification to calculate the rate. Taking into account the number of bytes received in the

previous and current period and the amount of time elapsed, we are able to calculate the throughput.

$$\frac{\Delta Bytes_{received}}{\Delta Time_{host}} \quad (4)$$

The throughput defines the bandwidth usage for video/audio in each endpoint, we can use this value to get an overall quality status of the call. A sudden drop of the rate means that the bandwidth available for that *PeerConnection* has been drastically reduced, this introduces problems in the transmission, or in the worst case, loss of communication between peers.

#### 4.2.4 Inter-Arrival Time (IAT)

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.

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 [34]. Jitter can be defined as the variation of IAT. The difference between when the packet is expected and when it is actually received is the jitter.

In WebRTC, IAT is given by the Equation 5, 6 and 7 [34]. This calculates the drift between the actual arrival time and the timestamp time of two packets (jitter), which should be approximately the same.

$$d(i) = t(i) - t(i-1) - (T(i) - T(i-1)) \quad (5)$$

$t$ : arrival time  
 $T$ : timestamp time

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

$$t_s = \frac{Size}{Capacity} \quad (6)$$

$Size$ : Frame size  
 $Capacity$ : Capacity of the path

The final model for the IAT in WebRTC can be defined as a function of serialization delay, queuing delay and network jitter.

$$d(i) = \frac{Size(i) - Size(i-1)}{Capacity} + w(i) \quad (7)$$

Here,  $w(i)$  is a sample  $i$  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  $w(i)$  is going to decrease. Alternatively we can consider  $w(i)$  equal to zero. IAT is usually calculated in the receiver and reported to the sender to enable 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 resources demanding encoding.

CPU usage in WebRTC is critical. The web browser is the one in charge of the encoding and decoding process. This internal engine produces a huge load on the CPU. The number of streams to handle will determine the status of the CPU. Having a normal load in the CPU helps WebRTC to perform better.

On the other hand, RAM usage is also considered a constraint in any mobile or old device. The amount of RAM used in each WebRTC call relies on the encoding process and the network activity of the device.

CPU/RAM usage is observed in order to calculate the stress on the host in WebRTC scenario. This information is crucial for the success of WebRTC in some topologies that demand large amount of resources.

#### 4.3.2 Setup time

We also measure the setup time required for each topology, with this we can possibly determine an average setup time for WebRTC sessions. The setup time is calculated using local timestamps when building the *PeerConnection* object. WebRTC calls are divided into three processes, media acquisition, stream transfer over the network and remote video playback. The protocol itself is only based on the network transfer. However, the total time to establish the session is obtained once the playback of the remote stream starts.

Once the media from the remote peer arrives, we can subtract both timestamps, start of the *PeerConnection* object and stream arrival, to check the elapsed time. This procedure is in the JavaScript layer.

#### 4.3.3 Call failure

Call failure is an important metric that measures the software quality of a WebRTC application. Compared with the previous metrics, this indicator do not relate to topology issues. This also fully depends on NAT situations and we will calculate

the average failed calls with STUN and TURN NAT technologies. This indicator can define the success ratio for establishing WebRTC calls.

#### 4.4 Summary of metrics

Performance metrics allow us to determine the behavior of the path. Those metrics are obtained by using the information provided in the RTCP messages and statistical APIs. We use them to evaluate each different scenario and reach a conclusion about the performance of WebRTC.

Throughput, delay and loss are important metrics in WebRTC. They will assist congestion control mechanisms when adapting the sender rate for the media. Those metrics also define the quality of the media session and are important in order to improve the user experience when using WebRTC applications.

From the monitoring perspective, we care about characterizing the performance and congestion control algorithm of WebRTC. We study how those mechanisms behave and try to reach a consensus whether it is a good alternative to use WebRTC or not. Lastly, some other metrics related to the host help us to define which devices might be ready to hold WebRTC applications.

## 5 Evaluation Environment

In this chapter we will describe the testbed we use for running all the WebRTC tests, the environment is described in Figure 13.

Some of the nodes carry specific software that enable some extra features or conditions that are required for deep testing.

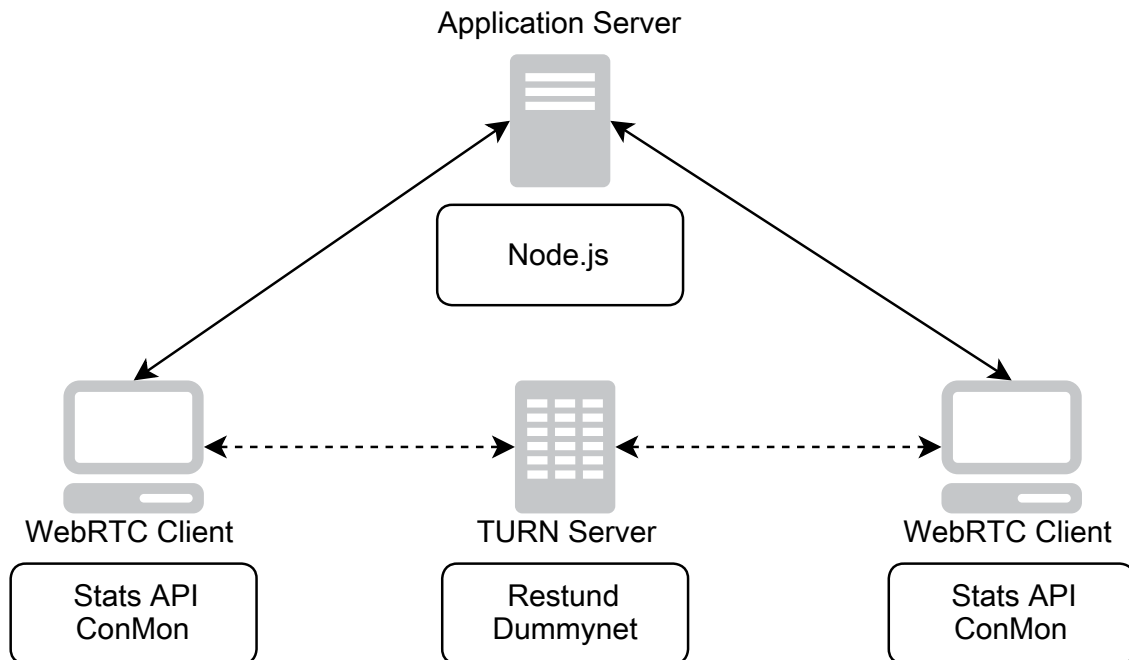


Figure 13: Description of simple testing environment topology for WebRTC.

With the testbed described in Figure 13 we are able to control the bottleneck link capacity, end-to-end latency, link loss rate and the queue size of the intermediate routers.

### 5.1 WebRTC client

WebRTC clients are virtual machines that run a lightweight version of Ubuntu (Lubuntu<sup>1</sup>) with 2GB of RAM and one CPU. This light version removes graphic acceleration providing better results in performance than compared with other distributions due to the virtualization of the graphic card.

Clients run Chrome Dev version 27.01453.12 as a WebRTC capable browser. We have disabled audio in our tests in order to avoid unexpected performance results due to a bug in the *Pulse Audio* module that controls audio input in WebRTC<sup>2</sup>. The amount of audio data transferred due to the echo cancelation systems can be neglected.

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<sup>1</sup><https://wiki.ubuntu.com/Lubuntu>

<sup>2</sup><https://bugs.launchpad.net/ubuntu/+source/pulseaudio/+bug/1170313>

This client will load the web that runs on the Application Server to handle the WebRTC calls.

### 5.1.1 Connection Monitor

Connection Monitor [*ConMon*] is a command line utility that works on the transport layer and uses *libpcap*<sup>3</sup> to sniff all the packets that go to a certain interface and port [41]. This utility is designed specifically to detect and capture RTP/UDP packets. *ConMon* detects and saves the header but discards the payload of each 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* at each endpoint. The *PeerConnection* carries real media so the testing environment is a real scenario of WebRTC usage.

*ConMon* captures are be saved into different files allowing us to plot separately different stream rates and calculate other parameters such as delay with post processing. *ConMon* allows us to compare network layer and JavaScript API monitoring tools, as *ConMon* is working directly over the network interface and avoids all the processing that the browser internals do to send the stats to the JavaScript statistical API. Figure 14 represents one video stream from the same call as Figure 15 captured from the *ConMon* application.

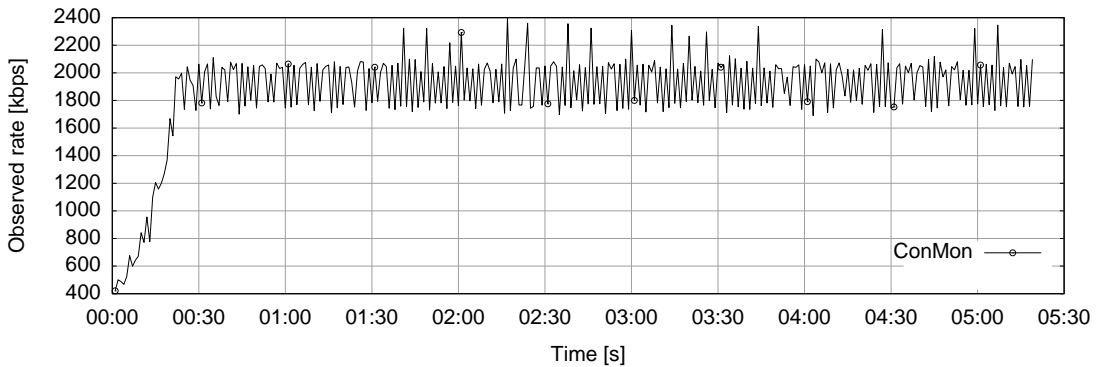


Figure 14: Point-to-point WebRTC video stream throughput graph using ConMon at the network layer.

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 each period of one second.

Furthermore, *ConMon* is used to provide OWD and RTT calculations for our tests, in order to do this we 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.

*ConMon* provides us with higher accuracy by subtracting the timestamps from both endpoints of the same stream, to obtain a good result in this test we have to

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<sup>3</sup><http://www.tcpdump.org/>

reduce the internal clock drift of both peers using systems such as Network Time Protocol daemon (NTPd) .

### 5.1.2 Stats API

WebRTC statistical API provides methods to help developers access the lower layer network information at the receiver, those methods return the information required to calculate the performance indicators used to build the 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 of both it interesting in order to check the differences between the browser API and an interface layer capture. By combining both methods we can verify the results and accuracy of the metrics.

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 [19] 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 [39].

*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 change slightly in the following versions of the WebRTC API. Methods involved in the *RTCStatsCallback* are available on the W3C editors draft [19].

We have built a high level *Stats API* that uses those statistics from the *RTCStatsCallback* to calculate the RTT, throughput and loss rate for the different streams that are sent over the *PeerConnection* during the call. Those stats can be saved into a file at the end of the call for further process. Our JavaScript API grabs any *PeerConnection* passed through the variable and starts a periodic iteration to collect those stats and, either plot them or save them for post-processing. Figure 15 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 but with unknown cross traffic on it. The measures were directly obtained from the *Stats API* we built and post-processed using *gnuplot*<sup>4</sup>.

Figure 15 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 in both directions 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 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.

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<sup>4</sup><http://www.gnuplot.info/>

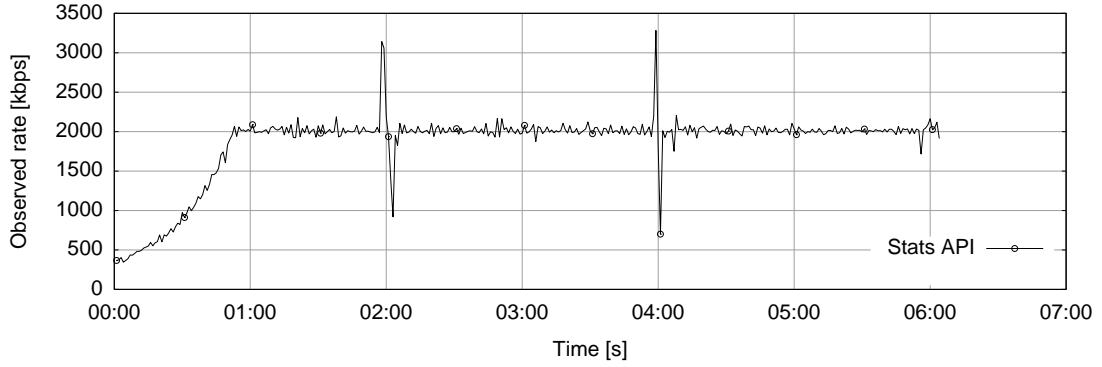


Figure 15: Point-to-point WebRTC video call total throughput graph using *Stats API* over public WiFi.

### 5.1.3 Analysis of tools

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

However, due to the periodical capture method, the output can produce strange plots as the information regarding to the next data period could be stored in the previous one when processing the averaged data on the system. 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.

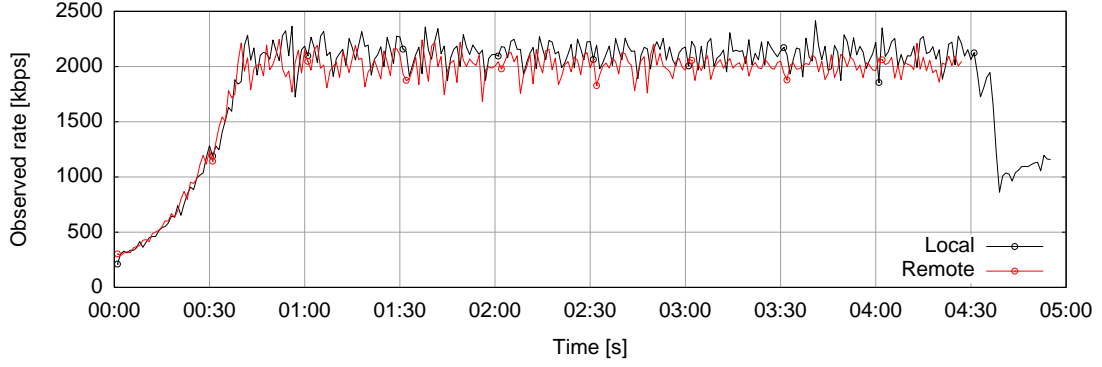
A second problem that we could face is the time it takes for the OS to process the stats from the RTCP packet and send them to the upper browser layer, at the receiver some of the stats are based on the current measurement of the metrics not in the RTP Receiver Report. Figure 16a and 16b plot two video streams being captured from *Stats API* and *ConMon*.

Figure 16 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 overhead is not considered when calculating the rate through *Stats API*. We can conclude that the real rate that WebRTC use is the one outputted by the result of *ConMon* instead of *Stats API*, but *Stats API* is going to be an accurate approach.

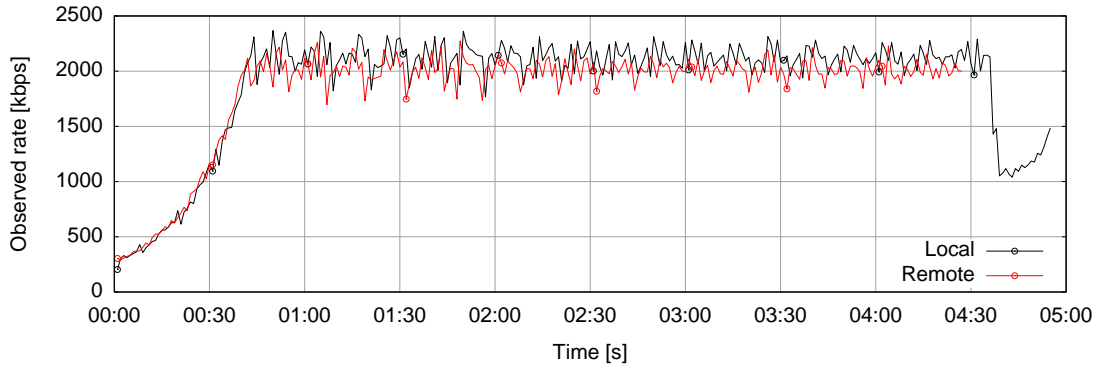
### 5.1.4 Automated testing

For our test scenario we considered two options, manual and automated testing. The first test environment does not give as much accuracy due to the difficulty 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 output. We try to automate the process in most of the tests.

In some environments, we won't be able to perform automated testing, when this happens the averaged results after the iterations will not show a leveraged value that



(a) Incomming stream.



(b) Outgoing stream.

Figure 16: P2P video stream comparison between *ConMon* and *Stats API*.

defines the behavior. This help us to deal with outliers that can provide false values due unexpected conditions.

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`<sup>5</sup> parameter, this video though, does not produce rate high enough for our purposes.

Figure 17 represents the approximate bandwidth that a real video call uses when receives media from another peer, that capture shows the same stream captured from the receiver *StatsAPI* perspective. The maximum rate increases up to 2000 Kbps. On the other hand, Figure 18 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 (17 and 18) show one single video 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.

Comparing global output from Figures 17 and 18, 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

<sup>5</sup><http://peter.sh/experiments/chromium-command-line-switches/>

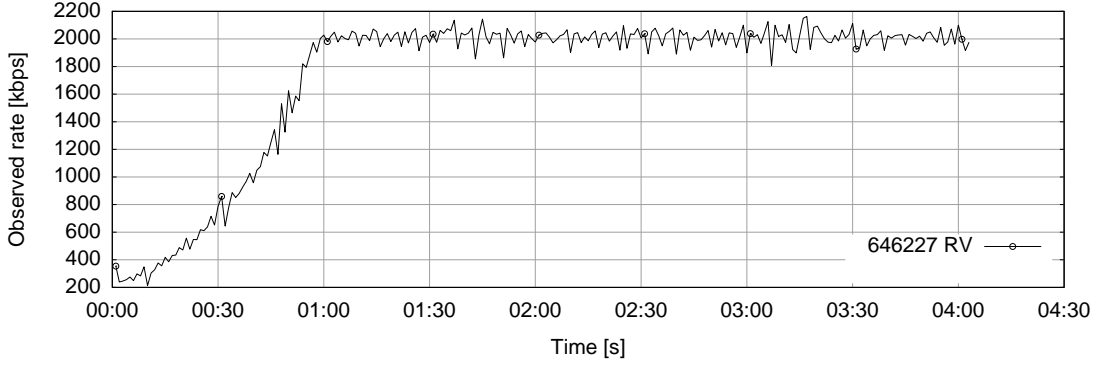


Figure 17: Video stream rate with SSRC 0x646227 captured using *Stats API* and webcam input.

