Real-Time Communication Network Architecture Design for Organizations with WebRTC

Pedro Vílchez

2015-07-13 Monday

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Why WebRTC?

1. It is Relatively new (2011); standardization process is not ended
2. It is a convergence between telecommunications and the Internet
3. There are already WebRTC applications in the market from telecommunication operators

Contents

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Chapter

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Motivation

- Previous project: start a simple community network[1].
- Now: provide Real-Time Communications (RTC) to a community network such as Guifi.net.
  - Add value to the network
  - Enhance social cohesion within the community

[1] Starting, Contributing and Empowering Community Networks in cities

Objectives

- Free and secure communication between users via an RTC system and community network infrastructure.
- Backward compatibility with VoIP[2] network. Hence, users can communicate to other VoIP operators from inside and/or outside Guifi.net.
- Designing RTC network architecture to fit the community network scenario.
- Ease of installation and usability of RTC.

[2] Voice over IP. The use of telephone adapted to the Internet network.

Chapter

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Fundamentals

- Communications
  - Protocol, signaling, gateway
  - Internet, flexible infrastructure for generic data.
    - Standardization: IETF
  - Traditional telephony, guaranteed delivery for real-time data.
    - Standardization: ITU
  - IETF and ITU did efforts to have networks which process real-time data and generic data
- Basic real-time quality parameters: bandwidth, mean delay, delay variation, etc.

