Janus, or: How I Learned to Stop Worrying and Love WebRTC Gateways

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Outline

What's Meetecho?

- A company born in 2009 as an academic spin-off
  - University research efforts brought to the market
  - Proudly brewed in sunny Napoli, Italy ☺
- Focus on real-time multimedia applications
  - Web conferencing only, at first
  - Then widened the scope to multimedia in general
  - Strong perspective on standardization and open source
    - *WebRTC rulez!*
- Several activities
  - Consulting services
  - Commercial support & licenses
  - Streaming of live events (e.g., IETF, ACM SIGCOMM, ...)
  - Products (conferencing, webinar, ...)

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(*Napoli looks a bit like this...)

Ok, ok, enough about you... what's WebRTC about?

- Real-time media in a browser
- Up to some time ago, no standard solution!
  - No interoperability
  - Plugins needed to be installed anyway

WebRTC = Joint standardization efforts

- Internet Engineering Task Force (IETF)
- World Wide Web Consortium (W3C)

- RTCWEB (IETF)
  - Real-Time Communication in WEB browsers WG
  - Defines protocols and formats to use
- WEBRTC (W3C)
  - Web Real-Time Communications WG
  - Defines UI and API to access devices

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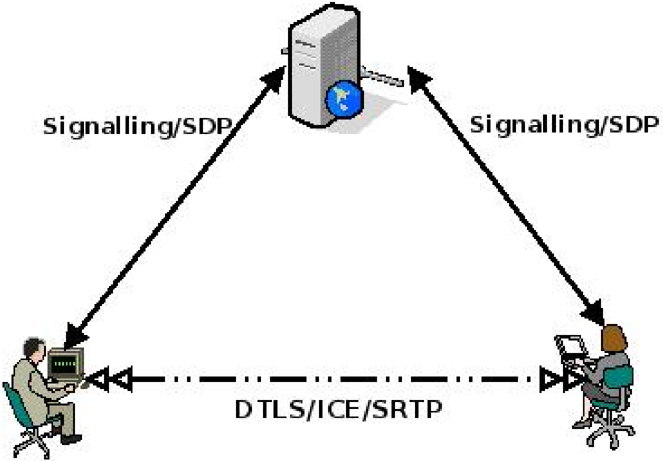
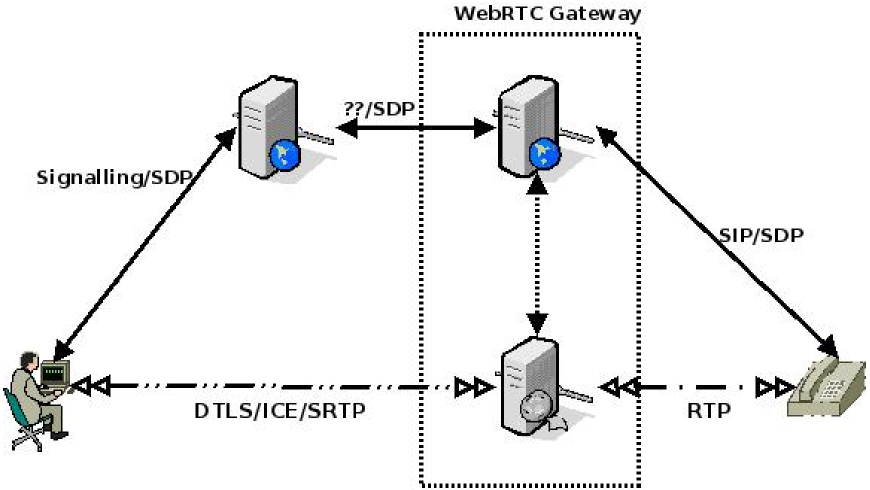
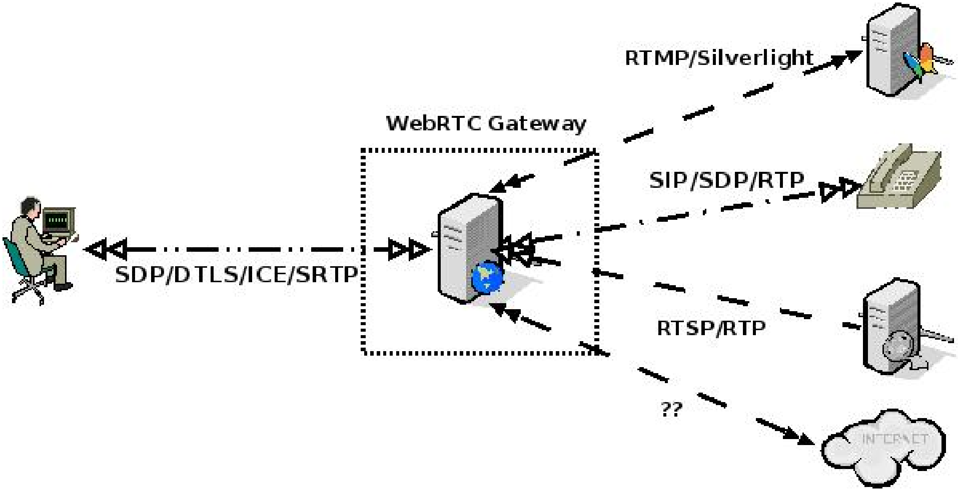
WebRTC reference architecture