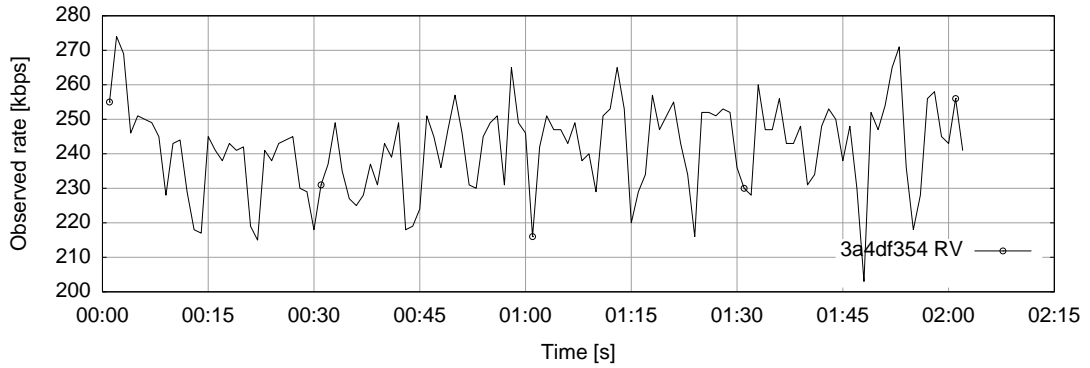


Figure 18: Video stream rate with SSRC 0x3a4df354 captured using *Stats API* and Chrome default fake video input.

uses an internal bitmap engine to draw the figures that are 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 not representative.

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 A. This technique utilizes a modified version of the *V4L2loopback* linux driver to create two extra video devices that play a selected YUV file.

Figure 19 represents the bandwidth of a fake video stream measured by our *Stats API* using an YUV<sup>6</sup> 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 19 and 17, both average rate output is approximately 2000 Kbps, which means that this procedure is a good approach to a real webcam.

The combination of the previous fake video setup and multiple Secure Shell

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<sup>6</sup>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.



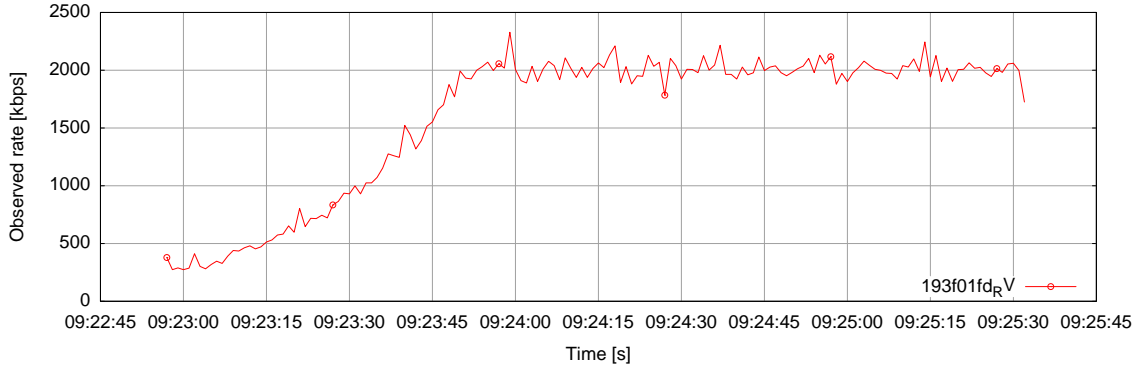


Figure 19: Video stream bandwidth using V4L2Loopback fake YUV file.

scripts enables the automation mechanisms to run multiple tests without the need of multiple physical devices.

## 5.2 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 *Dummysnet*.

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 [42]. We have modified the source in order to have a hardcoded password making it easier for our needs.