Chapter

Introduction

Fundamentals

State of the art

Methodology

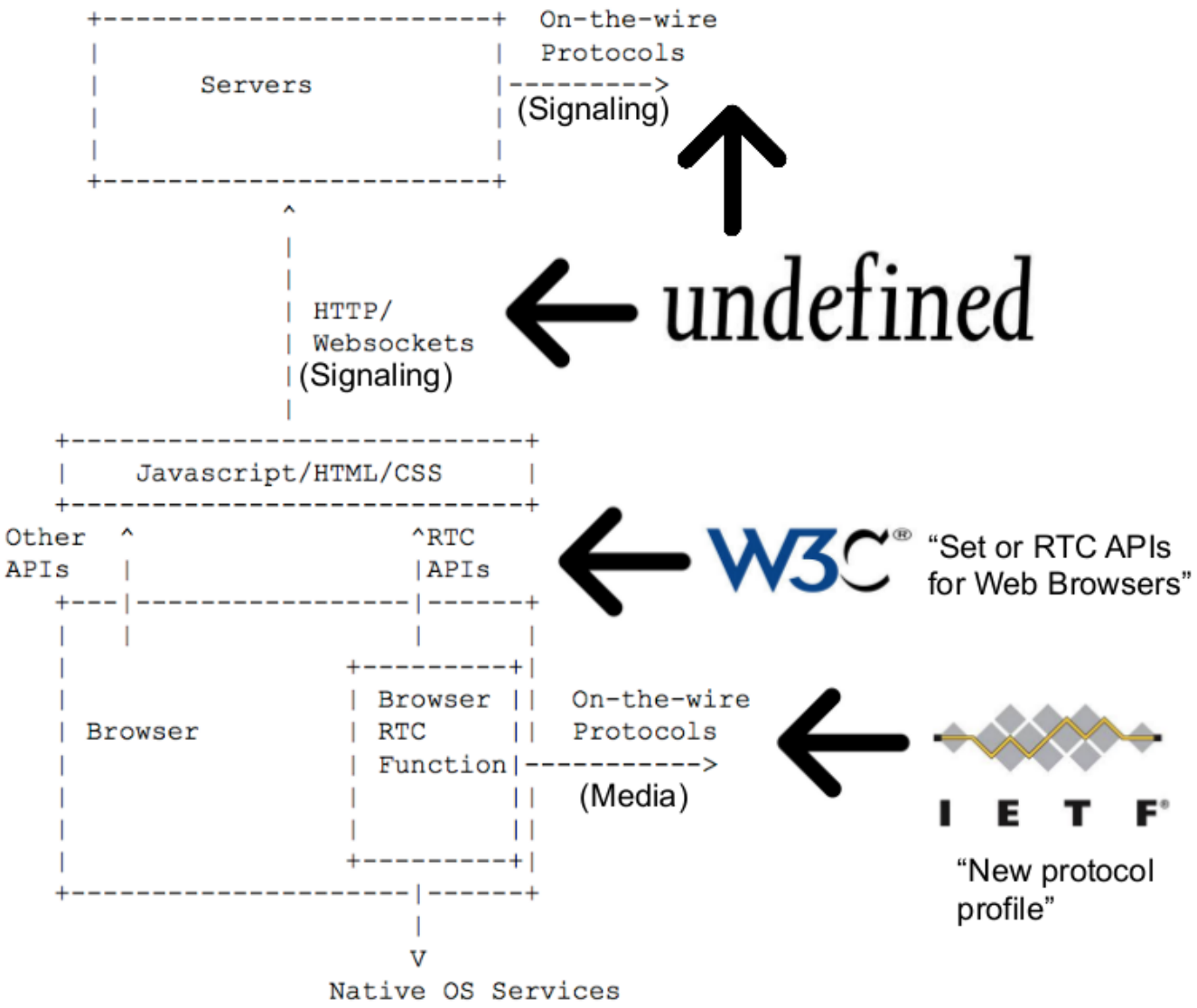
Contributions and Results

Conclusions and Future Work

State of the art

- SIP signaling
  - Nature: establish, modify and terminate multimedia sessions in the Internet
  - Implemented XMPP-based solution (SIMPLE)
- XMPP signaling
  - Nature: manage instant messaging and presence of your contact list
  - Implemented SIP-based solution (Jingle)
- WebRTC