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 interaction among few users
  - You don't want an infrastructure
  - You don't care about legacy stuff

"What is a WebRTC Gateway anyway?"

- https://webrtchacks.com/webrtc-gw/

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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)
  - What about statistics?
  - Reachability may be an issue
- Programmable interface
  - Different applications/technologies, different requirements
  - Dynamic management of media flows and users
  - Something *à la* MEDIACTRL?

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The WebRTC protocol suite

- Signalling (well, sort of) and Negotiation
  - Javascript Session Establishment Protocol (JSEP)
  - Session Description Protocol (SDP) adaptation
- Connection Establishment and NAT Traversal
  - Session Traversal Utilities for NAT (STUN)
  - Traversal Using Relay NAT (TURN)
  - Interactive Connectivity Establishment (ICE)
- Media Transport and Control
  - Real-time Transport (and Control) Protocol (RTP/RTCP)
  - Secure Extensions to RTP (SRTP)
  - Datagram Transport Layer Security (DTLS)
- Multimedia codecs
  - Opus audio codec (MTI, Mandatory-to-implement)
  - VP8 and H.264 video codecs (MTI, Mandatory-to-implement)
- Generic Data
  - WebRTC Data Channels (SCTP)

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."

— http://en.wikipedia.org/wiki/Janus

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: https://janus.conf.meetecho.com
- Community: https://groups.google.com/forum/#!forum/meetecho-janus

Modular architecture

- The core only implements the WebRTC stack
  - JSEP/SDP, ICE, DTLS-SRTP, Data Channels, ...
  - Modules for API over HTTP / WebSockets / RabbitMQ
- 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
  - Streaming ($\rightarrow$ Live events!)
  - Video Room ($\rightarrow$ Selective Forwarding Unit!)
  - SIP Gateway ($\rightarrow$ "Legacy" SIP!)
  - ...

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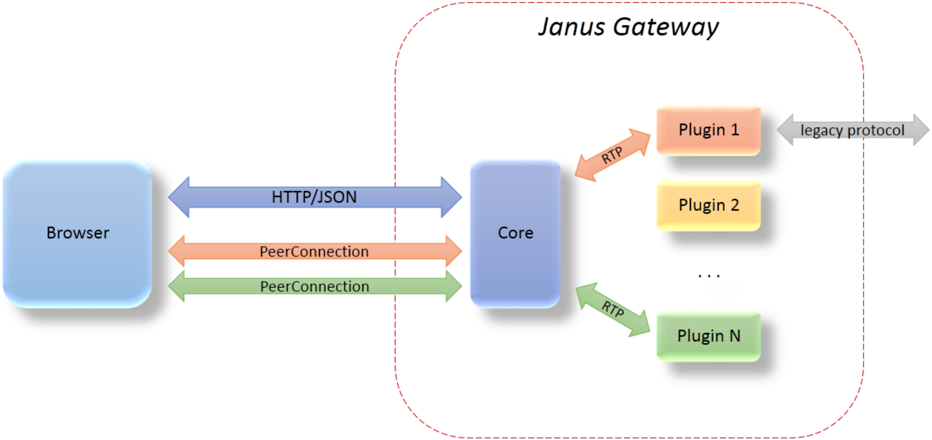
Extensible Architecture and API