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"}]
};
```

---

Listing 5 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 called every time we get a new candidate from 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.

### 5.2.1 Dummynet

To evaluate 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 allows us to add bandwidth limitations, delays, packet losses and other distortions to the ongoing link [43].

*Dummynet* is a standard tool for some Linux distributions and OSX [43]. 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 *Virtual-Box*<sup>7</sup> machines for some testing and for running TURN instance, read Appendix B

---

<sup>7</sup>VirtualBox is an x86 virtualization software package.

for the fixes in *Dummynet* configuration for virtual machines.

### 5.3 Application Server

Our application server runs the Node.js <sup>8</sup> instance to handle the WebRTC signaling part, this machine uses Ubuntu with a domain name specified as *dialogue.io*.

This app is a common working group application that allows people to chat and video call at the same time in their own private chat rooms, we have modified it to build a specific room for our tests, this instance simply allows 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 using JavaScript APIs and uses WebSocket <sup>9</sup> protocol to handle the signaling messages from peer to peer.

The web browsers in our clients establish an HTTPS connection between the Application Server to download the web page that contain the *call.js* file that executes the WebRTC features.

### 5.4 Summary of tools

Using all the previous mentioned tools together we are able to measure how WebRTC performs in a real environment, similar to Figure 13, 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 from the captures, some of them are explained in Appendix C.

---

<sup>8</sup><http://nodejs.org/>

<sup>9</sup><http://socket.io/>

## 6 Performance evaluation of WebRTC

In this chapter we study how WebRTC performs in different network environments using topologies previously described in chapter 3. All tests are carried out using the testbed described in chapter 5 and Figure 13. We have each component running in individual virtual machines, such as the clients, the TURN server and the application server that establishes the negotiation between calls. To introduce different impairments in the bottleneck we force some of the sessions to run through the TURN server and we use *Dummynet* to emulate the variation in link capacity, latency, intermediate router queue length. We also use the *Gilbert-Elliott Model* to model packet loss [44] [45].

### 6.1 Congestion Control for WebRTC

Several congestion control mechanisms have been proposed with the goal to match the rate with available capacity while maintaining a good user experience. Instead of just relying on RTT and loss, Garudadri *et al.* [46] also use the receiver play out buffer to detect the utilization of the link. Singh *et al.* [47] use frame inter-arrival time and play out buffer size to adjust the rate to the link capacity. Zhu *et al.* [48] use ECN and loss rate to get a more accurate metric on the loss during the path and adjust the rate control based on the result. O’Hanlon *et al.* [49] propose a congestion mechanism for cross-traffic environments, using a delay-based estimation it should adapt the rate when having similar traffic on the network. In case of TCP cross-traffic, it uses a windowed-approach to adapt the rate. They switch the operational modes by using a threshold on the observed end-to-end delay. While those congestion control mechanisms have been proposed, they have not been implemented yet in any WebRTC browser.

WebRTC-enabled browsers implement a congestion control algorithm proposed by Google called *Receiver-side Real-time Congestion Control (RRTCC)* [34]. In order to properly adapt the rate to the link conditions, RRTCC needs to use two components: sender-side and receiver-side.

This protocol enables congestion control from the receiver side, in the following sub-sections we will discuss which important features RRTCC enable. The receiver estimates the usage of the link based on inter-arrival time (IAT) of the incoming packets. Three different states can appear in a link: overuse, stable and under-use. The current receiver estimate,  $A_r(i)$  is calculated using the Receiver Rate  $RR$  as follows:

$$A_r(i) = \begin{cases} 0.85 \times RR & \text{overuse} \\ A_r(i-1) & \text{stable} \\ 1.5 \times RR & \text{under-use} \end{cases} \quad (8)$$

The receiving endpoint sends an RTCP feedback message to the sender containing the Receiver Estimated Media Bitrate (REMB) [50],  $A_r$  at one second intervals.

Once the feedback message is received, the sender side uses the TCP Friendly Rate Control (TFRC) mechanism to estimate the sending rate based on the loss

rate, RTT and bytes sent [51]. This equation specifies that the rate should not vary when the loss rate is stable between 2-10%, but must be modified otherwise. If no feedback message is received, the rate is halved on the sender side. The new sender available bandwidth ( $A_s$ ) is calculated using Equation 9,  $p$  being the packet loss ratio.

$$A_s(i) = \begin{cases} A_s(i-1) \times (1 - 0.5p) & p > 0.10 \\ A_s(i-1) & 0.02 \leq p \leq 0.1 \\ 1.05 \times (A_s(i-1) + 1000) & p < 0.02 \end{cases} \quad (9)$$

Furthermore, the  $A_s(i)$  calculated should be between the rate estimated by the TFRC equation and the previous REMB value received from the receiver report. This algorithm does not react to losses under 10% and increases the rate by 5% when having losses between 2-10%. Alongside this mechanism, WebRTC also includes Forward Error Correction (FEC) mechanism, this technique generates FEC packets to protect the media stream when losses appear on the path.

## 6.2 Experimental Methodology

In the following sub-sections we evaluate different scenarios for WebRTC:

- Single RTP WebRTC flow with different networks constraints
- Single RTP WebRTC flow with TCP cross traffic
- Multiple RTP flows
- WebRTC group calls using full mesh
- Wireless scenario for WebRTC
- Interoperability tests between different web browsers
- Mobile environment performance

Each scenario is evaluated by running multiple benchmarks of tests, most of the tests run up to 15 times to derive statistical significance, lasting 300 seconds in each iteration.

## 6.3 Effects of Varying Bottleneck Link Characteristics

We have performed different tests in a bidirectional RTP scenario to study how the WebRTC applications handle different bottleneck conditions. We vary network characteristics and observe the RRTCC performance.

### 6.3.1 Variable latency

We have benchmarked tests with different one-way latencies, 50, 100, 200 and 500ms. We should be able to observe that the increase of latency reduces the average media rate during the call.

Delay modeling for real time applications is difficult, an approach can be done using the remote timestamp of the incoming packets. By using this information we can calculate if the time taken for it to arrive is greater than the predecessor frame, this is done with the IAT. However, is still very hard to model those situations in real time applications.

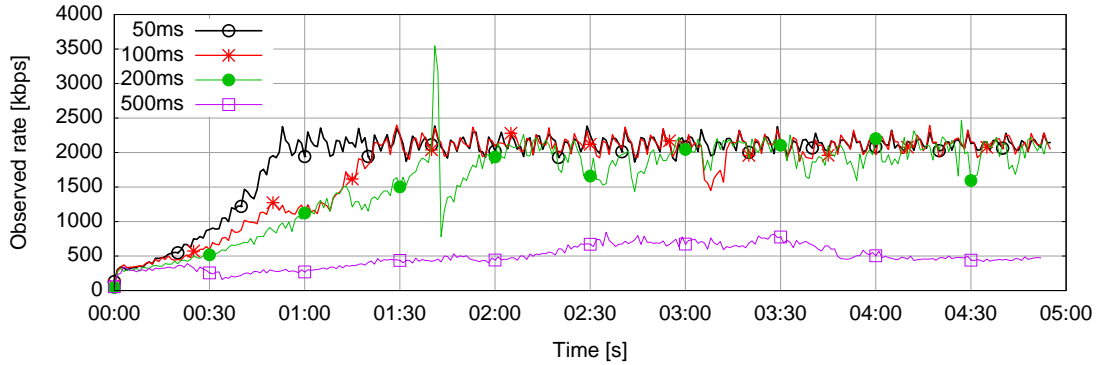


Figure 20: Media rate variation in receiver due to different bottleneck latency.

Table 2 represents the average bandwidth response to the different latency conditions, one important fact is that the loss observed in all situations is barely noticeable. The RTT is also not noticing any unexpected change. On the other hand, we can observe in Figure 20 and Table 2 that the increase in latency drastically reduces the media rate.

WebRTC uses RTP over UDP for packet transport, UDP does not carry the same congestion control mechanisms that TCP does. Even though, TCP provides higher RTT values that can delay the reaction of the RRTCC mechanism in WebRTC. Both options can be considered but modifying the encoding rate to accommodate the varying bandwidth can be difficult and slow in real time media applications.

Low latency networks will play a big role when using WebRTC in mobile devices where the ability to react to latency and packet losses will be crucial for its success against other alternatives.

### 6.3.2 Lossy environments

This section evaluates the response of WebRTC calls in lossy environments, those situations can be produced in mobile environments with poor coverage or when having packet drops in any link due to heavy 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.

	Metrics	Machine A	Machine B	Overall
50 ms	<b>Rate (Kbit/s)</b>	1909.31±258.09	1917.81±251.62	1913.56±254.86
	<b>RTT (ms)</b>	102.35±1.29	102.67±1.58	102.51±1.44
	<b>Residual Loss (%)</b>	0.02	0.09	0.05