  - Open to any existing signaling method such as SIP or XMPP

WebRTC


Servers
On-the-wire Protocols
(Signaling)
undefined
HTTP/ Websockets
(Signaling)
Javascript/HTML/CSS
Other APIs
^RTC APIs
W3C
"Set or RTC APIs for Web Browsers"
Browser
Browser RTC Function
On-the-wire Protocols
(Media)
I E T F
"New protocol profile"
Native OS Services

Chapter

Introduction

Fundamentals

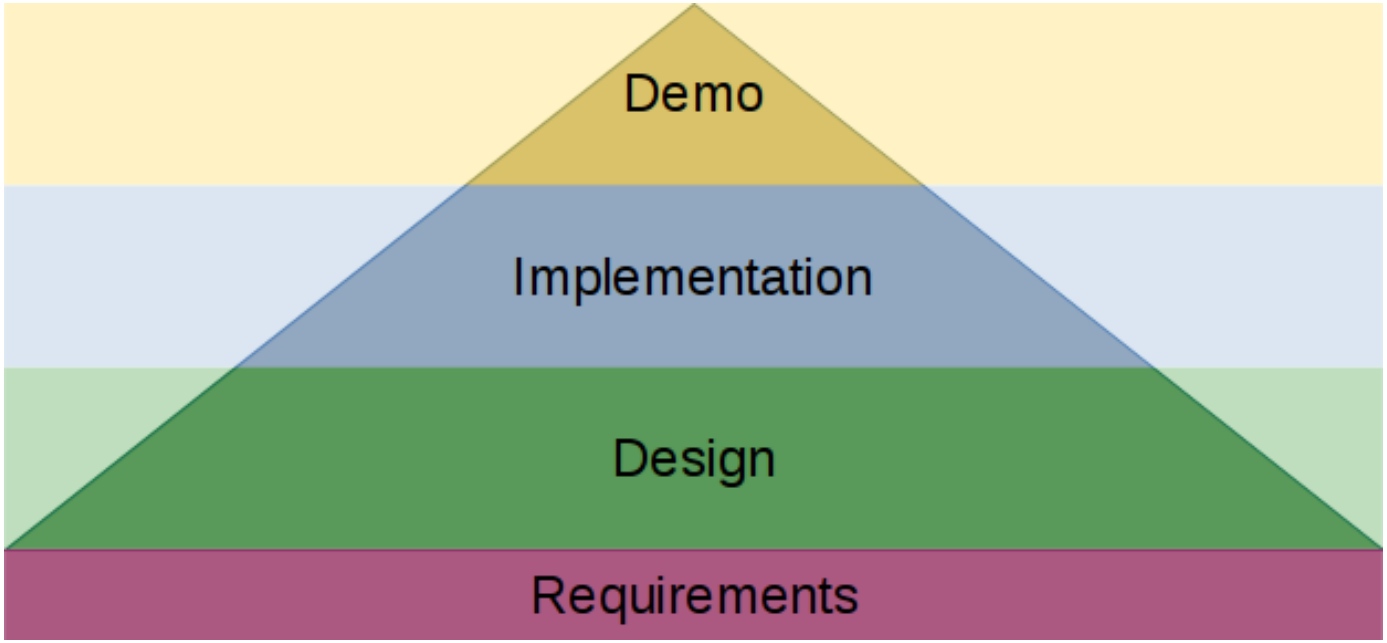
State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Methodology