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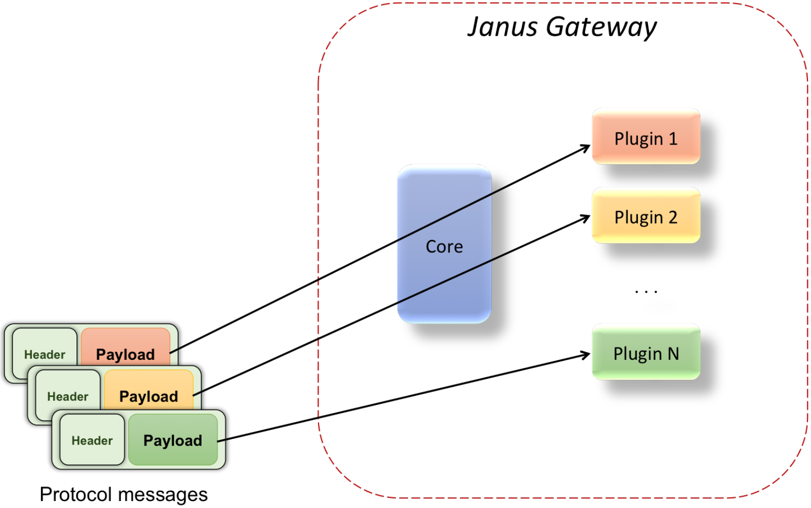

Plugins as "bricks"

- 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...
  - Multimedia conferencing with PSTN support
    - Video Room (participants video & screen) + SIP (participants audio)
  - Webinar with Q&A
    - Video Room (screen) + Video Room (speakers) + Audio Bridge (questions)
  - Social TV
    - Streaming (TV channel) + Video Room (interaction)
  - Contact center / Communication in social networks
    - SIP plugin (calls) + Echo Test (diagnostics) + Record & Play (messaging)