	<b>Packet Loss (%)</b>	0.02	0.09	0.05
100 ms	<b>Rate (Kbit/s)</b>	1516.07±263.43	1453.94±272.79	1485±268.11
	<b>RTT (ms)</b>	202.82±2.94	202.32±3.05	202.57±3
	<b>Residual Loss (%)</b>	0.1	0.02	0.06
	<b>Loss (%)</b>	0.1	0.02	0.06
200 ms	<b>Rate (Kbit/s)</b>	503.71±116.45	617.92±142.69	560.82±129.57
	<b>RTT (ms)</b>	402.06±3.3	401.75±3.31	401.91±3.33
	<b>Residual Loss (%)</b>	0.3	0.35	0.33
	<b>Packet Loss (%)</b>	0.36	0.43	0.4
500 ms	<b>Rate (Kbit/s)</b>	303.58±59.22	207.77±32.48	255.67±45.85
	<b>RTT (ms)</b>	1001.3±3.8	1001.41±4.09	1001.36±3.99
	<b>Residual Loss (%)</b>	0.6	0.09	0.35
	<b>Packet Loss (%)</b>	0.63	0.1	0.37

Table 2: Summary of averaged results with different latency conditions.

We have tested a bi-directional bottleneck call with 1, 5, 10 and 20% of packet loss, according to the results in Table 3, we can see that an increase of loss rate decreases the media rate. The RRTCC Google algorithm [34] does not react to losses under 10%. Figure 21 represents the rate variation for the different packet losses, the action of the RRTCC being clear in different cases.

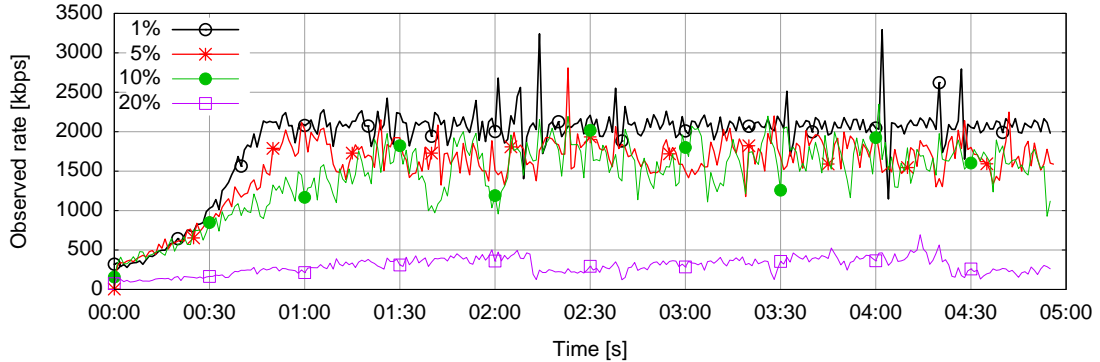


Figure 21: Media rate variation in receiver due to different bottleneck packet loss.

In Table 3, there are two different packet loss rates, the *Packet Loss* indicates the packet loss along the link, meanwhile *Residual Loss* refers to the losses after the error correction performed by RTP FEC mechanism. There are several schemes defined for use with RTP. WebRTC can negotiate the use of FEC mechanisms during the session negotiation between peers. WebRTC may use redundant transmission as a standard feature of the payload format used in a call. At the time of this thesis

there is still no recommendation on the choice of block-based FEC mechanisms for WebRTC use [39]. However, when having losses in a RTP stream, WebRTC FEC mechanism may ask for a retransmission of the lost packet on some occasions, if the delay is small it can be possible to forward them in real time, this metric is then referenced as *Residual Loss*. Those type of losses will be the ones directly affecting the data received on the peer so a low value will barely affect the quality of the call [52].

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.

	Metrics	Machine A	Machine B	Overall
1%	<b>Rate (Kbit/s)</b>	1913.59±252.11	1880.24±261.46	1986.91±256.78
	<b>OWD (ms)</b>	4.08±1.79	4.03±1.93	4.06±1.86
	<b>Residual Loss (%)</b>	0.13	0.05	0.09
	<b>Packet Loss (%)</b>	2.04	1.96	2
5%	<b>Rate (Kbit/s)</b>	1609.65±158.46	1527.74±178.52	1568.74±178.52
	<b>OWD (ms)</b>	3.72±1.82	3.26±1.76	3.49±1.79
	<b>Residual Loss (%)</b>	0.2	0.27	0.23
	<b>Loss (%)</b>	9.72	9.82	9.77
10%	<b>Rate (Kbit/s)</b>	1166.7±145.96	1114.94±177.88	1140.82±161.92
	<b>OWD (ms)</b>	3.17±3.8	3.12±2.67	3.14±3.24
	<b>Residual Loss (%)</b>	0.58	0.41	0.49
	<b>Packet Loss (%)</b>	18.98	19.05	19.02
20%	<b>Rate (Kbit/s)</b>	333.34±65.99	295.46±57.98	314.4±61.98
	<b>OWD (ms)</b>	2.65±4	2.78±4.06	2.71±4.03
	<b>Residual Loss (%)</b>	2.69	2.17	2.43
	<b>Packet Loss (%)</b>	36.08	35.95	36.01

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

Observing Table 3, 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 not being a good approach to reality in some cases and could lead to unexpected behavior [53]. However in wired environments, packet drops are usually due to queue overflows, 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.

On the other hand, the sender calculates the rate based on the receiver report that arrives from the peer. 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.



### 6.3.3 Loss and delay

Based on the two previous sections we introduce both loss and latency in the bottleneck link. We have set 10% packet loss with different delays such as 25ms, 50ms, 100ms and 200ms. Table 3 shows 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 RRTCC to handle congestion in those environments.

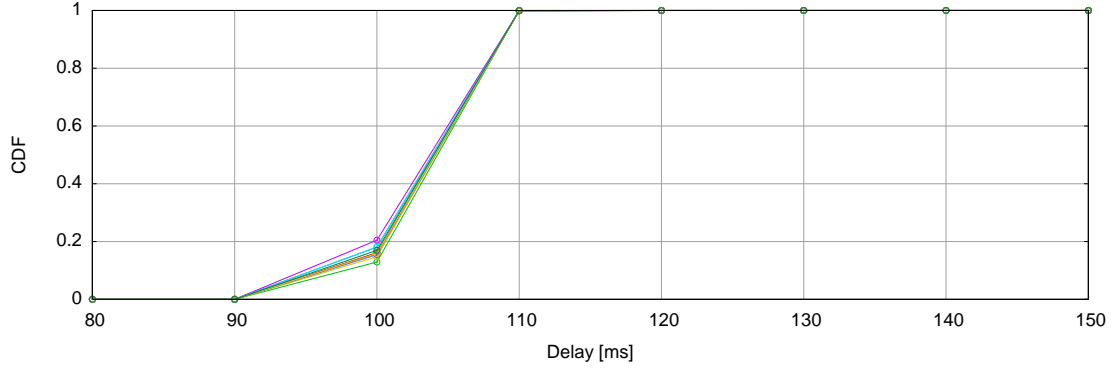
	Metrics	Machine A	Machine B	Overall
25 ms	<b>Rate (Kbit/s)</b>	72.59±18.54	70.96±18.09	71.78±18.32
	<b>RTT (ms)</b>	87.64±16.12	77.67±16.15	82.66±16.14
	<b>Residual Loss (%)</b>	6.5	7.02	6.77
	<b>Packet Loss (%)</b>	18,7	19.12	18.91
50 ms	<b>Rate (Kbit/s)</b>	50.32±15.39	60.36±18	57.19±16.63
	<b>RTT (ms)</b>	101.16±0.56	101.36±0.56	101.26±0.56
	<b>Residual Loss (%)</b>	19.29	11.32	15.49
	<b>Loss (%)</b>	26.46	19.13	22.97
100 ms	<b>Rate (Kbit/s)</b>	63.3±19.29	64.82±20.95	64.06±20.12
	<b>RTT (ms)</b>	201.59±3.03	201.36±6.09	201.48±4.56
	<b>Residual Loss (%)</b>	10.91	10.82	10.87
	<b>Packet Loss (%)</b>	18.81	18.67	18.77
200 ms	<b>Rate (Kbit/s)</b>	66.89±20.12	65.66±19.63	66.27±19.87
	<b>RTT (ms)</b>	403.73±25.67	403.93±31.71	403.83±28.69
	<b>Residual Loss (%)</b>	11.35	12.01	11.68
	<b>Packet Loss (%)</b>	18.59	19.07	18.83

Table 4: Averaged results with different delay conditions and 10% packet loss.

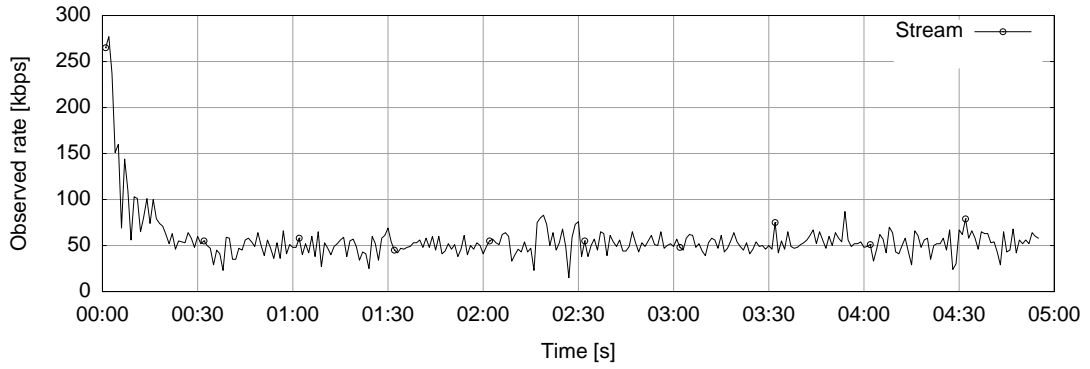
Compared with Table 3 and Table 2 , we can observe that RRTCC cannot handle the retransmissions of packets using FEC. The residual loss increases in all latencies from 0.49% to 6.77% in the best latency situation while the overall packet loss remains similar  $\approx 19\%$ . Figure 22b shows how the sending rate reduces when facing loss and delay on the same path RRTCC being unable to fix this issue with its internal mechanisms. On the other side, Figure 22a indicates that while the rate is reduced the RTT keeps constant during all the duration of the call at 100ms.

If we study the way WebRTC calculates the rate in this situation we can see that the sender decision will be based on the RTT, packet loss and available bandwidth that is estimated from the receiving side using Equation 9 [34]. Obviously the real output differs from the expected by using the formula, the reason is that even the congestion mechanism on WebRTC calculates the rate using Equation 9, the sender rate is always limited by the TCP Friendly Rate Control (TFRC) formula that is calculated using delay and packet loss ratio together [51].

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



(a) CDF of delay distribution for 100ms latency.



(b) Remote stream bandwidth for 10% packet loss rate and 50ms delay.

Figure 22: Bandwidth and mean for 1 Mbit/s with multiple queue sizes

### 6.3.4 Bandwidth and queue variations

In this section we vary the queue size at an intermediate router in order to observe the impact of the performance in the RRTCC algorithm. For this test we have selected to run 500 Kbit/s, 1 and 5 Mbit/s throughput rate with different queue lengths from 100 ms, 500 ms, 1s and 10 s. In total we have run 12 different tests with ten iterations each.

We calculate the queue size in function of time, this being the amount of time the packet will remain in the router before being discarded. However, the result is given in number of packets using Equation 10. The MTU size in this test is set to 1500 bytes.

$$Queue(packets) = \frac{Bandwidth(Bits)}{8 \times MTU} \times Queue(seconds) \quad (10)$$

For example, when setting a throughput of 1Mbps and 1s queue depth, a router would be able to handle 83 packets as a total queue length. A queue length of 100ms represents a short queue meanwhile a 10s queue identifies a buffer-bloated queue. Tables 5, 6 and 7 show the result for the rates and queue sizes and bandwidth used for the tests.

After running the tests we can state that the overall Average Bandwidth Utilization (ABU) for scenarios with 500 Kbit/s, 1 and 5 Mbit/s is  $\approx 0.4$  (40%) when running only RRTCC traffic. Furthermore we can assure that varying the queue lengths does not directly affect the performance of RRTCC.

500 Kbit/s				
Queue	Metrics	Machine A	Machine B	Overall
100 ms	<b>Rate (Kbit/s)</b>	243.17 $\pm$ 35.59	154.85 $\pm$ 30.41	199.01 $\pm$ 33
	<b>OWD (ms)</b>	39.19 $\pm$ 11.85	39.69 $\pm$ 17.34	39.44 $\pm$ 14.59
	<b>Residual Loss (%)</b>	1.94	1.87	1.91
	<b>Packet Loss (%)</b>	3.4	4.57	3.99
500 ms	<b>Rate (Kbit/s)</b>	203.71 $\pm$ 26.81	175.1 $\pm$ 20.84	189.41 $\pm$ 23.83
	<b>OWD (ms)</b>	35.92 $\pm$ 30.57	35.43 $\pm$ 30.34	35.67 $\pm$ 30.34
	<b>Residual Loss (%)</b>	0.34	0.24	0.29
	<b>Loss (%)</b>	0.78	0.74	0.76
1 s	<b>Rate (Kbit/s)</b>	214.81 $\pm$ 16.69	211.59 $\pm$ 15.74	213.2 $\pm$ 16.22
	<b>OWD (ms)</b>	47.43 $\pm$ 61.2	48.18 $\pm$ 62.23	47.8 $\pm$ 62.23
	<b>Residual Loss (%)</b>	0.33	0.43	0.38
	<b>Packet Loss (%)</b>	0.59	0.63	0.61
10 s	<b>Rate (Kbit/s)</b>	210.54 $\pm$ 14.92	218.56 $\pm$ 13.84	214.55 $\pm$ 14.38
	<b>OWD (ms)</b>	86.42 $\pm$ 148.97	89.34 $\pm$ 150.5	87.88 $\pm$ 149.73
	<b>Residual Loss (%)</b>	0.07	0.06	0.06
	<b>Packet Loss (%)</b>	0.06	0.05	0.06

Table 5: Averaged results with different queue configurations and 500 Kbit/s bandwidth constraint.

