Janus: back to the future of WebRTC

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Outline

1. A brief introduction

2. A stroll through time
   - IETF activities and “running code”
   - WebRTC related efforts

3. Janus: a general purpose WebRTC gateway
   - WebRTC and gateways
   - Programmable real-time media components
   - A modular and extensible architecture

4. Next steps
Who am I?

- Someone not used to this cold!
  - From sunny Sorrento, Italy 😊

- Current activities
  - Ph.D Student @ UniNA
  - Co-founder @ Meetecho

- Worked on real-time applications for a long time
  - IETF participant
    - Several WGs
    - First time in IETF67 San Diego (2006)
  - Open source contributor
    - libbfcp, libmsrp, confiance, mediactrl, Asterisk, ...
    - Janus WebRTC gateway main author

- Getting older but, unlike whisky, not getting any better
  - https://twitter.com/delusionsmaster
What is Meetecho?

- A startup focused on real-time multimedia applications
  - Academic spin-off of the University of Napoli Federico II
  - Official tool for remote participation @ IETF
- At first born to turn research into a product
  - Efforts on XCON (Centralized Conferencing) and MEDIACTRL (Media Server Control)
  - Meetecho Web Conferencing and Collaboration
- Widened the scope to cover multimedia in general
- How is it pronounced, you say??
  - Good question!
  - https://www.youtube.com/watch?v=TkgDOMSv9PE
A few more words on Meetecho

Meetecho comes in different flavours

- Interactive Webinar (IETF-style)
- Web conference
- UCC (Unified Communication and Collaboration)

What’s in there? Enjoy watching our teaser spot on:

http://www.meetecho.com

- WebRTC audio/video
- Jabber chat
- Slides
- Etherpad support
- Application/desktop sharing

- Whiteboard
- Polling
- Moderation
- Mobile apps
- IPv6 support
First steps: IETF67 @ San Diego (2006)

Don’t try this at home!
First steps: IETF67 @ San Diego (2006)

Demo scenario: XCON

Live demo of the BFCP/CCMP protocols in action (XCON)
Open source project: http://confiance.sourceforge.net
Binary Floor Control Protocol (BFCP)

Open source project: [http://sourceforge.net/projects/libbfcp/](http://sourceforge.net/projects/libbfcp/)
Centralized Conferencing Manipulation Protocol (CCMP)

Authors: CCMP http://tools.ietf.org/html/rfc6503
and examples http://tools.ietf.org/html/rfc6504
XCON/DCON: Stand-alone client
One step further: from XCON to MEDIACTRL

Open source project: http://mediactrl.sourceforge.net
MEDIACTRL: programming media control
MEDIACTRL: programming media control

TF-WebRTC
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Meetecho
History
IETF
WebRTC
Janus
Gateways
Requirements
Architecture
Next steps

Protocol Messages

Control Package 1
Control Package 2
Control Package 3

Header
Payload
MEDIACTRL: programming media control
Prototype and demos @ IETF ("running code")
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Prototype and demos @ IETF ("running code")
More scalability: Media Resource Brokering

First WebRTC steps

- RTCWEB BoF in IETF 80 (March 2011, Prague)
  - First Meetecho RPS “experiment” at the same meeting
- Meetecho already RTP- and web-based for a while
  - RTP Java applet implemented for the purpose
  - Other protocols (SIP, BFCP, XMPP, etc.) wrapped by HTTP
- Started working on WebRTC as soon as it came out
  - https://groups.google.com/forum/#!topic/discuss-webrtc/YhI5fUpx1jc
  - A custom VP8 video mixer for Asterisk MeetMe/ConfBridge
First WebRTC steps

Power Consumption of Hardware VS Software Implementation

• How much power is used? What resolution/frame rates? Overall impact on battery life of mobile devices

RTCWEB @ IETF84 Vancouver
Opus audio codec integration

- Opus is a standard, high quality, adaptive audio codec
  - Mandatory-To-Implement (MTI) in WebRTC!
- Open source library available
  - http://opus-codec.org
- Integrated by us in Asterisk
  - Transcoding support in open source patch
    - https://github.com/meetecho/asterisk-opus
  - Passthrough support in mainstream Asterisk 12
    - https://issues.asterisk.org/jira/browse/ASTERISK-21981
- Tested in Meetecho
  - Streaming (HTML5 <audio>)
  - Multimedia conferences (mixing and transcoding)
Technical Plenary @ IETF87 in Berlin

- Technical Plenary on Opus
- Invited speaker with Google, Mozilla and Jitsi
Technical Plenary @ IETF87 in Berlin

- Super-wideband (48kHz) Opus audio streaming
Meetecho IETF world tour
April 2011-November 2014

Meetecho @ IETF

Sessions

IETF80 Prague  IETF81 Quebec City  IETF82 Taipei  IETF83 Paris  IETF84 Vancouver  IETF85 Atlanta  IETF86 Orlando  IETF87 Berlin  IETF88 Vancouver  IETF89 London  IETF90 Toronto  IETF91 Honolulu
Meetecho @ IETF

Remote Presentations

- IETF80 Prague
- IETF81 Quebec City
- IETF82 Taipei
- IETF83 Paris
- IETF84 Vancouver
- IETF85 Atlanta
- IETF86 Orlando
- IETF87 Berlin
- IETF88 Vancouver
- IETF89 London
- IETF90 Toronto
- IETF91 Honolulu
“Director” room @ NOC in Honolulu

Completely WebRTC-based

- Slides as a video feed from the beamer
- Static video feed from the room
- Dynamic video feeds for remote speakers
WebRTC reference architecture
Involving a gateway (and applications)
Involving different technologies as well
Do we really need a gateway?