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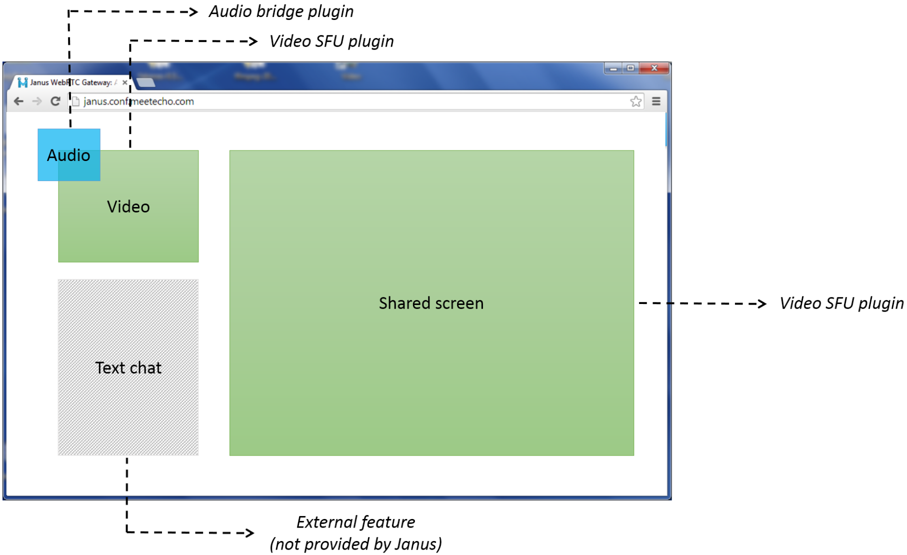
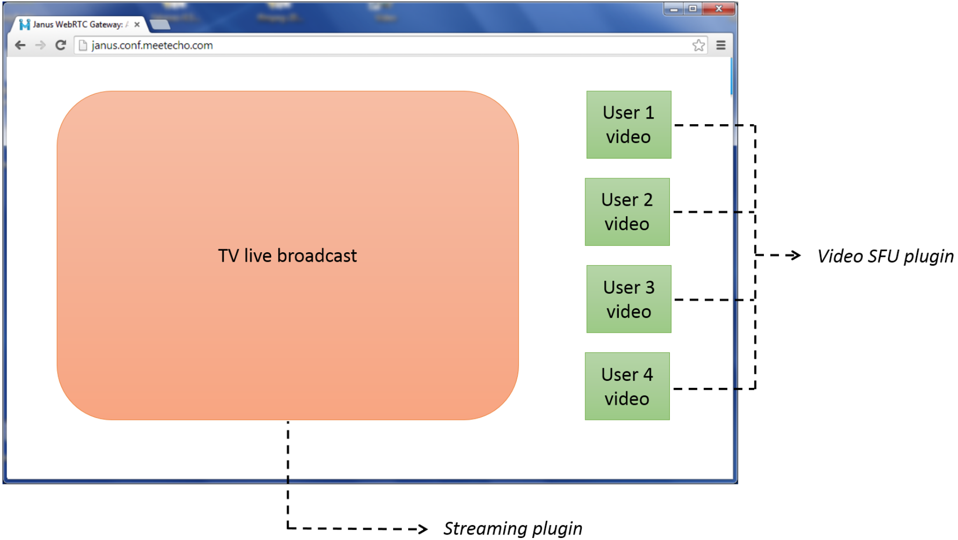
Webinar with Q/A



Social TV


Janus WebRTC Gateway:
janus.conf.meetecho.com
TV live broadcast
User 1 video
User 2 video
User 3 video
User 4 video
Video SFU plugin
Streaming plugin

Anything wrong? Check the Admin API!

- Requests/response API to poll Janus
  - Query server capabilities
  - Control some aspects (e.g., enable/disable debugging)
  - Inspect handles and WebRTC "internals"

http://www.meetecho.com/blog/understanding-the-janus-admin-api/

Yeah, yeah, but what about SIP?

- As anticipated, SIP already available as a Janus plugin
  - Demo: https://janus.conf.meetecho.com/siptest
- Basically a WebRTC-to-SIP gateway
  - WebRTC on one side, SIP(S)/(S)RTP on the other end
- Janus SIP plugin acts as a SIP endpoint
  - SIP stack implemented with Sofia-SIP
  - WebRTC users only see the Janus API (JSON)
  - No transcoding, media is only relayed
- Simplifies life for web developers
  - No need to worry about a SIP stack (only SIP URIs)
  - Basic methods/events to handle call (call, answer, hangup)
  - Allows headers injection, for ad-hoc cases

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But what if you DON'T want it simple?

- You may want to have more control on SIP messaging
  - e.g., to re-use stacks like JsSIP or SIP.js, or other reasons
- The existing SIP plugin doesn't allow for that
  - Complexity hidden from users, on purpose
  - Only partial control (e.g., custom headers, INFO DTMF, negotiating security)
- **BUT!** Janus is extensible, so why not a new plugin?
- @saghul's idea: "BoringSDP"!
  - A new plugin to only handle media gatewaying
    - WebRTC and SIP SDPs both available to web user
  - You handle SIP transactions yourself, and leave media to Janus
    - You still need to communicate with Janus as well, of course