However, we can study the result of the test performed with 1 Mbit/s throughput bottleneck limitation, as the maximum standard bandwidth for WebRTC is approximately 2 Mbit/s, setting 1 Mbit/s throughput limitation forces RRTCC to adapt the outgoing rate to the link conditions.

Figure 25 represents the bandwidth and mean plotted for all the different latencies with 1 Mbit/s limitation. We can see that the average rate response varies in small amount of bandwidth but with large deviation in each iteration, when having 500ms and 1s queue size (25c) we have larger deviation in means of packets being buffered in the relay. Otherwise, when the queue size reduces to 100ms (25d) the deviation gets smaller but delay response is worse.

We can compare Figure 26 delay distribution results for the best case (26a) and worse case (26d). The delay response with large queue is better. Furthermore, for the 100ms test, the delay is not constant during the test with a maximum delay of 100ms.

Delay experience with small queue sizes will be worst for the user experience, we might experience unexpected delays that RRTCC won't be able to handle, when having larger queue sizes we won't notice the delay variations as much as with the previous example. Having a curvy increase in delay distribution figure will result

1 Mbit/s				
Queue	Metrics	Machine A	Machine B	Overall
100 ms	Rate (Kbit/s)	436.72±72.7	289.02±63.66	362.87±68.18
	OWD (ms)	30.91±11.75	31.94±12.05	31.42±11.9
	Residual Loss (%)	1.22	0.37	0.79
	Packet Loss (%)	1.77	1.07	1.42
500 ms	Rate (Kbit/s)	323.59±82.33	285.21±73.71	304.4±78.02
	OWD (ms)	24.72±10.69	28.55±11.04	26.63±10.86
	Residual Loss (%)	0.16	0.08	0.12
	Loss (%)	0.16	0.08	0.12
1 s	Rate (Kbit/s)	401.41±58.14	347.86±59.15	374.64±58.64
	OWD (ms)	28.94±8.7	26.02±8.9	27.48±8.8
	Residual Loss (%)	0.06	0.04	0.05
	Packet Loss (%)	0.06	0.04	0.05
10 s	Rate (Kbit/s)	481.08±31.01	395.55±32.15	438.31±31.58
	OWD (ms)	26.83±7.08	27.5±7.56	27.16±7.32
	Residual Loss (%)	0.04	0.05	0.04
	Packet Loss (%)	0.04	0.05	0.04

Table 6: Averaged results with different queue configurations and 1 Mbit/s bandwidth constraint.

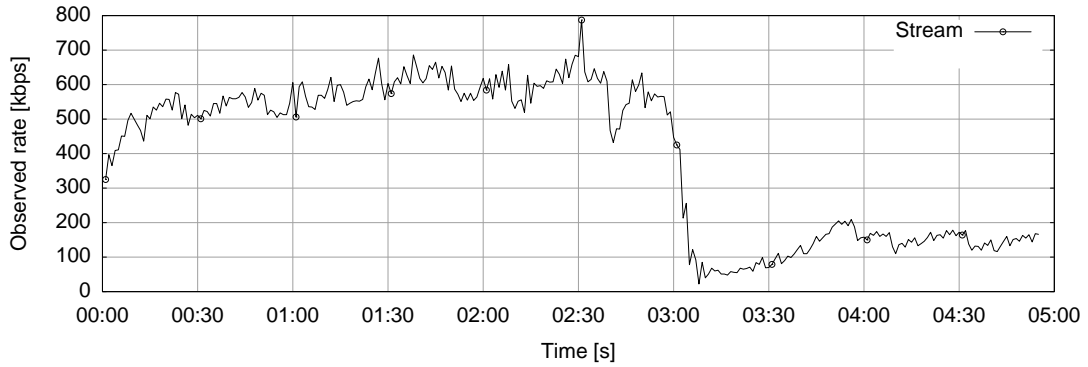


Figure 23: Remote stream bandwidth for 1 Mbit/s and 500ms queue size.

in sudden delay variations in the call. The conclusion is that RRTCC is able to adapt to low rate 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.

RRTCC stabilizes the rate until the amount of delay triggers the algorithm to fit the new queue state. Figure 23 and 24 show the rate 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 producing low delay.

Studying the way the sender RRTCC takes decisions about rate adaptation we

5 Mbit/s				
Queue	Metrics	Machine A	Machine B	Overall
100 ms	<b>Rate (Kbit/s)</b>	1963.15±227.1	1979±211.46	1965.95±224.02
	<b>OWD (ms)</b>	20.38±4.13	19.89±4.19	20.13±4.16
	<b>Residual Loss (%)</b>	0.02	0.01	0.02
	<b>Packet Loss (%)</b>	0.02	0.01	0.02
500 ms	<b>Rate (Kbit/s)</b>	1595.41±266.41	1565.36±264.06	1580.39±265.24
	<b>OWD (ms)</b>	18.15±14.06	17.83±12.99	17.99±15.52
	<b>Residual Loss (%)</b>	0.03	0.02	0.03
	<b>Loss (%)</b>	0.03	0.02	0.03
1 s	<b>Rate (Kbit/s)</b>	1919.25±231.35	1922.09±237.41	1920.67±243.38
	<b>OWD (ms)</b>	19.64±5.38	17.25±5.76	18.45±5.57
	<b>Residual Loss (%)</b>	0.02	0.01	0.02
	<b>Packet Loss (%)</b>	0.02	0.01	0.02
10 s	<b>Rate (Kbit/s)</b>	1595.41±266.41	1565.36±264.06	1580.39±264.24
	<b>OWD (ms)</b>	18.15±14.06	17.83±12.99	17.99±13.52
	<b>Residual Loss (%)</b>	0.03	0.02	0.03
	<b>Packet Loss (%)</b>	0.03	0.02	0.03

Table 7: Averaged results with different queue configurations and 5 Mbit/s bandwidth constraint.

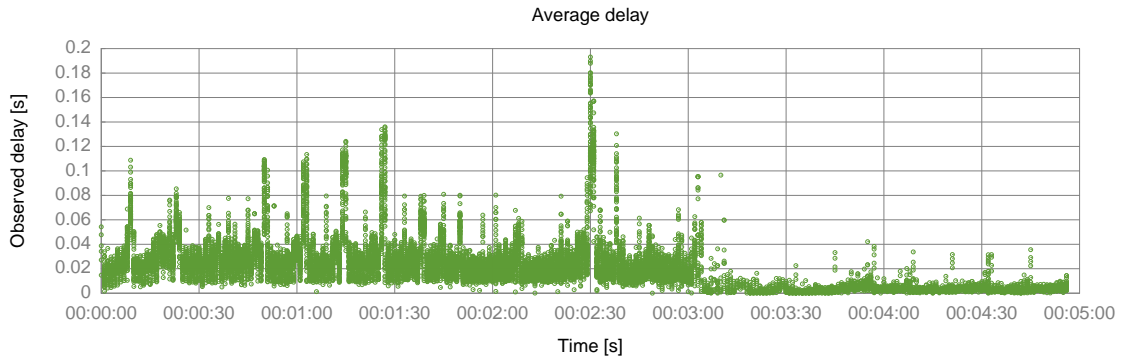
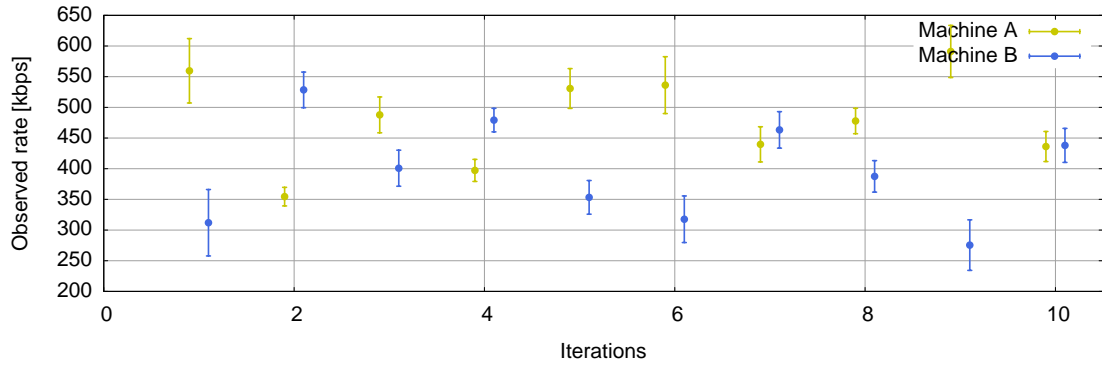
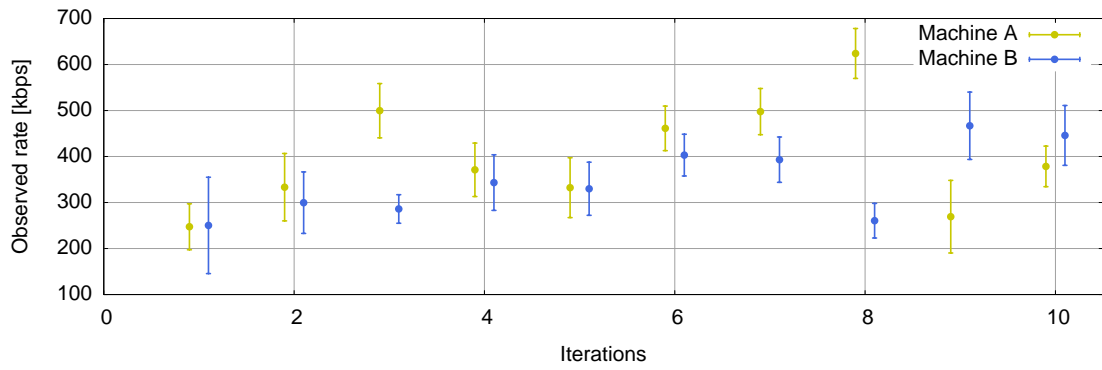


Figure 24: Stream delay for 1 Mbit/s and 500ms queue size.

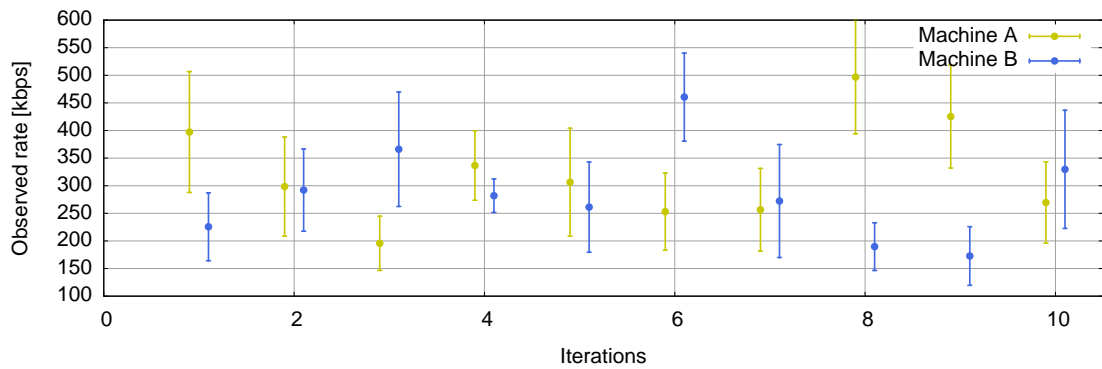
can observe that the available rate estimations calculated by the receiver are only reliable when the size of the queues along the channel is large enough [34]. When having short queues along the path, the maximum rate cannot be estimated without having packet loss on the link, as in this case the packet loss is negligible, the call is not able to use the maximum amount of bandwidth available on the path.



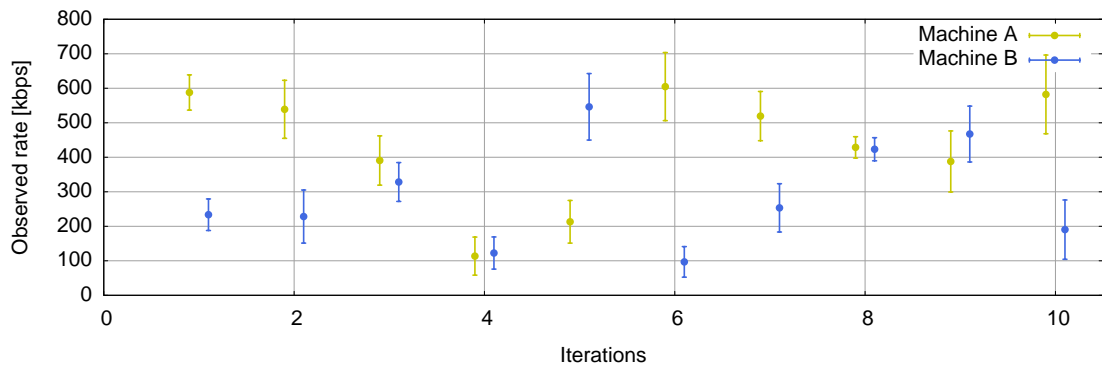
(a) 1 Mbit/s and 10s queue size.



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

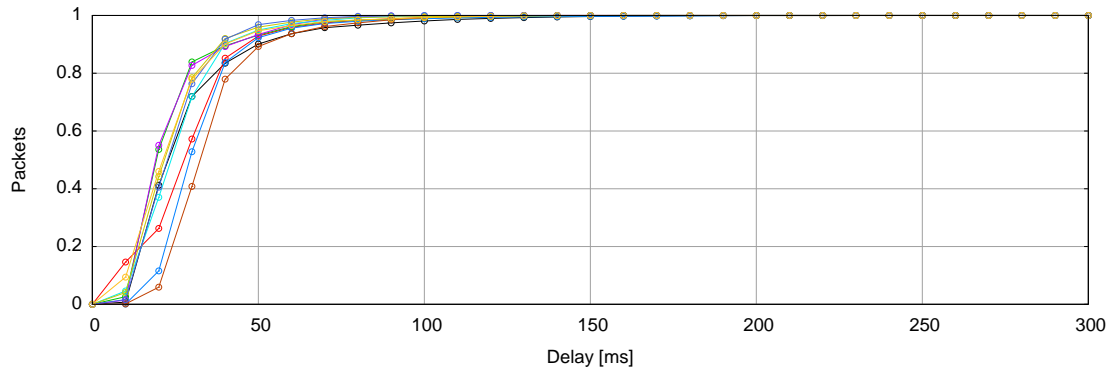


(c) 1 Mbit/s and 500ms queue size.

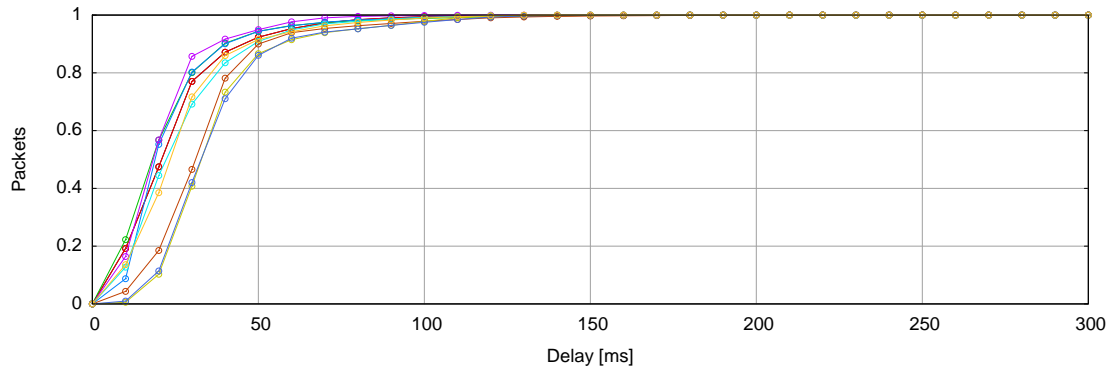


(d) 1 Mbit/s and 100ms queue size.

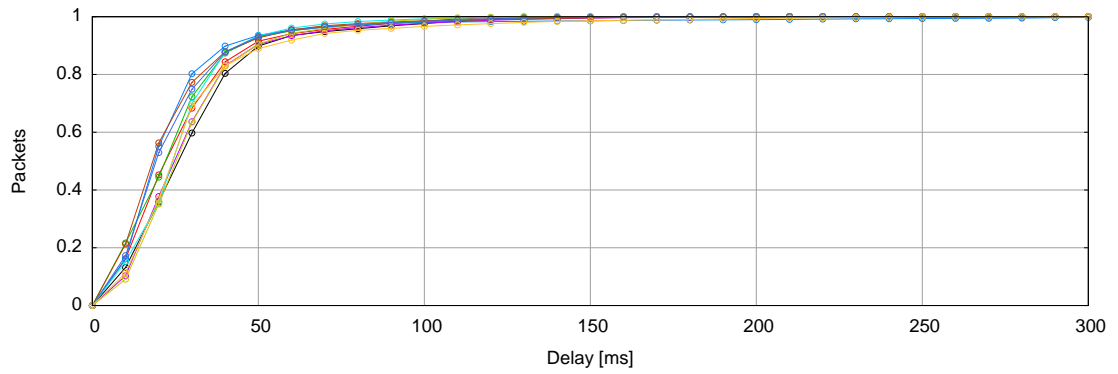
Figure 25: Bandwidth and mean for 1 Mbit/s with multiple queue sizes



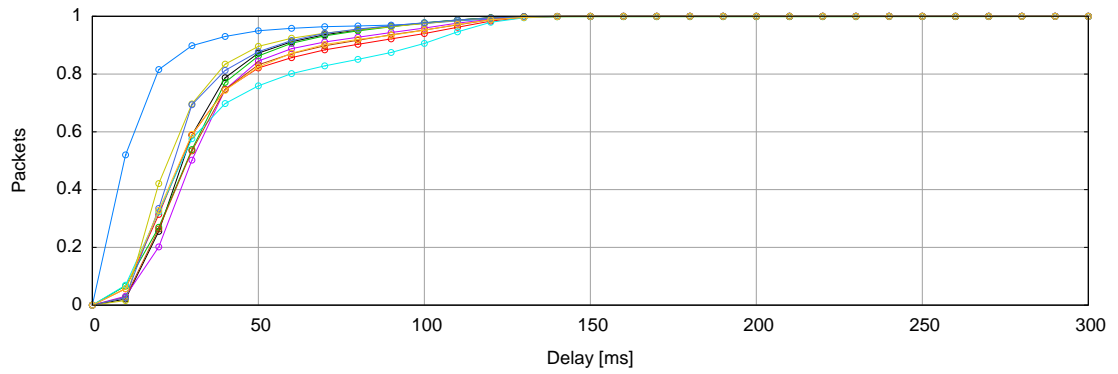
(a) 1 Mbit/s and 10s queue size.



(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 26: CDF of delay distribution for 1 Mbit/s with multiple queue sizes

### 6.3.5 Unconstrained link

In this section we are going to proceed with a sample test over an unconstrained link to check the performance of WebRTC.

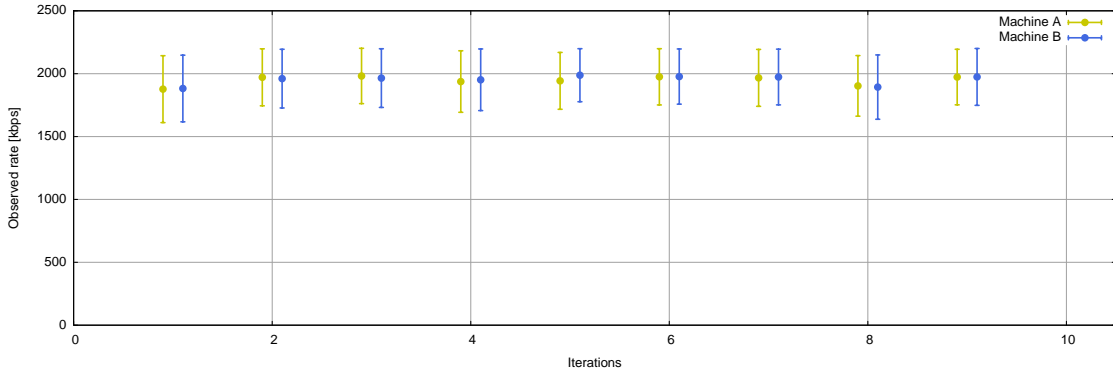


Figure 27: Rate average and deviation for unconstrained link in each iteration.

Figure 27 presents the average rate of every iteration of the test in a wired network without any link constraint, the average rate obtained in the test is  $1949.7 \pm 233$  Kbit/s, we can conclude that a standard rate for a video call in a unconstrained link using WebRTC is approximately 2 Mbit/s. In Figure 27 there are only nine iterations as one call failed to establish. Furthermore, obtained delay is  $5.1 \pm 1.5$  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 8, besides the network performance we also track the call failure rate, considering all those calls go through the bottleneck 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 from 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.

	Machine A	Machine B	Overall
<b>Rate (Kbit/s)</b>	1947.61±232.75	1951.76±234.5	1949.7±233.62
<b>RTT (ms)</b>	9.49±2.11	9.64±2.71	9.57±2.41
<b>OWD (ms)</b>	4.84±1.5	5.4±1.53	5.12±1.52
<b>Residual Loss (%)</b>	0.012	0.01	0.011
<b>Packet Loss (%)</b>	0.012	0.01	0.011
<b>Setup time (ms)</b>	1436.33±25	1447.44±22.71	1441.88±24.04

Table 8: P2P metrics output for WebRTC call with no link restriction.

Delay values in Table 8 are averaged using all the *ConMon* data obtained for



each stream in all iterations, thus if it is an approximate delay value it might not be representative of the exact delay occurred during the call. Observing the example in Figure 24 we can see that the delay is variable, this means that the averaged OWD may not be the most appropriate metric to measure the time delay user experience. 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.

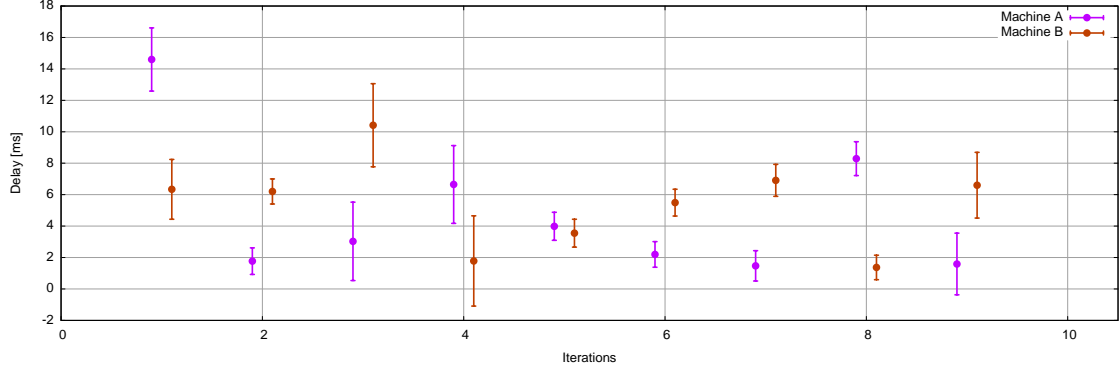


Figure 28: Mean and deviation for OWD in each iteration with no link constraints.

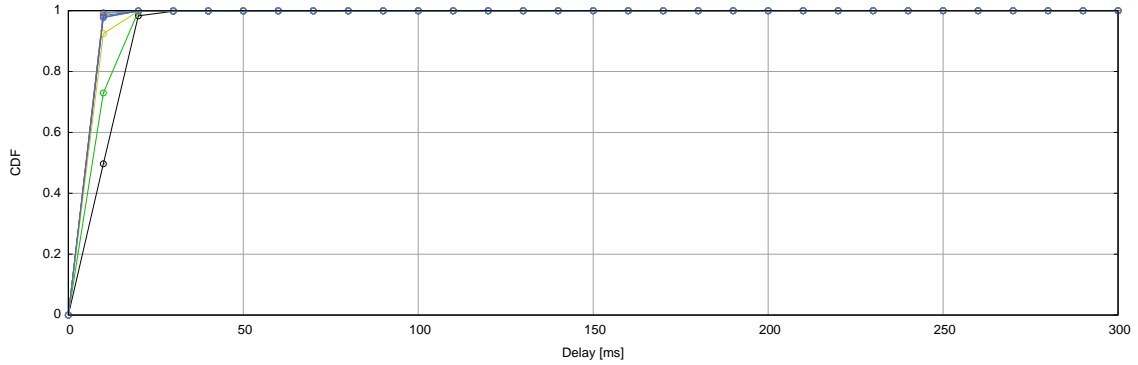


Figure 29: Delay distribution for each iteration with no link constraints.

Figure 28 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*. Machine A and B identify the capture in each receiver of the call. We run a Network Time Protocol daemon (NTPd) daemon to adjust the drift on the clock and sync both machines. Figure 28 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 29, CDF delay 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 are transferred

in less than 25ms delay in all the iterations for the unconstrained test. The user experience with this small amount of delay that do not vary on time is almost negligible. Figures 28 and 29 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 condition of the link, we use the delay distribution and OWD mean to evaluate the delay response.

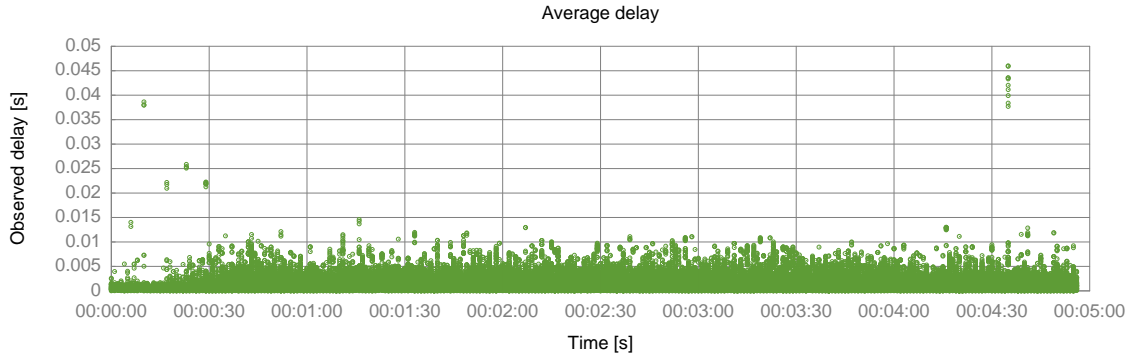


Figure 30: OWD response during the call for one video stream for an unconstrained link call.

Figure 30 represents the OWD response along the call of one specific video stream, it is easy to see that there are some small variations that may affect the call but the overall delay value is stable.

## 6.4 Effects of TCP Cross Traffic

In this scenario, we are measuring the performance of WebRTC RTP flows when competing with TCP cross-traffic on the bottleneck with different path constraints. We use *Iperf* on the TURN server to emulate the TCP cross-traffic. *Iperf* clients will run on the peers that are performing the call. We use long TCP flows that represent large files being downloaded by the peers. Those flows run in parallel with the RTP media sessions established in WebRTC between the same clients. The scenario used is the one shown in Figure 31 with the clients running *Dummynet* instead of the relay to simulate equal path conditions for both TCP and WebRTC.

First we run the server as daemon on the TURN by executing:

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

In the next step, *Iperf* sends TCP packets, to do so we will run:

