Conversas Hiperligadas: Novo Paradigma de Comunicação e Colaboração, potenciado pela Tecnologia WebRTC

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Abstract

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1 Introduction

The need to build a global comunication network in an era when almost nobody had access to it, caused that some protocols weren't suitable for a huge increase on the amount of publicly known users. IPv4 limits the number of public addresses in such a way that today are scarse [1]. One way to overcome this problem was the development of a mechanism that groups multiple address into a single one, the machine that is assigned that address is then responsible to redirect messages to members of its group through their private addresses, each element is identified publicly by the same IP address but different ports, this technique is also know as Network Address Translation (NAT).

Initially NAT offered an alternative for address exhaustion and a false sensation of better security, asymetric NAT became a vulgar configuration on the web. As a direct result, problems started to appear, the amount of ports that IP disponibilizes is also small compared to our current needs, worse than that, NAT also difficults end-to-end communication, forcing most of applications that follows this model to be implemented unificiently.

Applications based on multimedia and file sharing were one of the most strained by NAT. Those kind applications requires real time communication in order to achieve the best performance. STUN and TURN [2] servers are a possible solution to overpass NAT, although, none of those an establish direct connections on multiple level NATs.

Most of client-server applications aren't affected by NAT when the servers are public, but they aren't suitable for real time comunication between two private end points. Clearly this type of communication requires a more expensive infrastructure and, at most cases, more network usage, leading to a worse quality of service. The requirements of video comunication makes this kind of model out of question.

On the other hand, TURN uses public servers to redirect traffic between private end points, it may use a P2P network relay to find the best peer, but after that the behaviour is much like client-server. Direct communication is only achieved by STUN when NAT is type

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full cone. [reference to NAT types]. ICE uses STUN when it's possible and TURN otherwise.

When connection is established, either direct or indirect (via TURN servers), WebRTC cames to simplify how audio and video are transmited.

2 Hypermedia

Since the early days of video technology, one of the problems that raised with it consisted on how to add more information onto it without generating multiple versions. The need to translate movies, raised the problem wether is apropriate to change the original video or audio. For example subtitles should be an entity independent from the video, in order to be personalized or replaced easily.

Amongst multiple formats for subtitles, SAMI, SRT and SUB are used by video players. Although those formats have styling available, they are limited to text.

SMIL was introduced to describe temporal behaviour of multimedia, for instance, it could be used to overlay subtitles on films. With SMIL it's possible to synchronize multiple sections of video, either in parallel or in sequence, reproduce a diferent audio, overlay user interface elements with hyperlinks, amongst multiple other functionalities.

In order to create a multimedia rich hypercall, SMIL fits our goals, but it lacks on browser compatibility. Ambulant was one of the SMIL players that were developed for browsers, although this player implements most of SMIL 3.0 specifications, it needs to be installed on browsers as a plugin.

SmilingWeb attempts to implement SMIL 3.0 with javascript, which doesn't need to be installed and shouldn't have incompatibility issues. But SmillingWeb isn't fully implemented yet and their sheduler engine loads the SMIL file only once, which could raise problems when leading with SMIL changes in real time.

SVG is a format that implements part of SMIL [which version and functionalities?]

Although video and audio is almost standardized, WebRTC allows us to enrich web communications by overlaying multiple elements such as buttons, text and more complex user interfaces.

Our solution to overcome format limitation is to take advantage of languages like javascript, like any webpage, user interface elements are defined by html tags, more complex actions like animations can be easily implemented via CSS and javascript.

By using this already solidified web technologies, it's possible to raise communications to a new level. For example, with API's like WebGL, it is now possible to manipulate a three dimensional environment in the context of an hypercall. Another example would be a colaborative spreadsheet using WebRTC. With this, hypercalls are not limited to only video, audio and text.

In this project our goal is to enrich hypercalls with no limits, every user should be free to choose how it whant to be contacted.

2.1 Trash

Consider this case study, say that Alice whants to call Bob. Alice starts a WebRTC connection with Bob, at this point Alice doesn't know how the communication will proceed. Bob created a personal web page where he defined how he whants to be contacted. Alice downloads Bob's personal webpage and finds what kind of interactions could be done, for instance she knows that she can contact Bob via video streaming or can browse past videos. While Bob is connected, he can add links on videos, organise its content and enable different plugins for communication.

References

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[3] A Structured Document Model for Authoring Video-based Hypermedia