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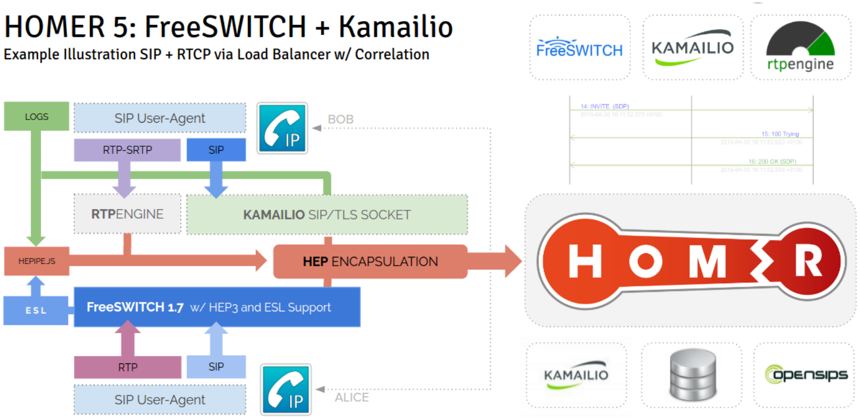
Another idea: Homer/HEP support!

It may be cool to monitor Janus SIP plugin as part of a SIP infrastructure

- Homer/HEP would allow for that
- Mirroring would require integration in SIP plugin and Janus core



What is Janus used for today, and by whom?

- We use it ourselves for many things (obviously)
  - Web conferencing and Webinars
  - WebRTC-to-SIP gateway
  - Streaming of live events (e.g., IETF meetings)
- Many folks/companies also using it in creative ways!
  - E-learning
  - Coworking
  - Contact centers
  - TV broadcasting and Social TV
  - Surveillance systems
  - Home automation & Internet of Things
  - Mobile devices, Raspberry Pis, drones, etc.
- New third-party tools are starting to come out
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    - New plugins for ad-hoc requirements
    - Server-side API wrappers (node.js, .NET, ...)

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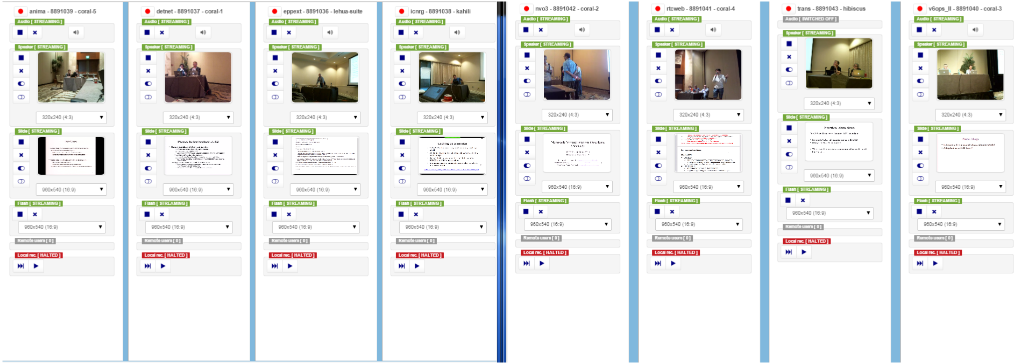
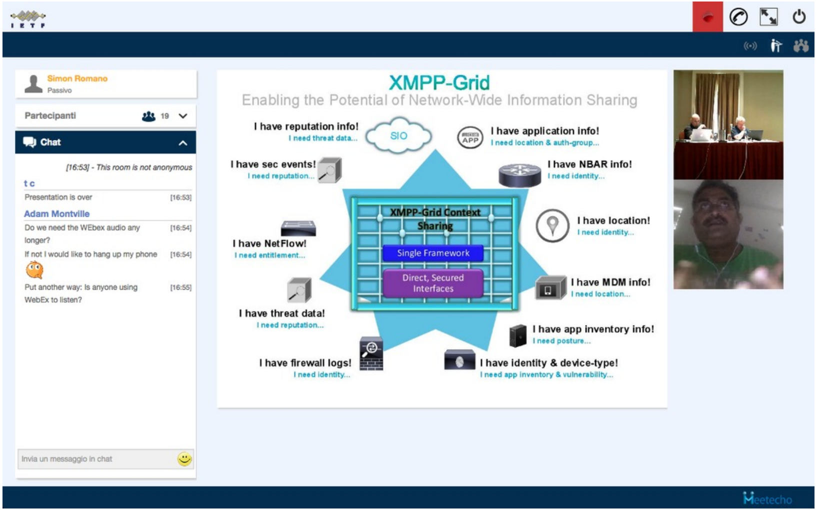
“Director” room @ IETF meetings