Chapter

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Network architecture analysis of Guifi.net

The selected scenario to design and implement an RTC system.

- Community: volunteers and professionals
- Government of Guifi.net: manage of Guifi.net's web, legal support
- Behavior, protocols and functionality: similar to Internet
- Network architecture: star topology and mesh
- Links: the vast majority of communication links are wireless
- Known services: Internet sharing and network control

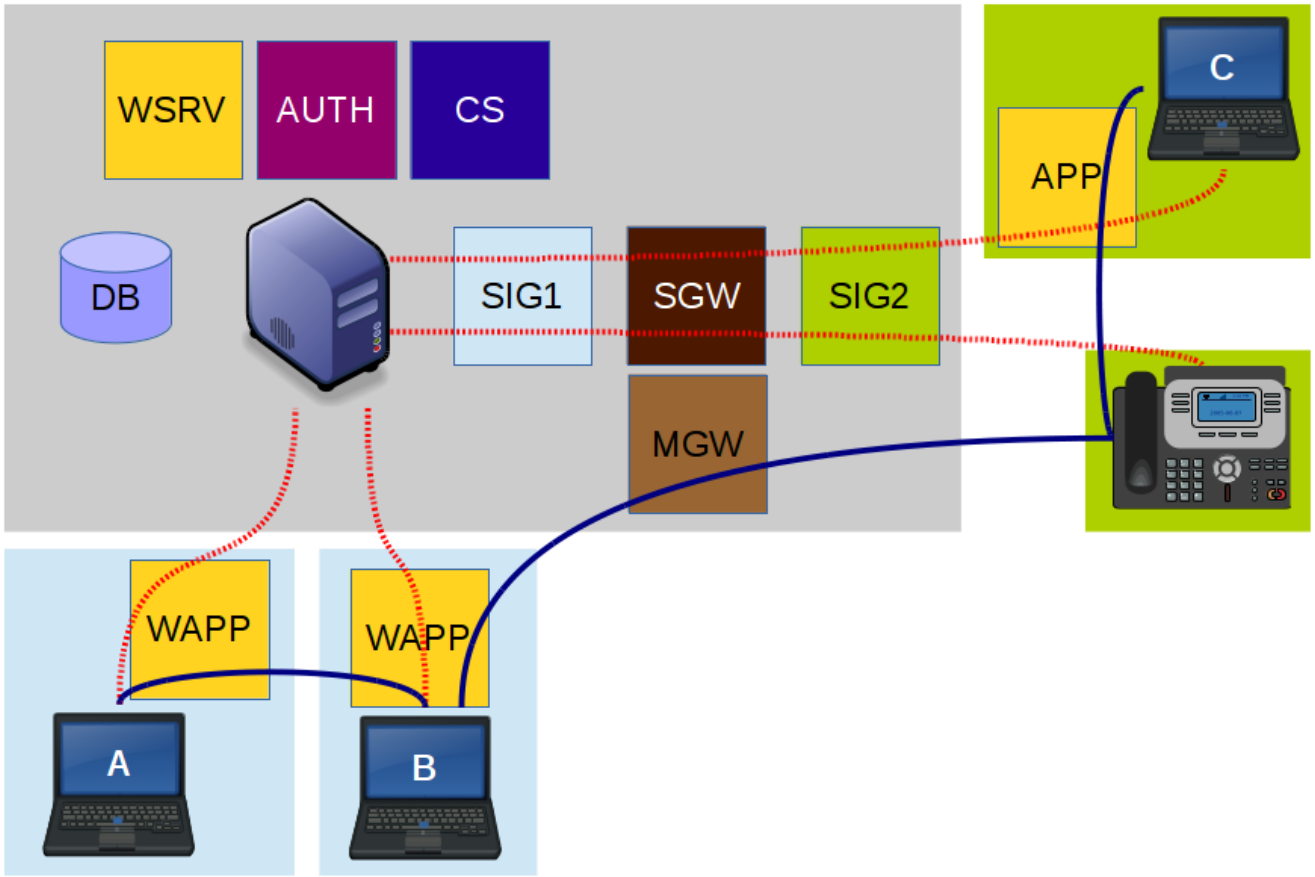
Network requirements

- QoS: differentiate real-time data from non real-time data
  - QoS on Guifi.net is complex
- G.114 says that delay should be below 150ms
  - Guifi.net seems to fit this requeriment
- Bandwidth: Audio 64 Kbps, Video 1-10 Mbps
  - Guifi.net has common links from 15 to 150 Mbps