- Several reasons for a YES, here
  - Relieve full-meshes (heavy on the client side)
  - Leveraging widespread technologies (e.g., SIP infrastructures)
  - Fixing things between implementations

- Reason for a NO?
  - You won’t go beyond 1-1 WebRTC communication
  - You don’t want an infrastructure
  - You don’t care about legacy stuff

“What is a WebRTC Gateway anyway?”
- http://webrtchacks.com/webrtc-gw/
Real-time Media Components

- Writing a gateway from scratch is a heavy task
  - Implementation of the WebRTC protocol suite
- Bridge between “legacy” stuff (SIP, RTMP, etc.) and WebRTC
  - Needs to support both (WebRTC gateway) → J1
  - What about statistics? → D1
  - Reachability may be an issue → D2
- Programmable interface
  - Different applications/technologies, different requirements
  - Dynamic management of media flows and users
  - Something a-la MEDIACTRL? → W1, B1, R1, R2
Janus: a general purpose WebRTC gateway

“In ancient Roman religion and myth, Janus [..] is the god of beginnings and transitions, and thereby of gates, doors, passages, endings and time. He is usually depicted as having two faces, since he looks to the future and to the past.”

Janus: a general purpose WebRTC gateway

- A door between the communications past and future
  - Legacy technologies (the “past”)
  - WebRTC (the “future”)

Janus

General purpose, open source WebRTC gateway

- https://github.com/meetecho/janus-gateway
- Demos and documentation:
  http://janus.conf.meetecho.com

- Design and implementation of the gateway → J3
  - WebRTC suite re-implemented (core)
  - Modular architecture rewritten from scratch
  - Plugins as the MEDIACTRL “packages”
Modular architecture

- The core only implements the WebRTC stack
  - JSEP/SDP, ICE, DTLS-SRTP, Data Channels, ...
  - REST (HTTP) / WebSockets / RabbitMQ APIs
- Application logic implemented in server side plugins
  - Users attach to plugins via the gateway core
  - The gateway handles the WebRTC stuff
  - Plugins route/manipulate the media/data
- Some proof of concept plugins implemented
  - Echo Test (→ Self testing!)
  - Streaming (→ Live events!)
  - Conferencing (→ Communication!)
  - SIP Gateway (→ “Legacy” SIP!)
  - ...

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Extensible Architecture and API
Extensible Architecture and API

Janus Gateway

Core

Plugin 1

Plugin 2

...

Plugin N

Protocol messages
Plugins as “bricks”

- Each plugin is a feature, not an application
- Application can be composed out of different features
  - Features as “bricks” for a complex scenario
- A few examples...
  - Screensharing with Q&A
    - Video MCU (screen) + Video MCU (speakers) + Audio Bridge (questions)
  - Video communication in social networks
    - SIP plugin (calls) + Echo Test (diagnostics) + Voice Mail (messaging)
  - Social TV
    - Streaming (TV channel) + Video MCU (interaction)
Screen Sharing with Q/A

Audio bridge plugin

Video MCU plugin

Audio

Video

Text chat

Shared screen

External feature (not provided by Janus)
Social TV

TV live broadcast

User 1 video
User 2 video
User 3 video
User 4 video

Video MCU plugin

Streaming plugin
What is it used for today?

- We use it ourselves for many things (obviously)
  - Web conferencing
  - Webinars and e-learning
  - Streaming of live events
  - WebRTC-to-SIP gateway
- Many people/companies using it as well in even more creative ways!
  - Coworking
  - TV broadcasting and Social TV
  - Home automation
  - Internet of Things
- New third-party tools are starting to come out
  - New plugins for ad-hoc requirements
  - Server-side API wrappers (node.js, .NET, ...)

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What to do next?

- Finalize the WebRTC implementation
  - Stabilize SSRC multiplexing
  - Add octets (besides strings) to DataChannels
  - Keep up-to-date with newest stuff

- Mobile access
  - Janus stack for Android basically done, to refine
  - Already interoperable with libjingle_peerconnection

- Improve the pluggable architecture
  - Plugins as “filters”, not only “sinks” (e.g., transcoders)
  - Plugins in series and/or in parallel

- Help us improve this!
  - Play with it, more testing is important
  - Write your own plugins/applications!
Questions? Comments?
Related Publications

- International journals

- Book chapters

- Conferences and workshops
  W1 L. Miniero, “Improving the scalability of real-time multimedia applications using brokering of media resources”, InQ 2013, June 13-14, 2013, Sorrento, Italy
  W2 A. Amirante, T. Castaldi, L. Miniero and S. P. Romano, “SOLEIL: Streaming Of Large-scale Events over Internet cLouds”, 11th Italian Networking Workshop, January 15-17, 2014, Cortina d’Ampezzo, Italy

- Request For Comments (RFC)
  R1 C. Boulton, L. Miniero, G. Munson, “Media Resource Brokering”, RFC6917, April 2013

- Internet Drafts
  D1 L. Miniero, S. Garcia Murillo, V. Pascual “Guidelines to support RTCP end-to-end in Back-to-Back User Agents (B2BUAs)”, draft-ietf-straw-b2bua-rctp-02