Completely WebRTC-based media streams

- Slides as a video feed from the beamer
- Static video feed from the room
- Dynamic video feeds for remote speakers

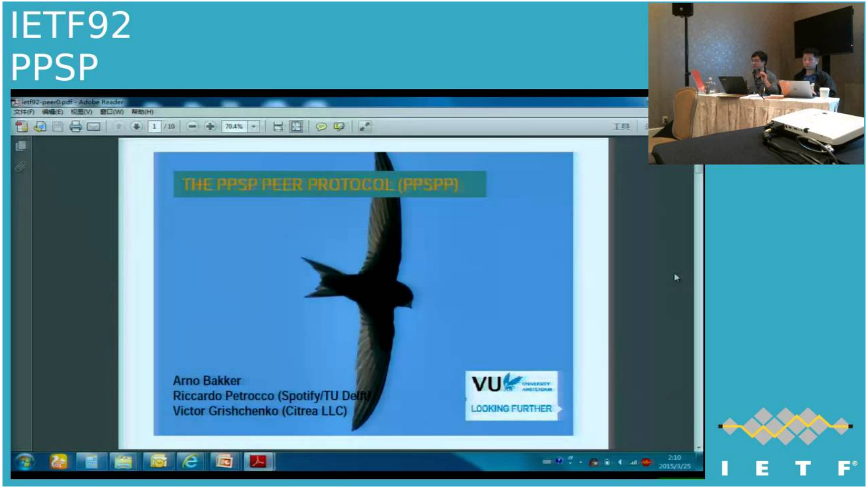


Meetecho: IETF meeting example



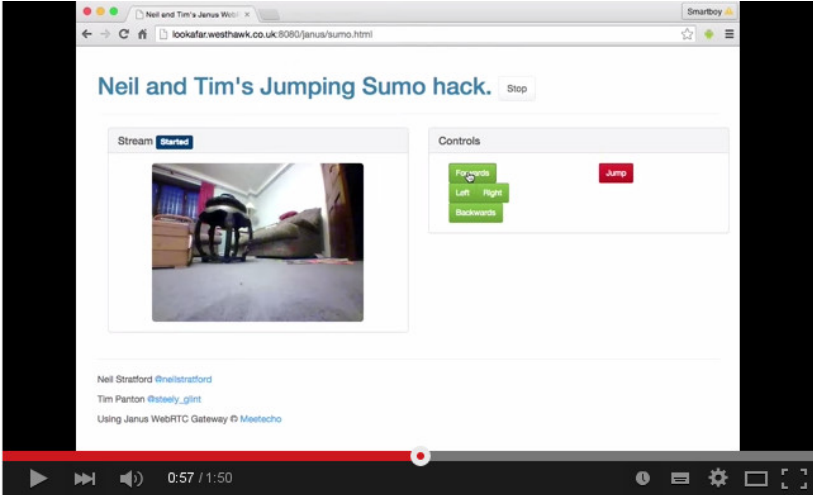
https://ietf.org/meeting/remote-participation.html

Meetecho: IETF recordings



https://www.youtube.com/user/ietf

A "silly" use case: The Jumping Sumo!