```
# iperf -c XXXX -t 300
```

In the above command, *-t* is the amount of time the test length and *-c* is the feature that sets the process to operate as a client to the recipient address.

The first test is configured with a RRTCC flow competing with two TCP flows on the bottleneck. The capacity on the bottleneck is set to 1 Mbit/s upstream and 10

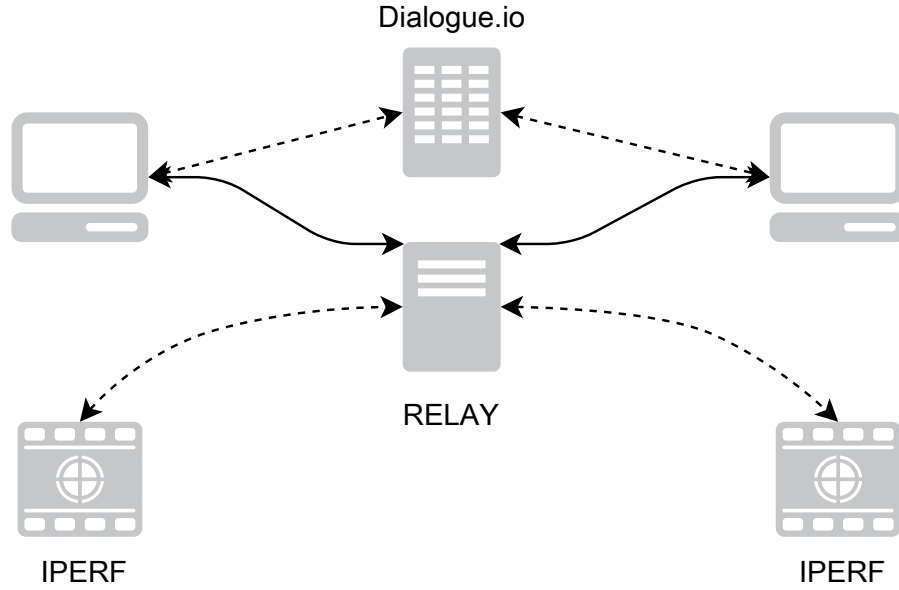


Figure 31: Topology for traffic flooded path using *Iperf*.

Mbit/s downstream. Considering the amount of data required by a RRTCC media stream, the 10 Mbit/s should be shared by TCP and the remote stream. However, the 1 Mbit/s upstream could be used only with the local RTP media.

Table 9 shows that, during the first test, RRTCC starves to provide enough rate to the outgoing RTP flow and cannot compete with the TCP cross-traffic on the path, only a fraction of the bandwidth is given to the outgoing local stream and the receiver might not get proper quality on the playback.

During the second test we set the bottleneck bandwidth to 10 Mbit/s upstream and downstream. The result, seen in Table 9, shows that RRTCC still cannot provide maximum rate to the outgoing media. We can conclude that congestion mechanisms in WebRTC cannot provide maximum bandwidth when competing with TCP traffic in a constrained environment.

To analyze how RRTCC performs during a call we can observe Figure 32 and 33, both graphs represent the same stream in the duration of a call. During the whole period, TCP keeps on increasing its congestion window and, at the same time, filling the queues of the middle routers. This action produces delay on the path triggering a reaction in the RRTCC algorithm that reduces the rate on the ongoing stream. This is the reason why the ramp in Figure 32 increases slowly until timeouts appear, at that time RRTCC reduces the rate by half trying to adapt to the new conditions. We can also observe that the rate increase and delay decrease is related in Figures 32 and 33.

The delay decrease seen in Figure 33 could be given by the routers buffers that suffer from bufferbloat adding high delay in the path, this causes the TCP to go into slow-start triggering RRTCC mechanisms to increase the rate value. In this scenario buffers filled with TCP packets increase the latency on the path. This leads to the poor performance observed in Figure 33.

10/1 Mbps				
Delay	Metrics	Machine A	Machine B	Overall
0 ms	<b>Rate (Kbit/s)</b>	76.16±25.12	74.81±24.93	75.49±25.02
	<b>OWD (ms)</b>	4.71±12.71	5.12±14.9	4.92±13.81
	<b>Residual Loss (%)</b>	0.23	0.44	0.26
	<b>Packet Loss (%)</b>	0.23	0.21	0.22
25 ms	<b>Rate (Kbit/s)</b>	79.97±25.7	88.07±32.65	84.02±29.17
	<b>OWD (ms)</b>	29.56±3.03	27.52±2.92	28.54±2.97
	<b>Residual Loss (%)</b>	0.13	0.38	0.24
	<b>Loss (%)</b>	0.13	0.25	0.19
100 s	<b>Rate (Kbit/s)</b>	78.2±27.62	82.2±33.57	80.2±30.6
	<b>OWD (ms)</b>	108.29±4.19	108.08±4.04	108.19±4.11
	<b>Residual Loss (%)</b>	0.24	0.38	0.19
	<b>Packet Loss (%)</b>	0.24	0.14	0.19
10/10 Mbps				
Delay	Metrics	Machine A	Machine B	Overall
50 ms	<b>Rate (Kbit/s)</b>	76.16±25.12	74.81±24.93	75.49±25.02
	<b>OWD (ms)</b>	4.71±12.71	5.12±14.9	4.92±13.81
	<b>Residual Loss (%)</b>	0.23	0.44	0.26
	<b>Packet Loss (%)</b>	0.23	0.21	0.22

Table 9: Metrics for a bottleneck with varying amount of TCP cross-traffic and link constraints.

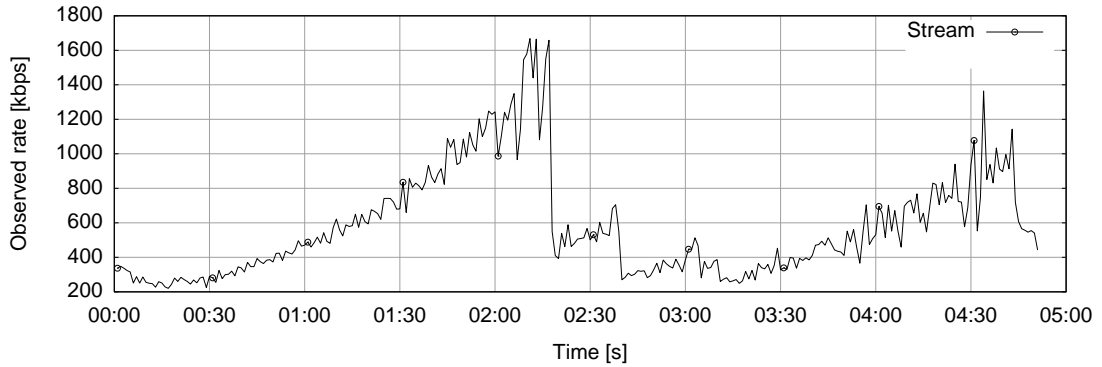


Figure 32: Instantaneous receiver rate during one call, bottleneck capacity set to 10/10 Mbit/s.

We can observe some interesting behavior in all three iterations when looking at Figure 34 CDF delay distribution, the response varies from all three tests all of them being bad, a lot of unexpected delays occur during the call. The delay deviation is small but the tolerance for TCP flooded networks is low in WebRTC.

Lastly, Figure 35 shows the rate output for the 10/1 Mbit/s scenario. We can observe that RRTCC lowers the rate to adapt to the path conditions, due to the

TCP cross-traffic, the throughput lowers to about 400 Kbps which is far from the maximum possible media stream rate.

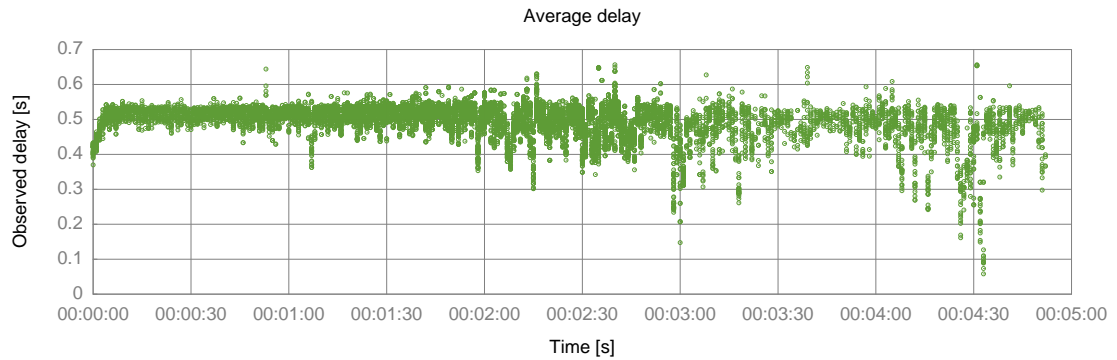


Figure 33: Instantaneous delay during one call, bottleneck set to 10/10 Mbit/s.

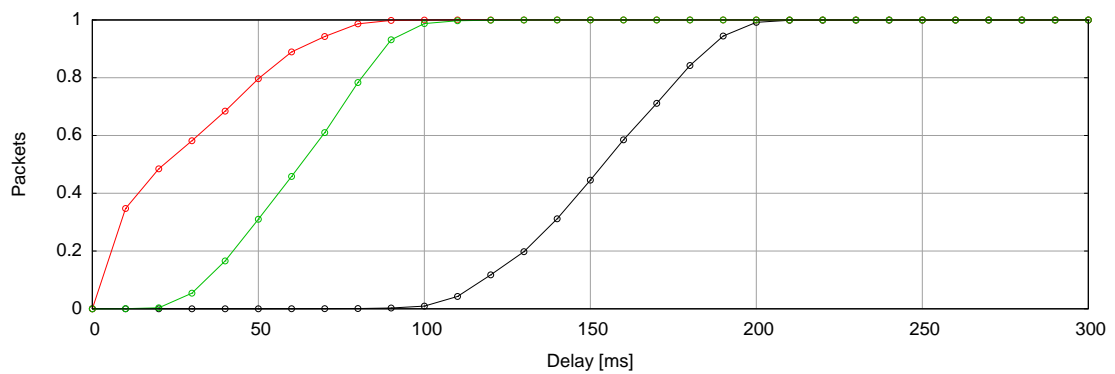


Figure 34: Total CDF delay distribution for TCP cross-traffic with 10/10 Mbps link.

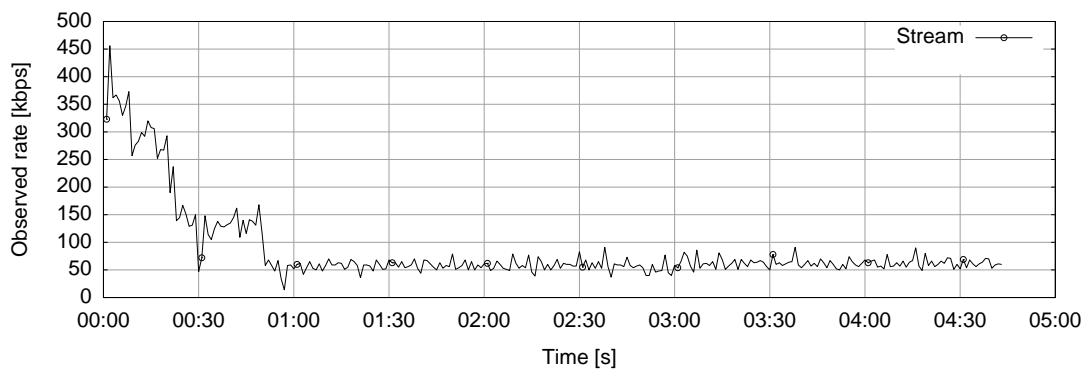


Figure 35: Instantaneous receiver rate during a call with TCP cross-traffic, bottleneck set to 10/1 Mbit/s.

## 6.5 Effects of RTP Cross Traffic

Similar to the previous section, in this chapter we are testing the performance of RRTCC when having multiple RTP flows on the same path through the bottleneck. Figure 36 represents the topology used for the test.

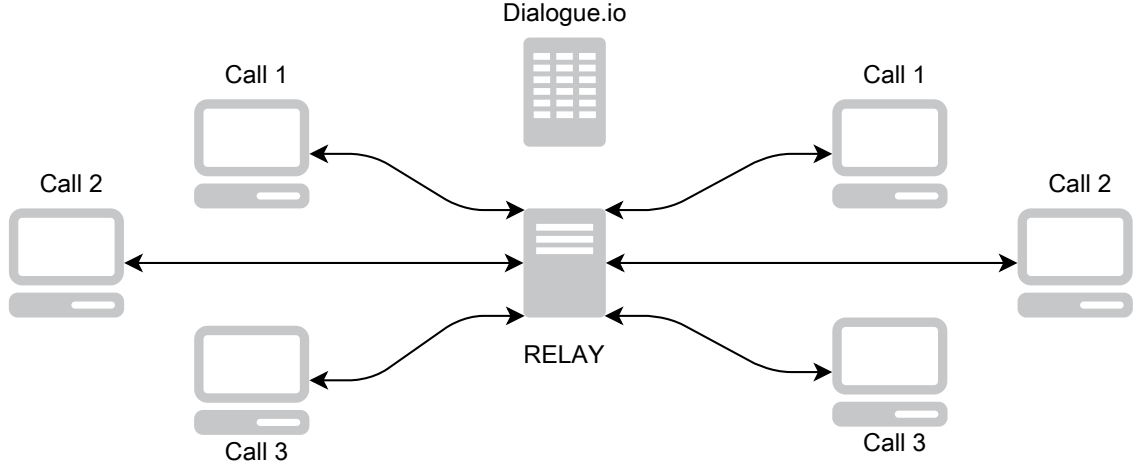


Figure 36: Topology for three different parallel RTP calls over the same link.

Three different environments are tested, firstly two WebRTC calls share a common bottleneck. For this test we have two RTP streams running through the TURN server with different *Dummynet* constraints. We limit the bottleneck throughput with none, 10 or 20 Mbps. Table 10 shows the result for the test.

10/1 Mbps				
	Metrics	Machine A	Machine B	Overall
-	Rate (Kbit/s)	392.08±182.9	545.94±259.27	469.01±221.09
	OWD (ms)	5.1±4.8	5.9±5.69	5.5±5.26
	Residual Loss (%)	0.03	0.07	0.04
	Packet Loss (%)	0.03	0.04	0.03
10 Mbit/s	Rate (Kbit/s)	178.65±60.05	141.83±42.02	160.24±51.04
	OWD (ms)	8.34±9.78	8.18±9.67	8.26±9.72
	Residual Loss (%)	0.04	0.13	0.07
	Loss (%)	0.04	0.08	0.06
20 Mbit/s	Rate (Kbit/s)	432.56±141.31	531.13±169.82	481.85±155.56
	OWD (ms)	19.27±11.64	20.76±12.68	20.02±12.16
	Residual Loss (%)	0.65	1.06	0.85
	Packet Loss (%)	0.89	0.57	0.61

Table 10: Two parallel simultaneous calls with different bandwidth constraints on the link.

Observing Table 10 we can state that the calls are unable to occupy any sub-

stantial share of the bottleneck in any of the situations, the difference being between 20 and 10 Mbps almost negligible. They both struggle to obtain the necessary rate of 2 Mbps standard RRTCC maximum target.

Furthermore, the following test arrange three different calls that share the same bottleneck without any *Dummynet* constraint. The results seen in Table 11 shows that the average rate is higher but still not able to reach the maximum level of 2 Mbps.

	Machine A	Machine B	Overall
<b>Rate (Kbit/s)</b>	768.04±180.93	850.1±223.84	809.07±202.38
<b>OWD (ms)</b>	31.36±24.37	31.6±25.49	31.48±24.93
<b>Residual Loss (%)</b>	0.15	0.43	0.23
<b>Packet Loss (%)</b>	0.15	0.27	0.21

Table 11: Three simultaneous parallel calls on the same path without any link constraints.

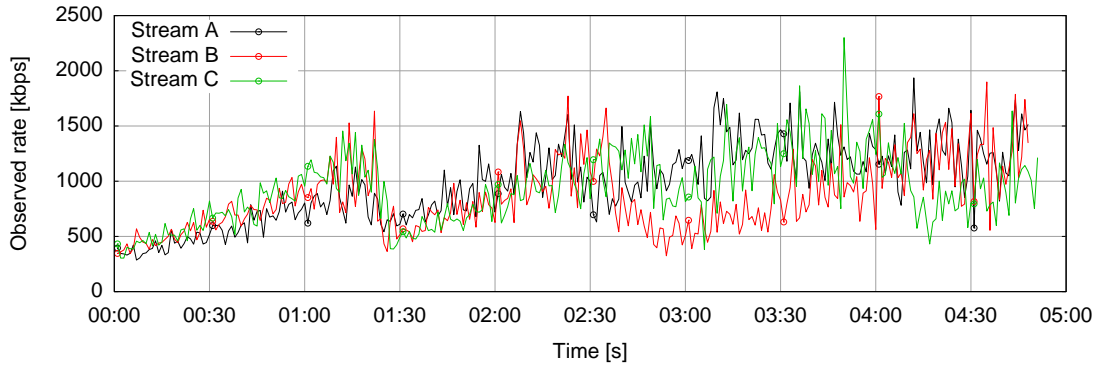


Figure 37: Variation in receiver rate for three parallel calls starting together.

Figure 37 shows three different RTP streams ramping up at a similar rate, reaching the similar peak and drop their rate together. All three calls start with different endpoints and using separate browsers, even though rates synchronize on the bottleneck as the RRTCC is acting in the same manner.

Figure 38 represents the delay on the same streams during the call, we can observe some peaks of delay during the same period as the rate decreases, the result of this is the sudden drop of rate.

In general the delay response in all the streams is bad, Figure 39 plots the delay distribution of the three simultaneous calls, the delay distribution produces variable unexpected delays, probably the user experience is not going to be optimal.

The last test is done by running the three calls starting at 30s intervals, by this we try to generate a more realistic approach to what could happen in a network. All three calls share the bottleneck link. Table 12 shows that the averaged rate for this test is slightly higher than the previous case. While the rate has improved compared

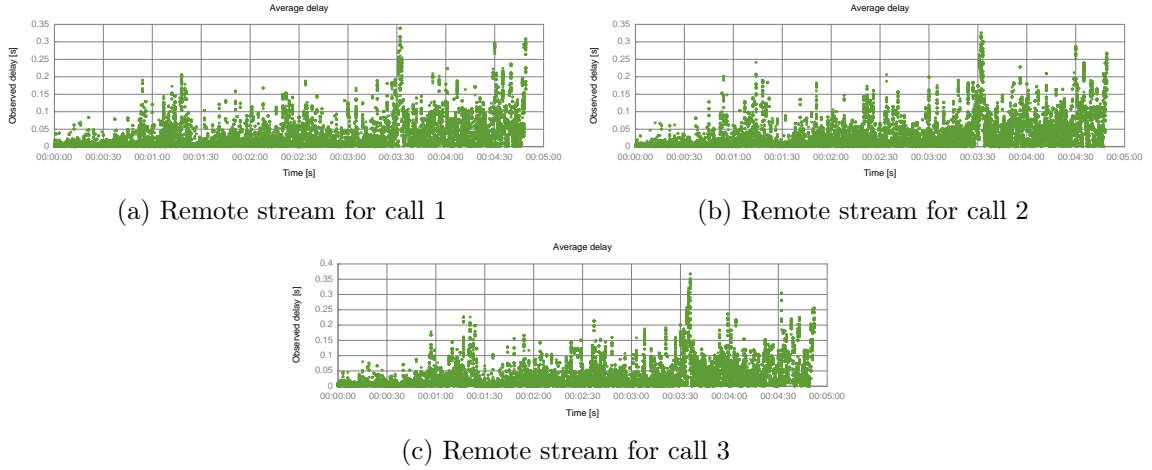


Figure 38: Delay representation for all remote streams in a three peer parallel call.

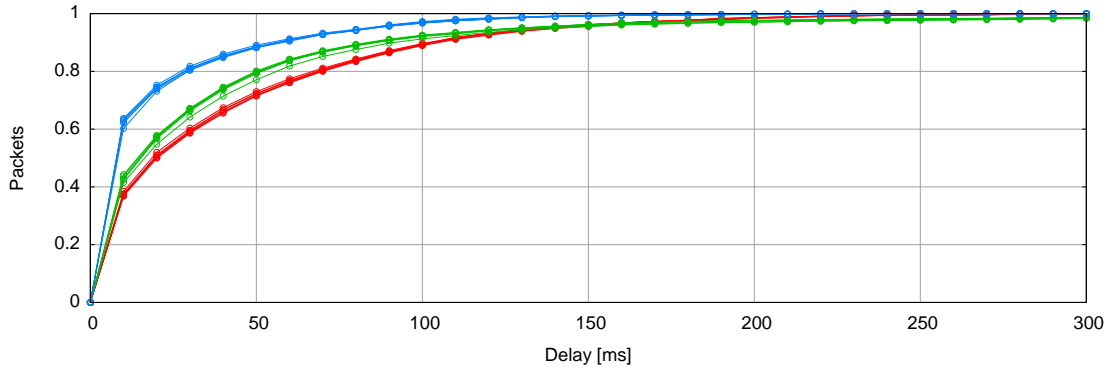


Figure 39: Total delay distribution for three parallel calls.

to the first test, Figure 40 shows that the first call has a disadvantage and, in all the cases, temporarily starves when new flows appear on the bottleneck, ramping up after a period of time.

	Machine A	Machine B	Overall
<b>Rate (Kbit/s)</b>	1214.66±247.41	1093.98±253.68	1154.32±250.54
<b>OWD (ms)</b>	34.86±27.09	35.44±28.68	35.15±27.88
<b>Residual Loss (%)</b>	0.08	0.17	0.91
<b>Packet Loss (%)</b>	0.08	0.09	0.08

Table 12: Three time shifted (30s) parallel calls on the same path without any link constraints.

Figure 40 shows that the first call temporarily increases the rate meanwhile there are no more flows on the path. When the second stream appears, the existing session has to compete with the new one that observes some queues from the existing media on the path. The reaction from the first flow is to reduce the sending rate when it



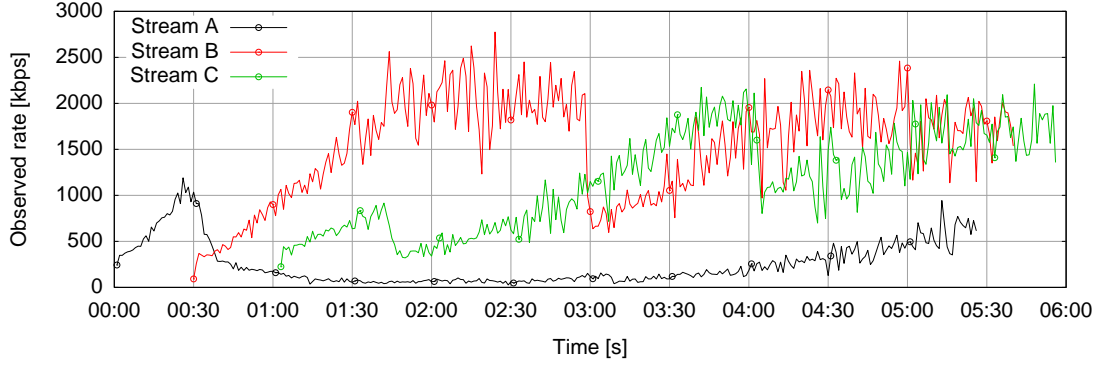


Figure 40: Variation in receiver rate for three parallel calls with 30s interval start time.

observes an increase at the queues in order to avoid congestion.

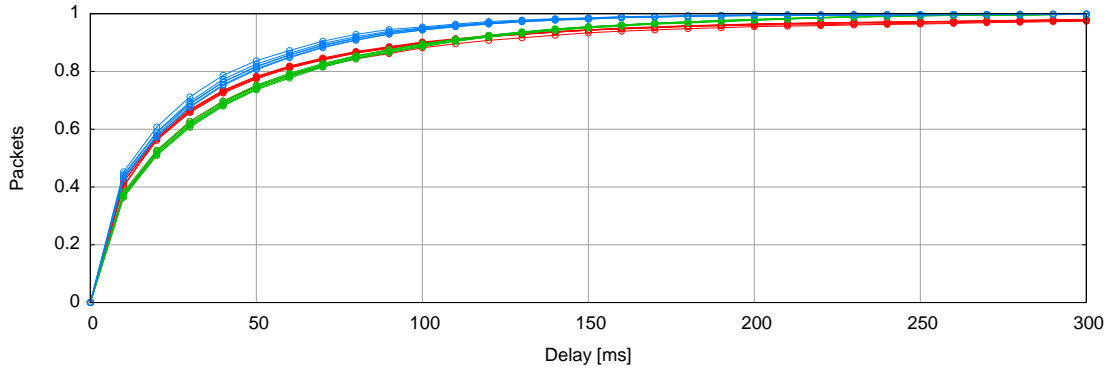


Figure 41: Total delay distribution for three asynchronous parallel calls.