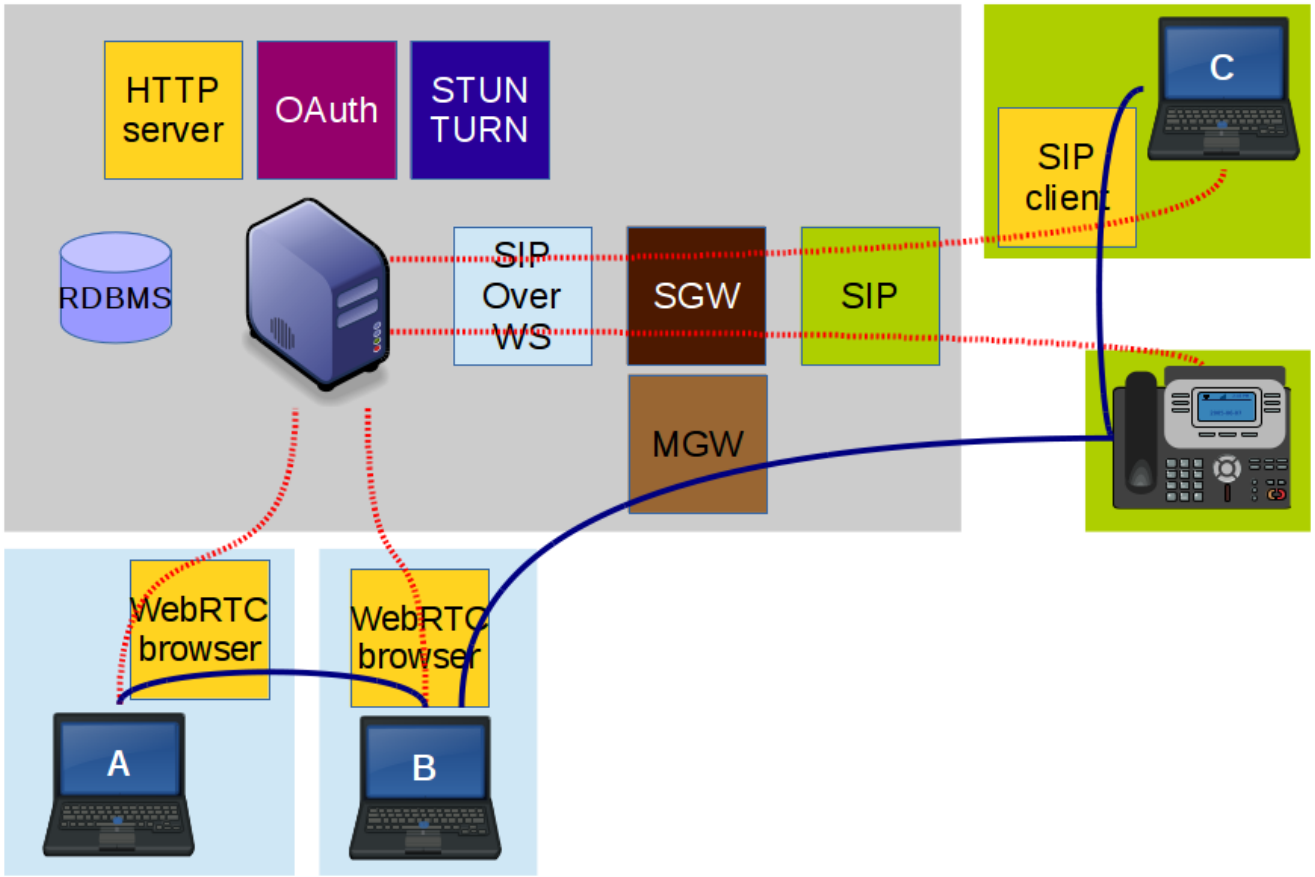
Use cases

- Send calls: a user calls another user with an audio channel. Optional channels of communication if available: video and chat
- Receive calls: a user receives a call only if is connected to the service with at least one device and is available
- Integration: all the services are integrated and is the same account