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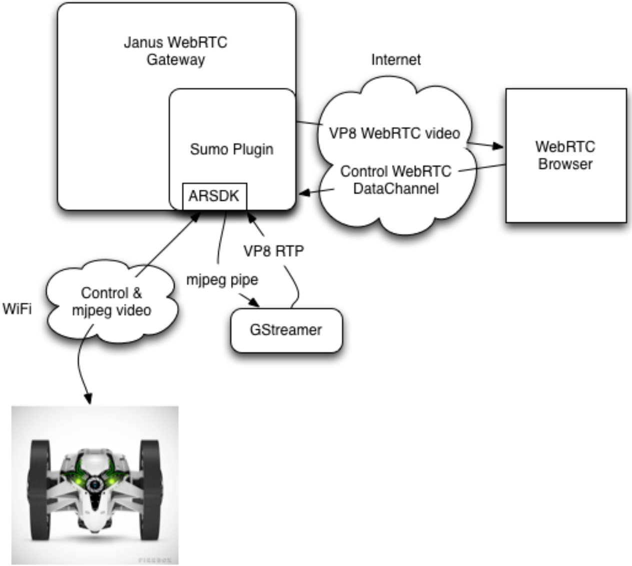

"Matrix wins Best of Show at WebRTC World!"

https://www.youtube.com/watch?v=OMzDklvDS3c

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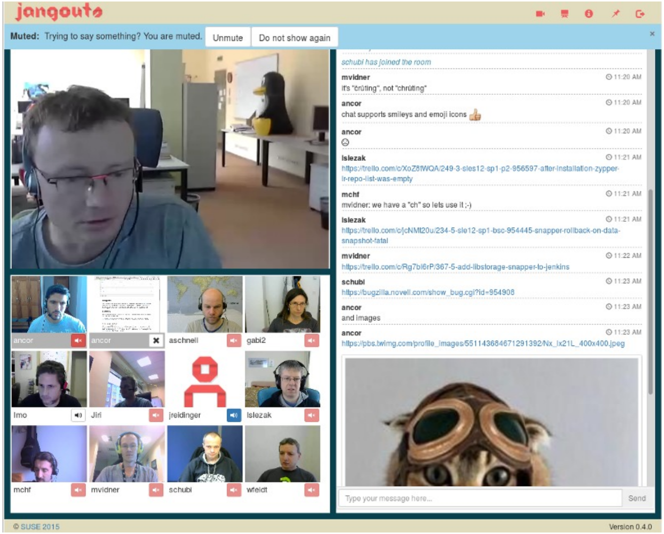
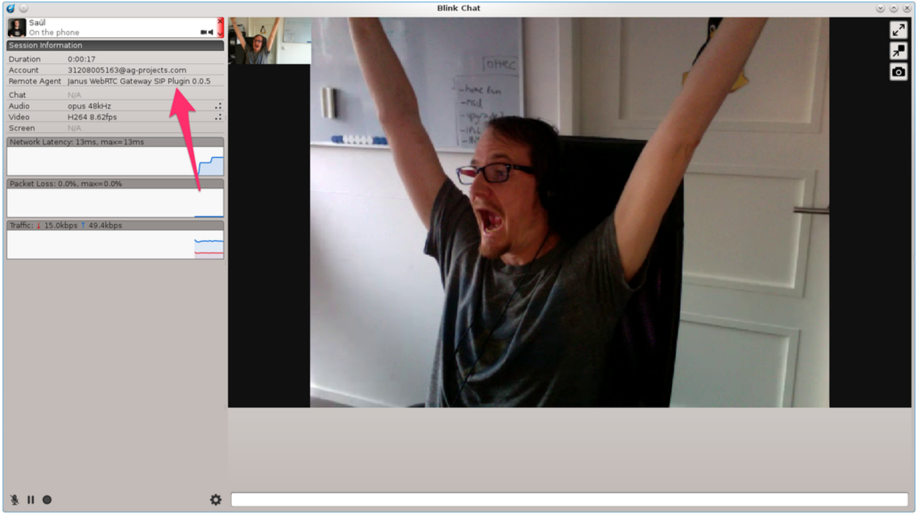
https://www.youtube.com/watch?v=NpBStIIq6fM