Figure 41 represents the CDF delay distribution for the time shifted test, similar to the previous example (Figure 39) but with a worst delay response. Delay will affect the user experience in the same way with more random variable delays.

## 6.6 Multiparty Calls: Full-Mesh

A common setup in real-time communications is video conferencing. We try to determine if the RRTCC algorithm is able to cope with the requirements of a multiparty call. For this environment we setup a group call between three participants in a full-mesh topology. With this, we have each participant sending the media to the other two participants and also receiving the individual media from them. This scenario determines whether the performance of the endpoints is heavily affected by the multiple calls and if the bottleneck constraints interfere with the call quality.

Figure 42 shows a simple three-peer mesh topology like the one we are using in this test. In this scenario we do not use any TURN server or path constraint. The common bottleneck in an environment like this is the last and first hop in the path, this device has to handle lots of traffic and its queues might be heavily loaded.

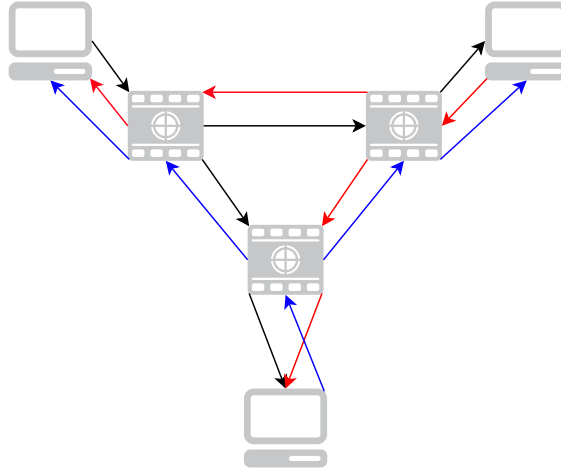


Figure 42: Mesh topology for WebRTC.

Table 13 shows the result for this setup. Observing the average rate seems obvious that the WebRTC stack cannot cope to deliver maximum rate to any channel even without having path constraints. This might be given due to performance issues on the device itself (see Section 6.11). When handling more than three peers the call cannot be established in most of the cases with the actual device resources. However, for a three peer mesh call time response is good for real-time communication.

No MCU				
	Machine A	Machine B	Machine C	Overall
<b>Rate (Kbit/s)</b>	333.38±115.13	344.48±95.43	410.77±115.97	362.88±108.84
<b>OWD (ms)</b>	6.1±5.09	5.79±5.09	5.82±5.15	5.91±5.11
<b>Residual Loss (%)</b>	0.01	0.01	0.00	0.00
<b>Packet Loss (%)</b>	0.01	0.01	0.00	0.01

Table 13: Three-peer mesh call with and without TURN.

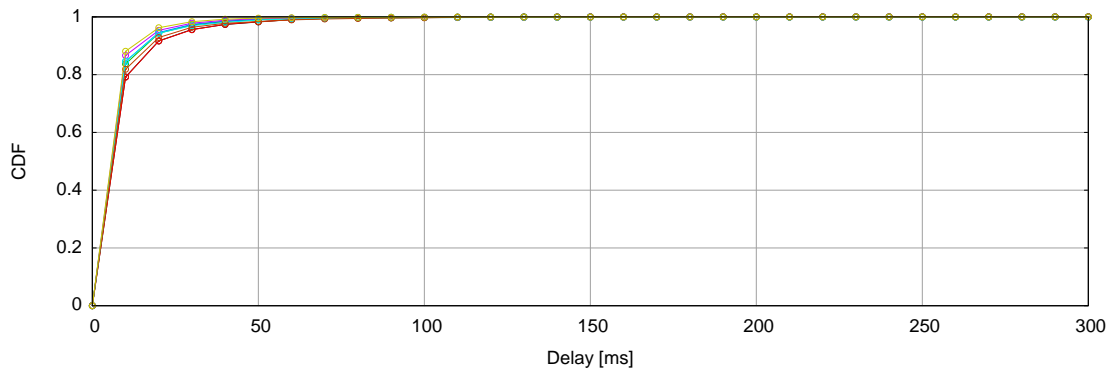


Figure 43: Total delay distribution for three peer mesh call without relay.

Figure 43 shows the receiver rate at each endpoint for a mesh call. We can notice that each endpoint runs as many congestion control mechanisms (RRTCC) as streams flows even the media encoding is done in the same stream multiple times. Comparing with Table 13, we can also see that the rate deviation given on the results (Figure 44) is identified in Figure 43 with an slow ramp-up on the rate during the duration of the call.

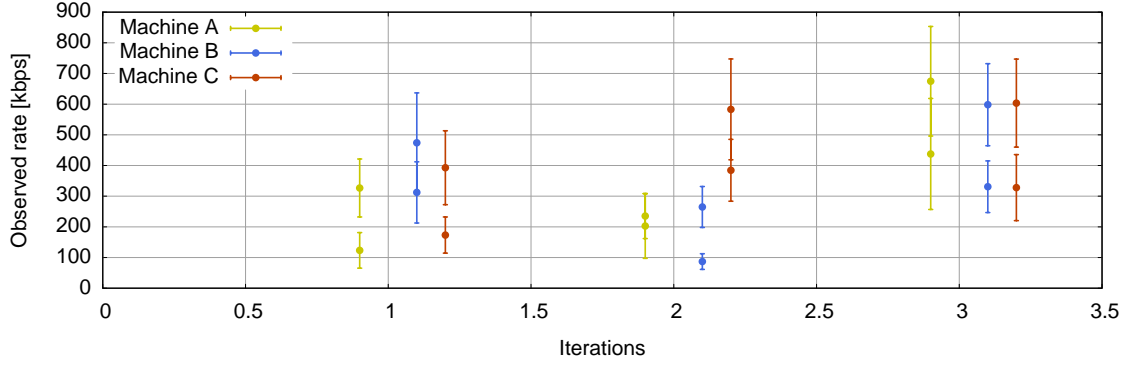
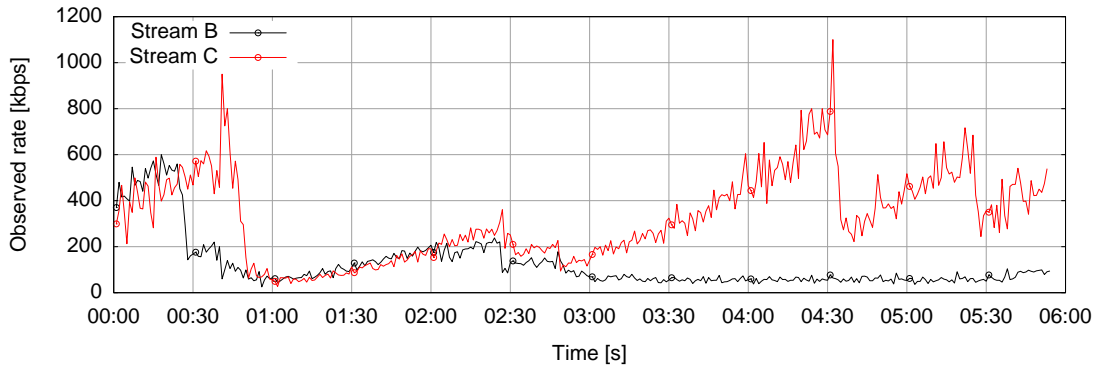
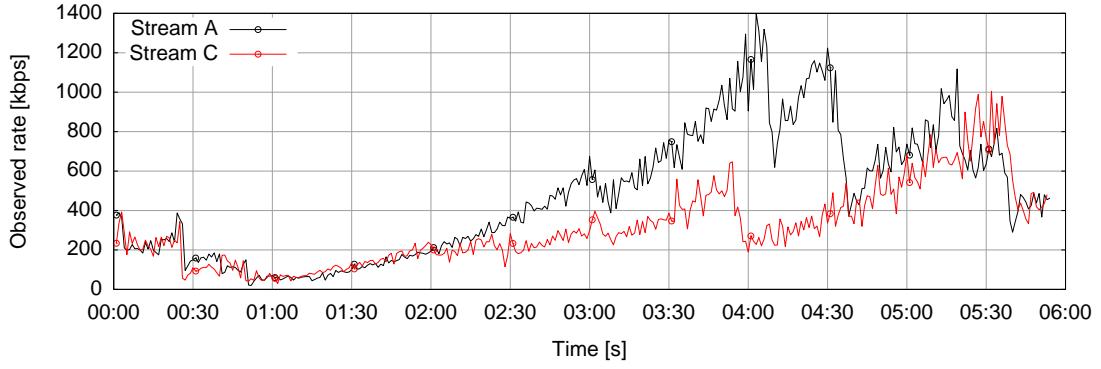


Figure 44: Bandwidth average and deviation for three peers mesh call.

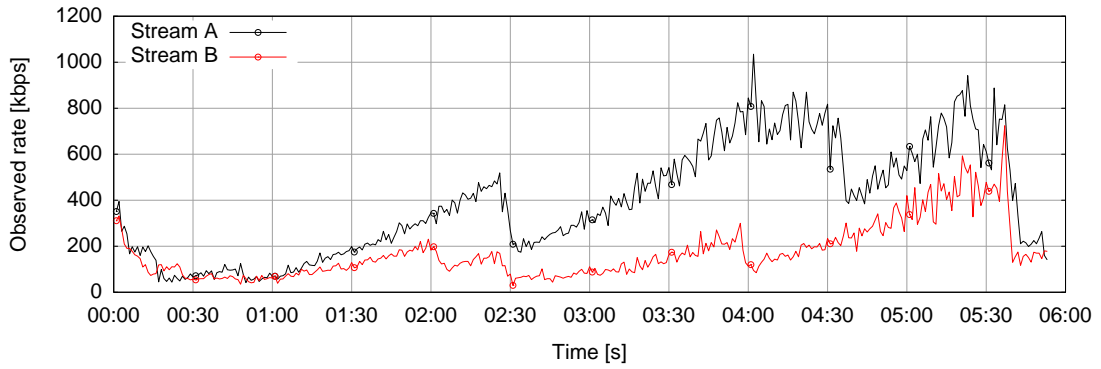
The CDF delay distribution response is good compared to previous scenarios, we can compare the obtained one in Figure 43 with the three parallel calls in Figure 39 and 41. Considering the curve of the delay for the mesh networks we can say that the delay won't be significantly affecting the user experience during the media session. This means that from the perspective of a non relayed mesh call we can have three peers with an acceptable rate and 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.



(a) At peer A



(a) At peer B



(b) At peer C

Figure 43: Variation of rate at the receiver endpoint for every incoming stream in a mesh call.

## 6.7 Multiparty Calls: Switching MCU

One common alternative for multiparty calls is the use of an MCU to perform the relaying of the media through a unique device. There are some MCUs available in the market for WebRTC but the API is still not evolved enough to allow multiplexing of streams over the same Peer Connection. 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.

MCU				
	Machine A	Machine B	Machine C	Overall
<b>Rate (Kbit/s)</b>	604.31±149.38	403.74±93.99	882.94±228.45	630.33±157.27
<b>OWD (ms)</b>	6.88±3.94	6.31±3.8	6.4±3.64	6.54±3.8
<b>Residual Loss (%)</b>	0.05	0.06	0.07	0.02
<b>Packet Loss (%)</b>	0.05	0.06	0.07	0.06

Table 14: Three-peer mesh call with and without TURN.

We group a call of three peers using a centralized conferencing server operating a switching MCU, Figure 44 shows an example of a mesh topology using an MCU. This centralized server is a simple address translator and packet forwarder that receives the media and sends the stream to the individual remote participants. In this specific scenario, our MCU won't be generating the feedback packets for the RRTCC. Furthermore, the MCU collects the feedback packets from the endpoints and sends them back to the sender.

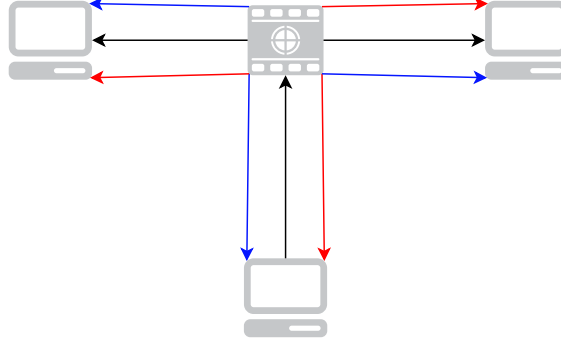


Figure 44: Mesh topology using a centralized MCU.

As result, even though an endpoint sends one media stream, it receives back feedback reports for two participants. With this type of behavior the sender might get some conflict when taking decisions about the rate adaptation, as it might get two different reports stating to increase and decrease the rate. There is no specific solution for this issue in the RRTCC algorithm and it takes the best decision for each case. Table 14 shows the results for this scenario. We can see that the rate has doubled but the deviation has also increased.

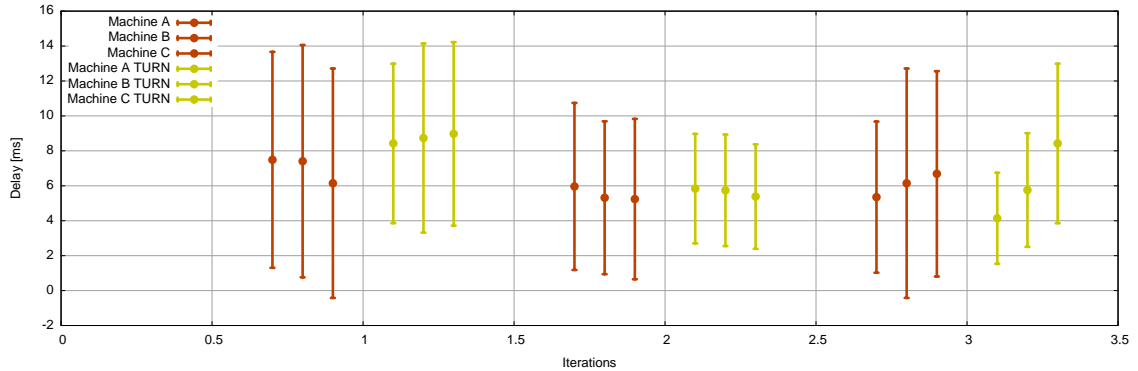
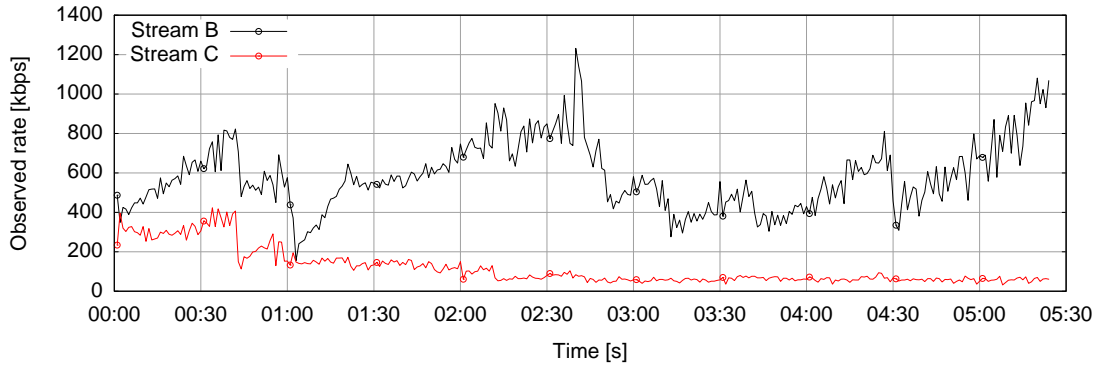


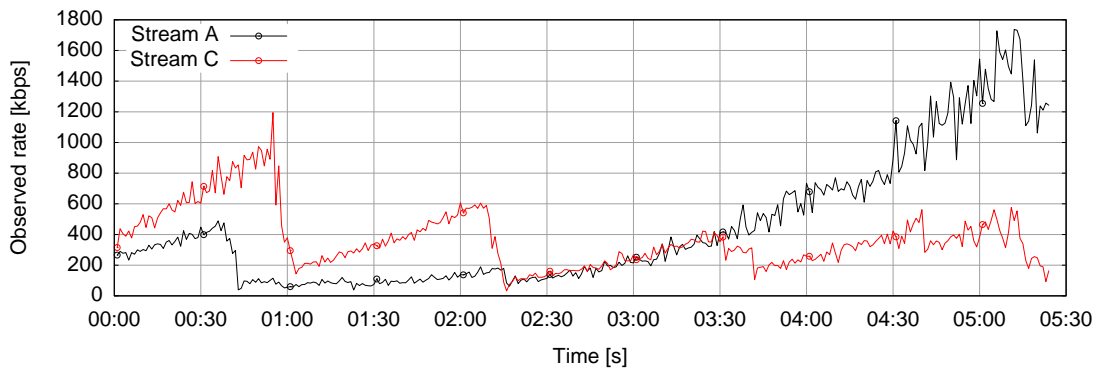
Figure 45: Averaged delay and deviation for TURN and non relayed mesh call for all iterations.

Observing Figure 46 we can study the instantaneous rate at each endpoint. In some specific cases the link is unable to carry all the three media streams at a constant rate, as result one stream suffers from low rate. Stream C is suffering from this issue in Figure 46a and starves at both endpoints. We can observe a

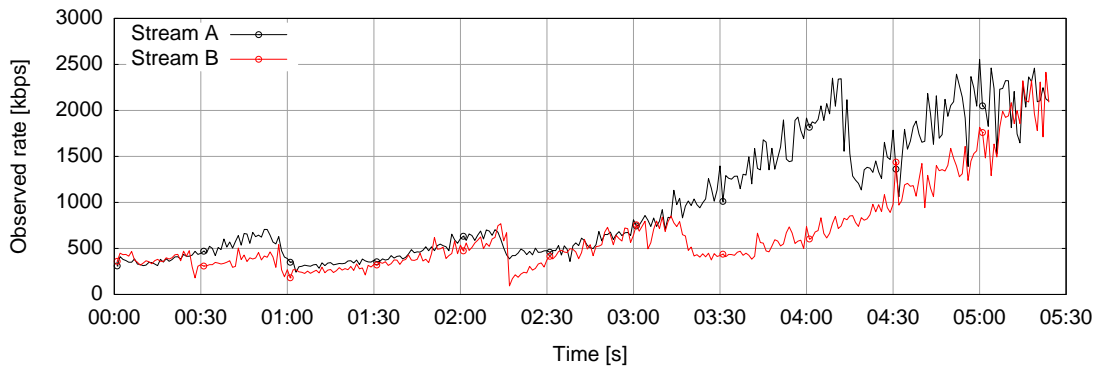
comparison between the previous scenario delay (no MCU) and the actual test in Figure 45.



(a) At peer A



(b) At peer B



(c) At peer C

Figure 46: Variation of rate at the receiver endpoint for every incoming stream in a mesh call with MCU.

Figure 45 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 scenarios.

## 6.8 Interoperability

One of the main goals of WebRTC is to provide interoperability between different browser vendors. In this chapter, we analyze if it is possible for all browsers to be interoperable with their implemented congestion control mechanisms.

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, those should be interoperable. Google Chrome has included WebRTC *PeerConnection* API for long time, but Mozilla Firefox delivered the *PeerConnection* API much later [18]. Making both browsers compatible required work from the vendors. Both engines should be able to understand their signaling messages and respond with the adequate SDP answer.

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 while the API from Chrome sends the candidates separately by default, at the same time, the signaling messages from Chrome includes some extra information about the streams that is not given in the Firefox version [54].

Furthermore, both browsers must implement cross-compatible codecs to encode 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 [54].

One of the goals when checking the interoperability of the browsers, is to evaluate the congestion mechanisms already implemented in both internal engines. They should be able to manage the different environments in a similar way. At the time of writing 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.

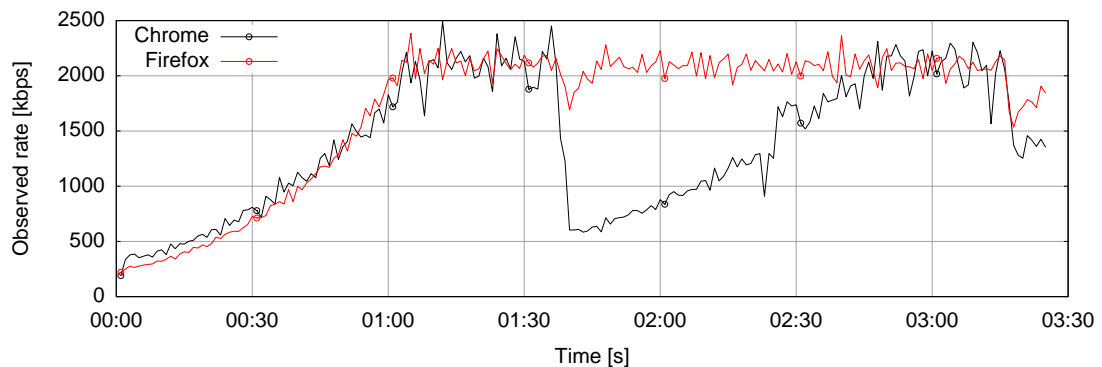


Figure 47: Stream rate for a point-to-point call between Mozilla Firefox and Google Chrome.

Figure 47 shows the given rate of the streams from both vendors captured at each receiver during the call. We can observe that congestion mechanism is not

triggered in Firefox when some congestion occur on the path, Chrome applies rate adaptation while Firefox maintains the same stream rate over the link.

We should consider that Chrome RTP feedback messages use Receiver Estimated Maximum Bitrate (REMB) extra field for rate adaptation [34] [50]. This mechanism provides an extra field in the RTP feedback 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 Wireshark <sup>10</sup>. However, it is important to advise that some of the congestion mechanisms, such as REMB, are not detectable by this software as it might be unable to decode non-standard fields.

Firstly, we obtained a capture from a call between two Chrome browsers, after analyzing the data, we can observe that both peers were exchanging *Sender Reports* control packets. In RRTCC, 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 Chrome. We also notice that WebRTC multiplexes the RTP and RTCP packets in the same port to avoid extra usage of network resources [39].

Those messages carry important information required by the WebRTC internals to build the *Stats API*. The following fields (Listing 7) are extracted from a RTCP packet sent by Google Chrome during the test. We can see that most of the metrics are available in the packet, other metrics such as REMB are not able to be decoded [39].

---

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
```