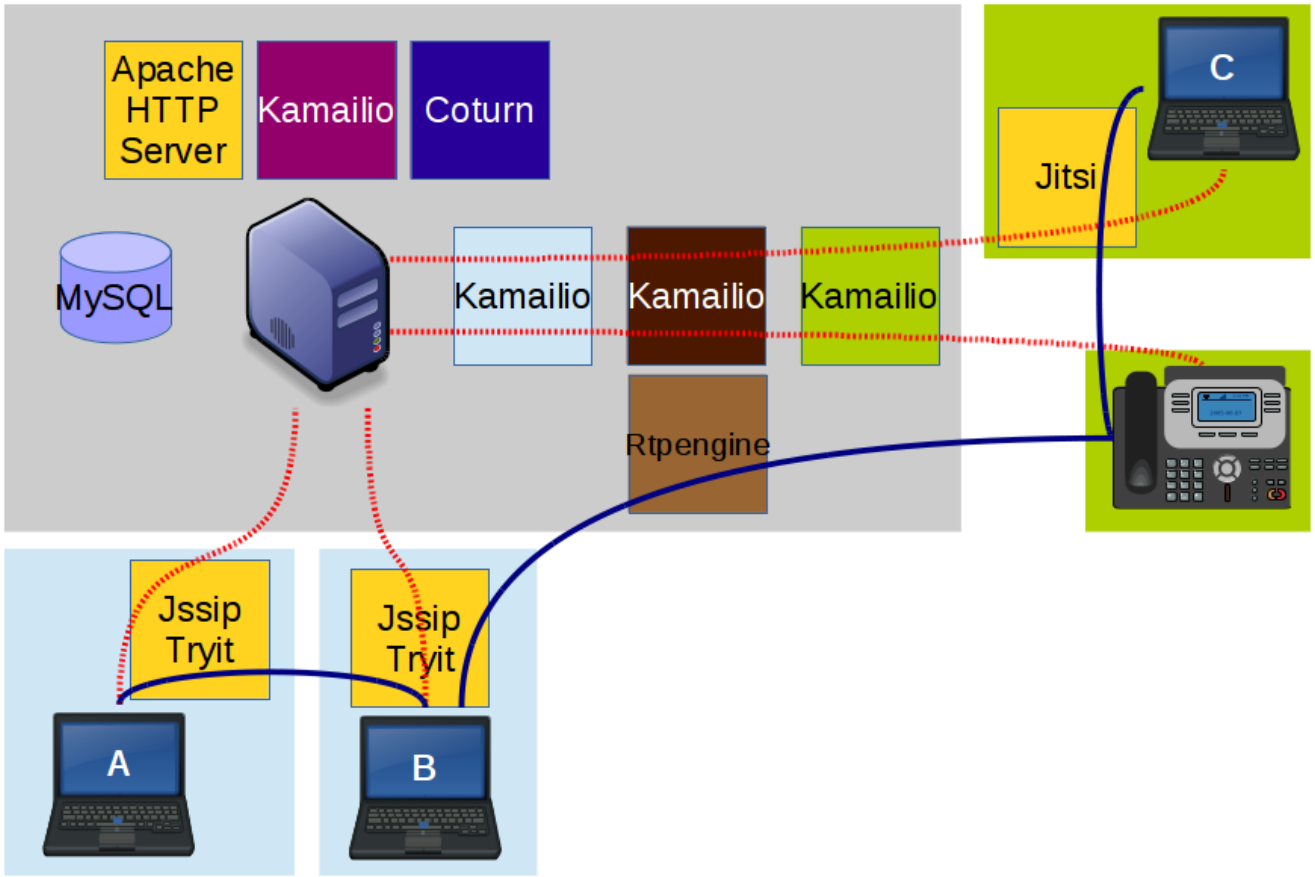
Network architecture design



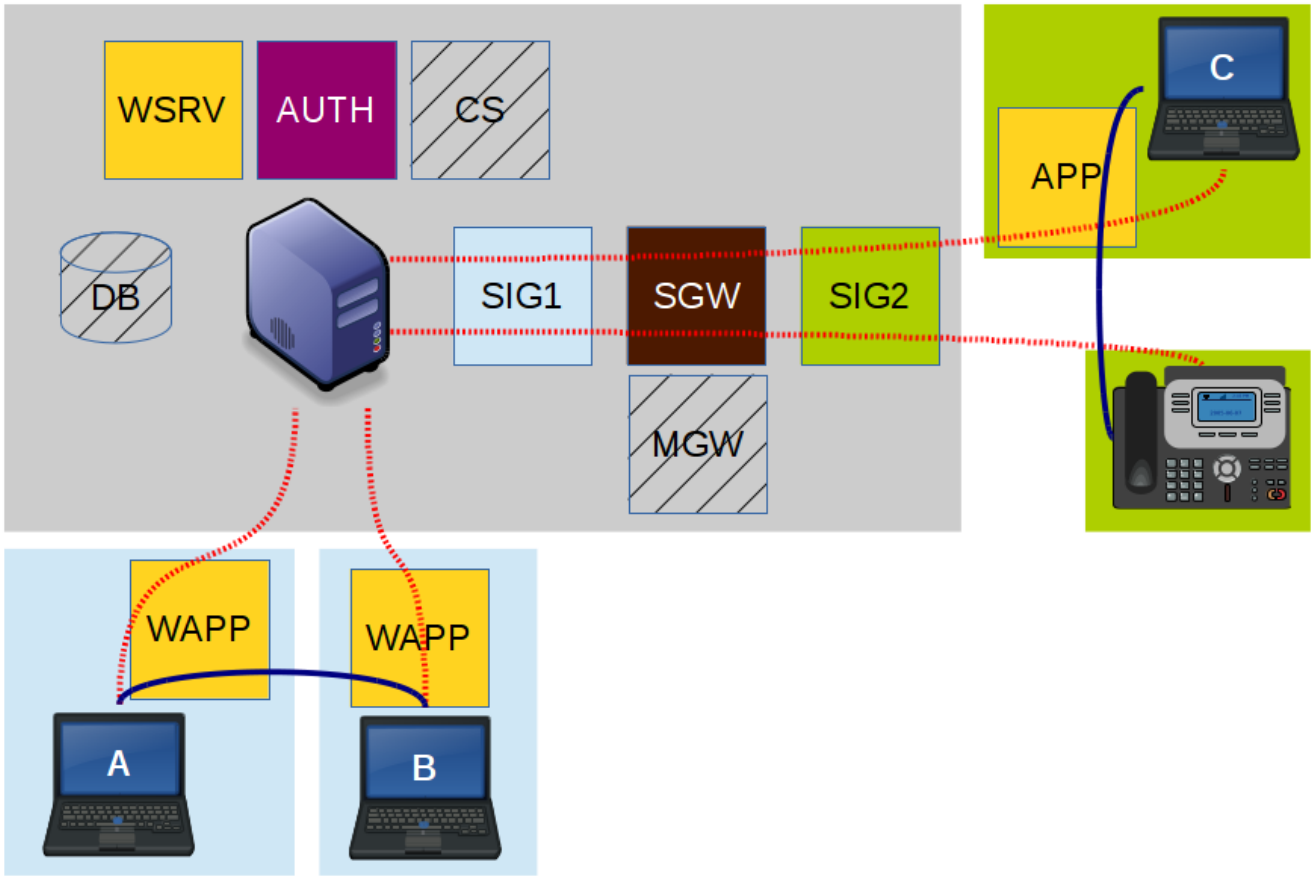
Component selection



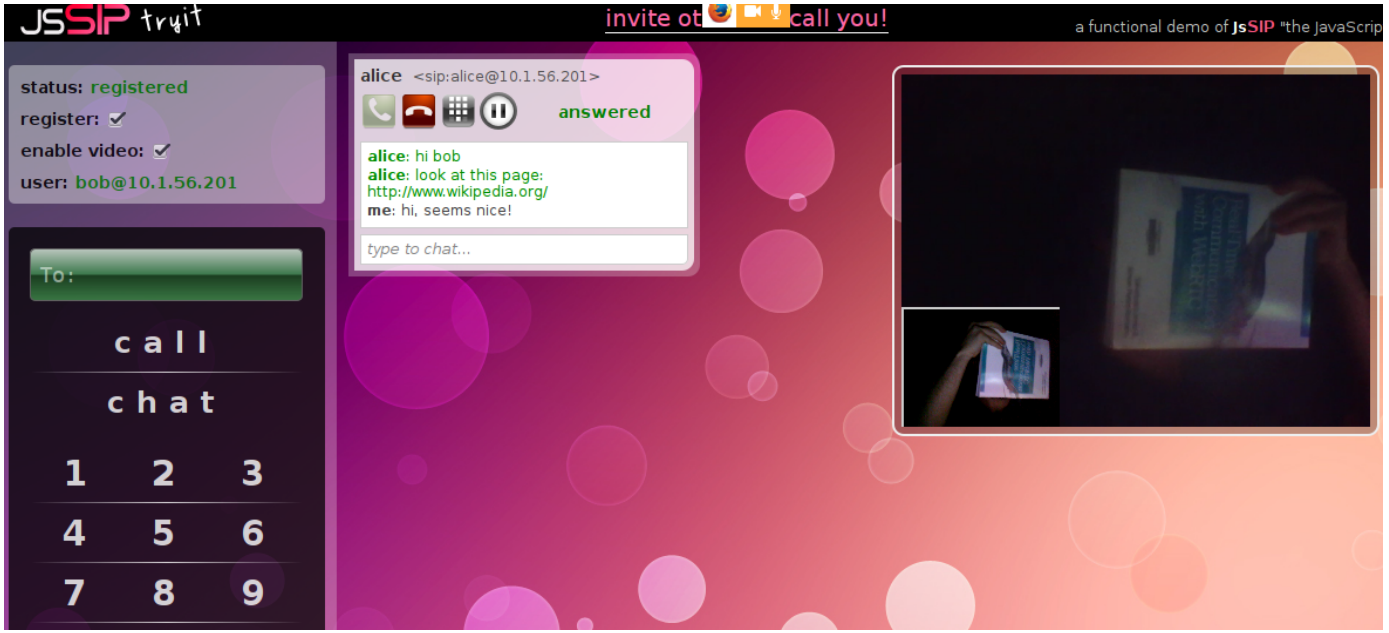
Implementation



Demo Evaluation

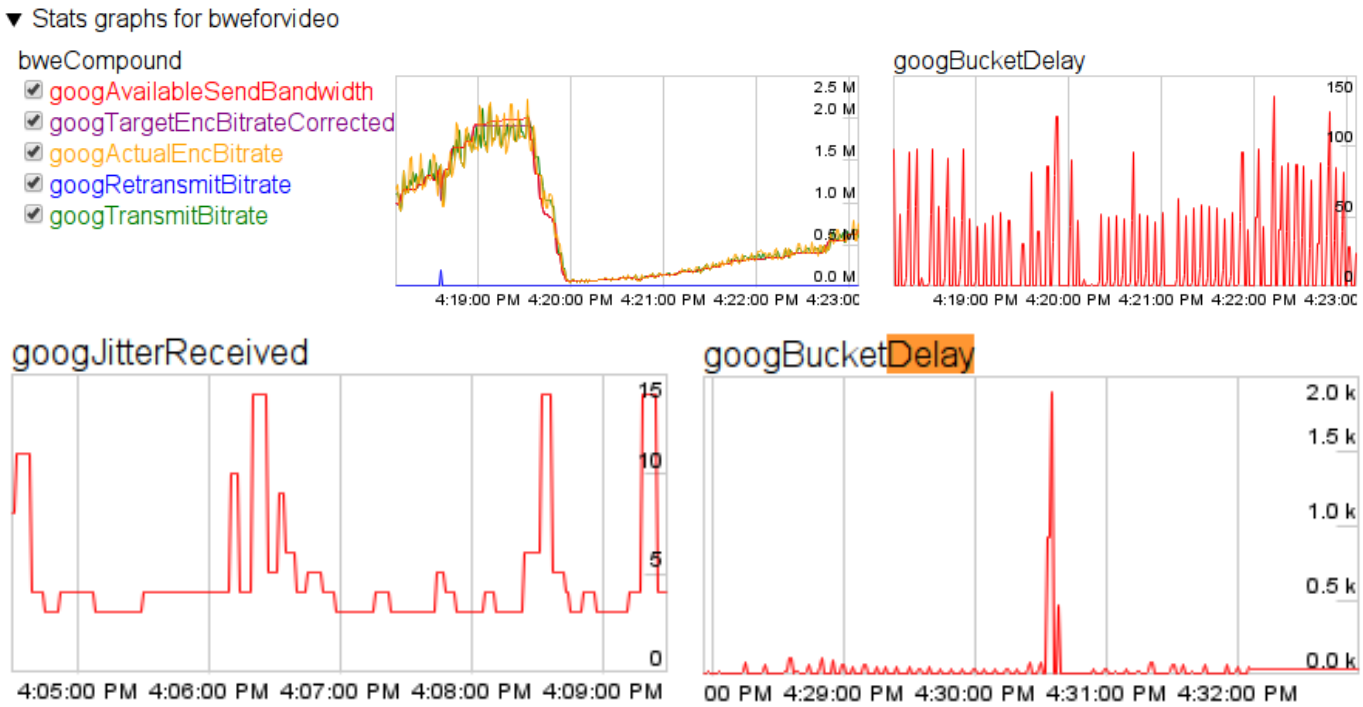


Demo



Installation instructions in the appendix of the memory

Demo Performance



Chapter

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work

Conclusions

- Objectives of the project are attainable
- WebRTC facilitates use of RTC in organizations of all sizes
- WebRTC requires some centralization
- Pending featured topics
  - Anonymity
  - WebRTC media cannot go to HTTP Proxy
  - Interoperability between WebRTC and SIP needs a media gateway

Future Work

- Improve implemented use cases
- Implement remaining use cases
- Stress test of the demo
- Improve integration of XMPP and SIP signaling inside Guifi.net open source firmwares

Thanks! Questions?

Introduction

Fundamentals

State of the art

Methodology

Contributions and Results

Conclusions and Future Work