---

<sup>10</sup><http://www.wireshark.org/>



```

    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, other features might be negotiated by the WebRTC signaling engine, but are not required by the *PeerConnection* if they are not implemented. This is why features such as REMB may still not provide the RTCP *Sender Report*, if this field is not available, the congestion mechanism of Chrome calculates the estimated rate by using RRTCC mechanisms [34], those mechanisms use the information extracted from the RTCP report (7).

During this test with different vendors we have observed different behavior in the RTCP mechanisms between Chrome and Firefox. While Chrome continuously provides 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 Listing 7. No information about local stream in Firefox is being sent to Chrome, this forces Chrome not to provide any feedback control messages to Firefox that would trigger their congestion control mechanisms. This behavior may affect the rate adaptation mechanisms in Firefox providing an output similar to Figure 47.

We can state that congestion mechanisms on Firefox are still not available and this avoids any rate adaptation in Firefox, in conclusion, using Firefox in multiple scenarios would lead to unexpected rate response and poor call quality. Besides from the congestion control mechanisms, Firefox and Chrome APIs are fully interoperable to perform calls.

## 6.9 Mobile Environment

In this section, we test the response of the RRTCC algorithm when using mobile networks, for this test we have used a mobile device with 3G connectivity that connects to a desktop device connected via ethernet. Figure 48 represents the delivered rate in this scenario.

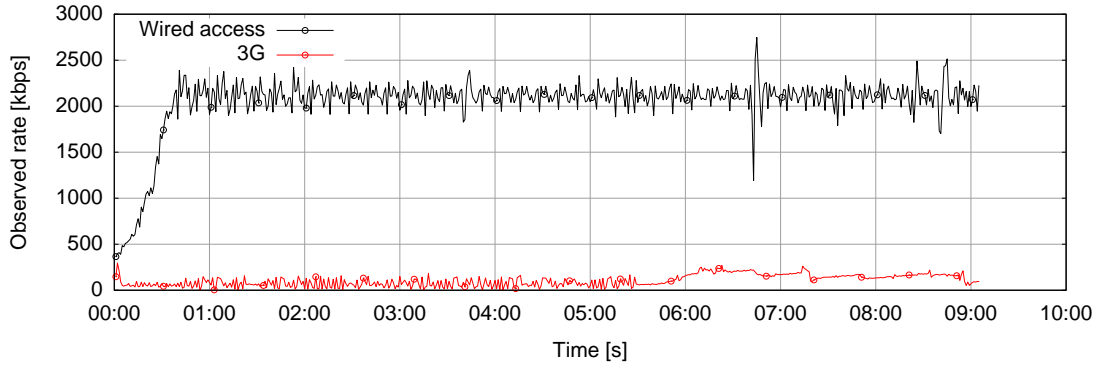


Figure 48: Rate obtained in mixed 3G and wired internet scenario.

We can observe that the average rate for the wired internet stream is about 2 Mbps which is the maximum rate given by the RRTCC. However, the rate obtained in the 3G connectivity is approximately 200 Kbps.

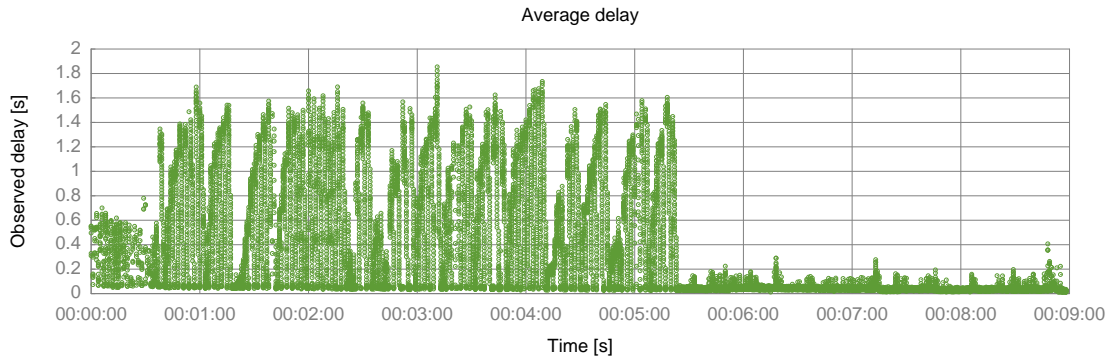


Figure 49: Delay response for the 3G stream in Figure 48.

Another important fact in this scenario is to observe the delay response. Figure 49 shows the delay produced in the 3G stream of Figure 48, we can observe a large delay of over a second in some periods of the call. After the period of big delay there is a sudden drop to 200ms. This delay response can be produced by the 3G signal or the RRTCC mechanism on the mobile device. During the first part of the call there is no rate increase due the delay produced on the path. Once the delay is attenuated, the rate increases and stabilizes for the rest of the call where the throughput is acceptable for a 3G environment. RRTCC is not able to adapt the rate with the delay produced when using 3G environment.

## 6.10 Session Establishment Time

Another sensible factor in a WebRTC call is to analyze the setup time required for each scenario. Setup time is defined as the time required from when the point in time when the local video access is accepted until the remote stream starts to play. Figure 50 show the average time that each scenario requires to start a call.

We can see that the scenario that takes longer to succeed is the one with cross traffic on it. This can be problematic as the existence of cross traffic in the same environment is something common for most internet applications. We also have to consider that in our tests the devices used for the cross-traffic test had to acquire the same local media multiple times which leads to an increase of time in the session setup.

Furthermore, another interesting fact is the time difference required for the multiparty calls. The usage of MCU increases the time required to setup the call by half a second due to the need to relay the packets.

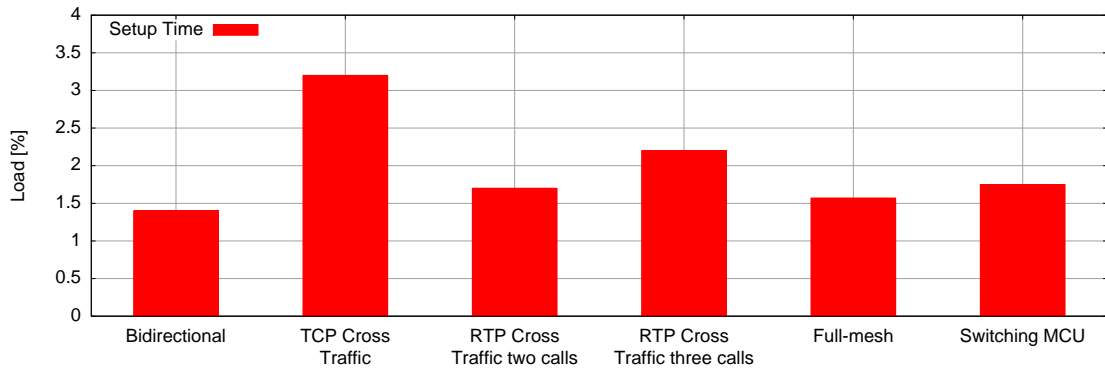


Figure 50: Average setup time (in seconds) required for each different scenario, Parallel2 or 3 indicates the amount of peers in the test.

## 6.11 Resource Analysis

Lastly, we study the impact of WebRTC in terms of performance in our testing devices. For the tests we have used standard VirtualBox virtual machines with Ubuntu fitted on them. Figure 51 shows the averaged used resources in each different scenario. The only processes running during the tests have been the required tools for itself, no extra software is executed. We have measured the performance by using the scripts in Appendix C.

We can see that WebRTC consumes a large amount of resources. Furthermore, when more than one stream is being handled the CPU usage increases. The amount of CPU used in the three parallel calls test compared with the mesh can be compared. CPU usage in both scenarios is  $\approx 90\%$ , the consumption in the three parallel calls is still greater as there are three different local streams being accessed and played on the test, meanwhile in the case of mesh there is only one local stream captured and sent to the rest of peers. When the CPU usage hits the maximum value, the quality of the call might be affected due to encoding limitations.

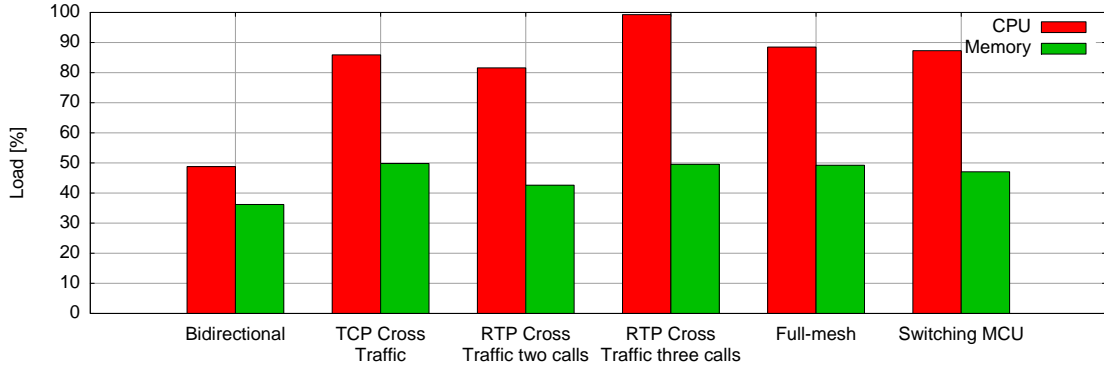


Figure 51: CPU and Memory usage in all the different executed tests, Parallel2 or 3 indicates the amount of peers in the test.

## 6.12 Summary of results

After all the performed tests we can conclude that RRTCC algorithm implemented in Google Chrome performs well in low latencies networks up to 200ms but it collapses in networks with larger values, this condition might be given in some specific environments such as mobile networks or long distance paths.

On the other hand, sending data rate under-utilizes the channel when competing with other TCP traffic on the same path, this is done to avoid an increase of latency. This increase of latency could provoke a rate reduction and directly affect the user experience.

RRTCC response improves when sharing the capacity with similar RTP flows. When there are time-shifted flows, old streams can starve and state a low rate for a short period of time. This happens when multiple devices establish a time-shifted video call using the same bottleneck.

With mesh environments we can conclude that there is an under-utilization in the capacity of the path, this is given as a result of using independent congestion control engines for each flow. Each participant sends its local media in a lowered rate than expected.

Meanwhile APIs are cross-compatible, we can state that WebRTC does not perform well when using different browser providers as the congestion control mechanisms haven't been fully implemented in all platforms.

Furthermore, our analysis suggest that RRTCC is still not ready to be used in mobile environments due the high delay produced by the network. Other analysis related to setup time, performance and failure rate state that WebRTC has to improve its performance to deliver a better user-experience.

We can summarize that RRTCC works well in low delay networks and can tolerate transient changes, competing correctly with some variable cross-traffic, within some limitation.

## 7 Conclusion

In this thesis we have analyzed and evaluated WebRTC protocol for real-time web applications in different environments. We also compared this protocol with already existing real-time communications alternatives and described the usage of the actual implementation of WebRTC. Furthermore, we chose a set of key indicators that are important when measuring the performance of a RTC protocol. Those indicators are used when evaluating the congestion control mechanisms of WebRTC. We also described the possible real-time topologies used to test the performance of WebRTC, some of them are still not possible to implement due to API constraints. To evaluate the protocol we also built a specific setup for this thesis, the environment is used during the development of all the thesis. Finally, we executed the tests after describing the proposed congestion control mechanisms and the ones already implemented in WebRTC.

After the tests, we can conclude that WebRTC is a solid protocol for real-time web applications that performs correctly in constrained environments with low latencies, up to 200ms, but cannot hold greater values. This condition may affect mobile applications relying on WebRTC. The congestion control mechanism implemented in WebRTC, Receiver-side Real-time Congestion Control (RRTCC), copes correctly with packet losses protecting the packets with different mechanisms such as Forward Error Correction (FEC). To improve the performance of WebRTC, different congestion control mechanisms that react to other indicators could be implemented in the internals of the browsers. Given the results obtained in this thesis, we can conclude that RRTCC is still unfinished and requires to evolve further to fit all the requirement of different real time scenarios.

Future researches could be done in order to determine the usability limits of WebRTC. Future works include a deeper experience in mobile environments with the two possible implementations: native mobile application and mobile web application. Both are crucial for the expansion of WebRTC into mobile platforms. Besides this, the analysis of WebRTC session on mobile moving nodes is also important to check the response of the call quality.

WebRTC APIs also need enhancements in order to test all possible environments. One of the most interesting areas to investigate on is the development of a transcoding MCU in WebRTC. During this thesis we have used a packet relying MCU environment for the tests, the transcoding MCU multiplexes and mixes the media streams into a unique channel to improve performance over the path. This type of environment is commonly used for multiparty calls. A better understanding of the Stats API implemented in WebRTC is also needed to better adequate the constraints of the media acquisition and the PeerConnection API. Those metrics are provided by the browser internals and can provide valuable feedback for the application developer that can be used to improve the quality of the session.

Lastly, a follow-up of the cross-compatibility of the browser congestion control mechanisms should be studied in order to provide full interoperability between vendors and devices. At the development of this thesis, this compatibility is not achieved due to the lack of congestion control mechanisms in some browser providers.

## References

- [1] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler. SIP: Ssession Initiation Protocol. <http://www.ietf.org/rfc/rfc3261.txt>, June 2004.
- [2] NetMarketShare. Market Share Statistics for Internet Technologies. <http://www.netmarketshare.com/>.
- [3] Daniel C. Burnett, Adam Bergkvist, Cullen Jennings, and Anant Narayanan. Media Capture and Streams. <http://dev.w3.org/2011/webrtc/editor/getusermedia.html>, December 2012.
- [4] Salvatore Loreto. WebRTC: Real-Time Communication between Browsers. <http://www.sloreto.com/slides/Aalto022013WebRTC/slides.html>, 2013.
- [5] The Future of Real-Time Video Communication, 2009.
- [6] Jonathon N. Cummings and Brian Butler. The quality of online social relationships, 2000.
- [7] Web Real-Time Communications Working Group. <http://www.w3.org/2011/04/webrtc/>, May 2011.
- [8] Harald Alvestrand. Welcome to the list! <http://lists.w3.org/Archives/Public/public-webrtc/2011Apr/0001.html>, April 2011.
- [9] Adam Bergkvist, Daniel C. Burnett, Cullen Jennings, and Anant Narayanan. WebRTC 1.0: Real-time Communication Between Browsers. <http://www.w3.org/TR/2011/WD-webrtc-20111027/>, October 2011.
- [10] Real-Time Communication in WEB-browsers. <http://tools.ietf.org/wg/rtcweb/>, May 2011.
- [11] Magnus Westerlund, Cullen Jennings, and Ted Hardie. Real-Time Communication in WEB-browsers charter. <http://tools.ietf.org/wg/rtcweb/charters?item=charter-rtcweb-2011-05-03.txt>, May 2011.
- [12] Google release of WebRTC source code. <http://lists.w3.org/Archives/Public/public-webrtc/2011May/0022.html>, June 2011.
- [13] J. Rosenberg and H. Schulzrinne. An Offer/Answer Model with the Session Description Protocol (SDP). <http://tools.ietf.org/html/rfc3264>, June 2012.
- [14] M. Thornburgh. Adobe Secure Real-Time Media Flow Protocol. <http://tools.ietf.org/html/draft-thornburgh-adobe-rtmfp>, February 2013.
- [15] Adobe. Cirrus FAQ. <http://labs.adobe.com/wiki/index.php/Cirrus:FAQ>, 2012.

- [16] MS Open Tech publishes HTML5 Labs prototype of a Customizable, Ubiquitous Real Time Communication over the Web API proposal. <http://msopentech.com/blog/2013/01/17/>, January 2013.
- [17] Stefan Alund. Browser - The World First WebRTC-Enabled Mobile Browser. <https://labs.ericsson.com/blog/bowser-the-world-s-first-webrtc-enabled-mobile-browser>, October 2012.
- [18] Serge Lachapelle. Firefox and Chrome interoperability achieved. <http://www.webrtc.org/blog/firefoxandchromeinteropachieved>, February 2013.
- [19] Adam Bergkvist, Daniel C. Burnett, Cullen Jennings, and Anant Narayanan. WebRTC 1.0: Real-time Communication Between Browsers. <http://dev.w3.org/2011/webrtc/editor/webrtc.html>, January 2013.
- [20] C.Holmberg, H. Alvestrand, and C. Jennings. Multiplexing Negotiation Using Session Description Protocol (SDP) Port Numbers. <http://tools.ietf.org/html/draft-ietf-mmusic-sdp-bundle-negotiation>, 2013.
- [21] H. Alvestrand. Overview: Real Time Protocols for Brower-based Applications. <https://datatracker.ietf.org/doc/draft-ietf-rtcweb-overview/>, 2012.
- [22] R. Huang, R. Even, and V. Singh. Consideration for Selecting RTCP Extended Report (XR) Metrics for RTCWEB Statistics API. <http://tools.ietf.org/html/draft-huang-xrblock-rtcweb-rtcp-xr-metrics>, July 2013.
- [23] J. Uberti and C. Jennings. Javascript Session Establishment Protocol. <http://tools.ietf.org/html/draft-ietf-rtcweb-jsep>, October 2012.
- [24] J. Rosenberg. Interactive Connectivity Establishment (ICE). <http://tools.ietf.org/html/rfc5245>, 2010.
- [25] J. Rosenberg, R. Mahy, P. Matthews, and D. Wing. Session Traversal Utilities for NAT (STUN). <http://tools.ietf.org/html/rfc5389>, 2008.
- [26] J. Rosenberg, R. Mahy, and P. Matthews. Traversal Using Relays around NAT (TURN). <http://tools.ietf.org/html/rfc5766>, 2010.
- [27] JM. Valin, K. Vos, and T. Terriberry. Definition of the Opus Audio Codec. <http://tools.ietf.org/html/rfc6716>, 2012.
- [28] R. Jesup, S. Loreto, and M. Tuexen. RTCWeb Datagram Connection. <http://tools.ietf.org/html/draft-ietf-rtcweb-data-channel>, 2012.
- [29] R. Stewart. Stream Control Transmission Protocol. <http://tools.ietf.org/html/rfc4960>, 2007.

- [30] E. Rescorla and N. Modadugu. Datagram Transport Layer Security Version 1.2. <http://tools.ietf.org/html/rfc6347>, 2012.
- [31] S. Dhesikan, D. Druta, P. Jones, and J. Polk. DSCP and other packet markings for RTCWeb QoS. <http://tools.ietf.org/html/draft-ietf-rtcweb-qos-00>, October 2012.
- [32] J. Babiarz, K. Chan, and F. Baker. Configuration Guidelines for DiffServ Service Classes. <http://tools.ietf.org/html/rfc4594>, 2006.
- [33] C. Perkins and V. Singh. Multimedia Congestion Control: Circuit Breakers for Unicast RTP Sessions. <http://tools.ietf.org/html/draft-ietf-avtcore-rtp-circuit-breakers>, 2013.
- [34] H. Alvestrand, H. Lundin, and S. Holmer. A Google Congestion Control Algorithm for Real-Time Communication. <http://tools.ietf.org/html/draft-alvestrand-rmcat-congestion>, 2012.
- [35] C. Holmberg, S. Hakansson, and G. Eriksson. Web Real-Time Communication Use-cases and Requirements. <http://tools.ietf.org/html/draft-ietf-rtcweb-use-cases-and-requirements>, December 2012.
- [36] E. Rescorla. Security Considerations for RTC-Web. <http://tools.ietf.org/html/draft-ietf-rtcweb-security>, January 2013.
- [37] A. Barth. The Web Origin Concept. <http://tools.ietf.org/html/rfc6454>, 2011.
- [38] E. Rescorla. RTCWEB Security Architecture. <http://tools.ietf.org/html/draft-ietf-rtcweb-security-arch>, January 2013.
- [39] C. Perkins, M. Westerlund, and J. Ott. Web Real-Time Communication (WebRTC): Media Transport and Use of RTP. <http://tools.ietf.org/html/draft-ietf-rtcweb-rtp-usage>, October 2012.
- [40] J. Ott, S. Wenger, N. Sato, C. Burmeister, and J. Rey. Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF). <http://www.ietf.org/rfc/rfc4585.txt>, 2003.
- [41] V. Singh. Conmon: App for monitoring connections. <http://vr000m.github.com/ConMon/>, 2013.
- [42] Creytiv. Restund. <http://www.creytiv.com/restund.html>.
- [43] Marta Carbone and Luigi Rizzo. The dummynet project. <http://info.iet.unipi.it/~luigi/dummynet/>.
- [44] E. N. Gilbert. Capacity of a Burst-Noise Channel. *Bell Syst. Tech.*, 39(9):1253–1265, 1960.



- [45] E. Elliot. Estimates of Error Rates for Codes on Burst-Noise Channels. *Bell Syst. Tech.*, pages 1977–1997, 1963.
- [46] Harinath Garudadri, Hyukjune Chung, Naveen Srinivasamurthy, and Phoom Sagnetong. Rate Adaptation for Video Telephony in 3G Networks. *Proc. of PV*, 2007.
- [47] V. Singh, J. Ott, and I. Curcio. Rate-control for Conversational Video Communication in Heterogeneous Networks. *Proc. of IEEE WoWMoM Workshop*, 2012.
- [48] X. Zhu and R. Pan. NADA: A Unified Congestion Control Scheme for Real-Time Media. <http://tools.ietf.org/html/draft-zhu-rmcat-nada>, March 2013.
- [49] P. O’Hanlon and K. Carlberg. Congestion control algorithm for lower latency and lower loss media transport. <http://tools.ietf.org/html/draft-ohanlon-rmcat-dflow>, April 2013.
- [50] H. Alvestrand. RTCP message for Receiver Estimated Maximum Bitrate. <http://tools.ietf.org/html/draft-alvestrand-rmcat-remb>, 2013.
- [51] M. Handley, S. Floyd, J. Padhye, and J. Widmer. TCP Friendly Rate Control (TFRC): Protocol Specification. <http://tools.ietf.org/html/rfc5348>, 2008.
- [52] Xunqi Yu, James W. Modestino, and Dian Fan. Evaluation of the Residual Packet-Loss Rate Using Packet-Level FEC for Delay-Constrained Media Network Transport. <http://info.iet.unipi.it/~luigi/papers/20091201-dummynet.pdf>, November 2006.
- [53] Marta Carbone and Luigi Rizzo. Dummynet Revisited, 2009.
- [54] Interoperability notes between Chrome and Firefox. <http://www.webrtc.org/interop>, 2013.
- [55] Patrick Hglund. Broken PyAuto test: WebRTC Ignores Fake Webcams. <https://code.google.com/p/chromium/issues/detail?id=142568>, August 2012.
- [56] Patrik Hglund. V4L2 File Player. [https://code.google.com/p/webrtc/source/browse/trunk/src/test/linux/v4l2\\_file\\_player/?r=2446](https://code.google.com/p/webrtc/source/browse/trunk/src/test/linux/v4l2_file_player/?r=2446).
- [57] Kernel Timer Systems: Timer Wheel, Jiffies and HZ. [http://elinux.org/Kernel\\_Timer\\_Systems](http://elinux.org/Kernel_Timer_Systems), 2011.

## A Setting up fake devices in Google Chrome

To address the issue of the video transferred from our automated devices we have built a fake input device on the virtual machines that is fed with a RAW YUV file. This device is configured using a hacked version of the *V4L2Loopback* code which is originated from the *V4L* driver in Linux. The modified version of the *V4L2Loopback* builds two extra devices in the Linux interface mapping. This is done as Chrome is unable to read from the same reading/writing device for security reasons. The first device is used to feed the video and the other one to read it [55] in an input/output configuration.

The following additions have been done in the *V4L2Loopback* code:

- 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.

---

```
strcpy(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 Höglund [55] for the *V4L2Loopback* hacked version.

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

Now we should see two extra video devices in our system. The following step is to feed the */dev/video1* with a YUV file. In order to do this, we use the *V4l2 File Player* [56], this player executes on top of *Gstreamer* but adds a loop functionality to the file allowing long calls to succeed by looping the video multiple times. YUV sample videos can be obtained from the Network Systems Lab page.<sup>11</sup>

```
# 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.

---

<sup>11</sup>[http://nsl.cs.sfu.ca/wiki/index.php/Video\\_Library\\_and\\_Tools](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 different path conditions in our tests.

However, *Dummynet* has been natively developed for *FreeBSD* platforms and the setup for *Linux* environments is sometimes not successful. Our devices run with Ubuntu Server 12.10 with a 3.5.0 kernel version inside VirtualBox, this setup requires us to modify some code in *Dummynet* in order to achieve good test results.

The accuracy of a network 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 [53]. 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 internal clock in a Linux kernel 2.6.13 and above is 250Hz [57]. The previous 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 recompilation of itself. This change will reduce the timing error from 4ms (default) to 1ms.

Once the kernel timing is done we need to compile the *Dummynet* code, the version we are using in our tests is 20120812. *Dummynet* can be downloaded from the *Dummynet* project site [43].

We should try the code first and check if we are able to set defined queues to our network pipes, this procedure 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 *./ipfw/dummynet.c* and *./ipfw/glue.c* files.

---

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);
}

```

---

This issue arise from 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 parameter 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 parameter 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, *limit = XXX* should determine the value of the queue maximum length in slots and Bytes. Slots in for this thesis are defined considering a maximum MTU size of 1500 Bytes.

By default, maximum queue size is set to 100 slots, this setup is not designed for large bandwidth tests over 10Mbit/s. In order to modify this we will need to set a higher value according to the maximum required for our setup. Once this is set we need to recompile *Dummynet* from the root directory of the download source code and follow the install instructions in the README file.

Furthermore, once we have allowed *Dummynet* to accept more than 100 slots we might not be able to configure them into the network pipe. The last 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 in *FreeBSD* instead of Linux.

We need to modify 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 every time the system starts. Finally, 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 save 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 ./common 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 common >> 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 common >> 1to1.log 2>&1
killall v4l2_file_player >> 1to1.log 2>&1
killall chrome >> 1to1.log 2>&1

```

---

---

Listing 9: Measure 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=(100*($DIFF_TOTAL-$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
```

---