Jangouts (for "Janus Hangouts" ☺)



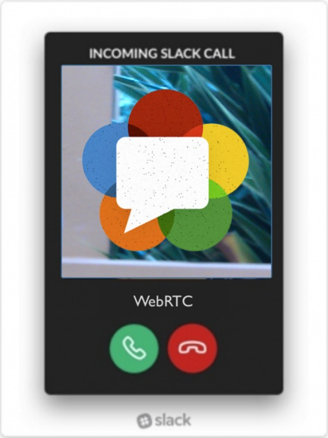
https://github.com/jangouts/jangouts

SylkServer (SIP/XMPP Application Server)



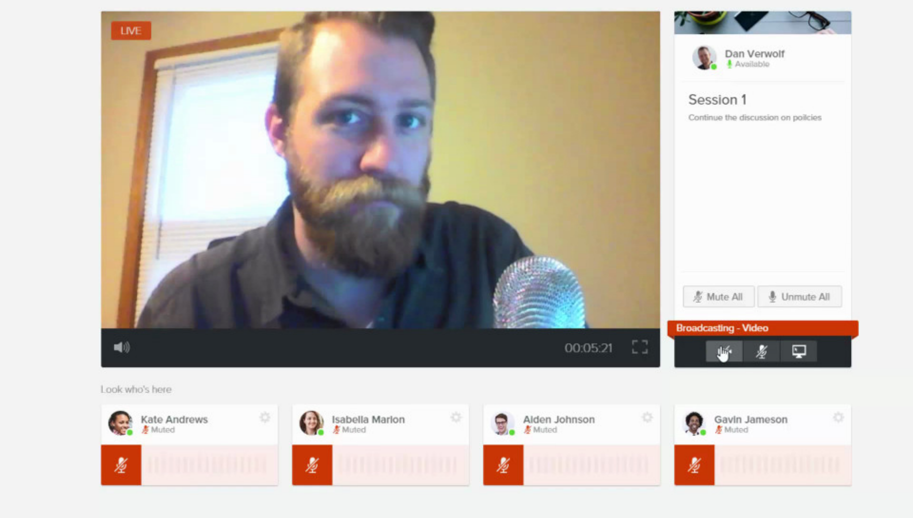
http://sylkserver.com/

Slack? (team co-working)



https://webrtchacks.com/dear-slack/
https://webrtchacks.com/slack-webrtc-slacking/

Lenovo's AirClass (e-learning)



https://www.airclass.com

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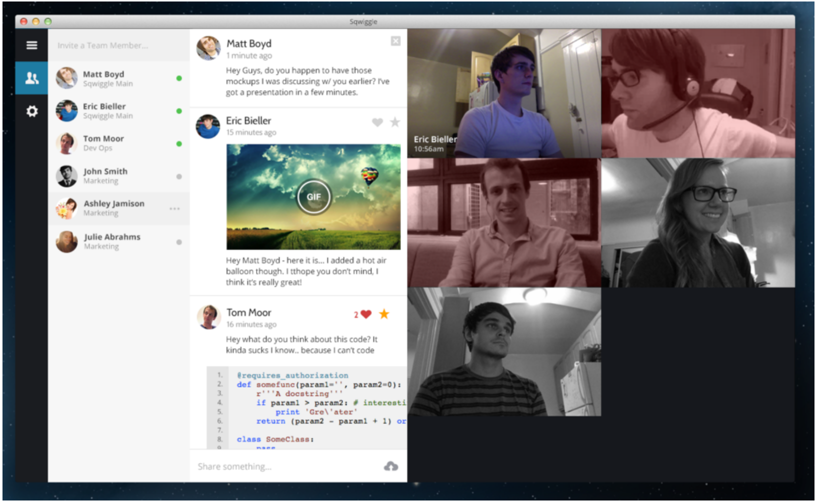

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https://speak.io

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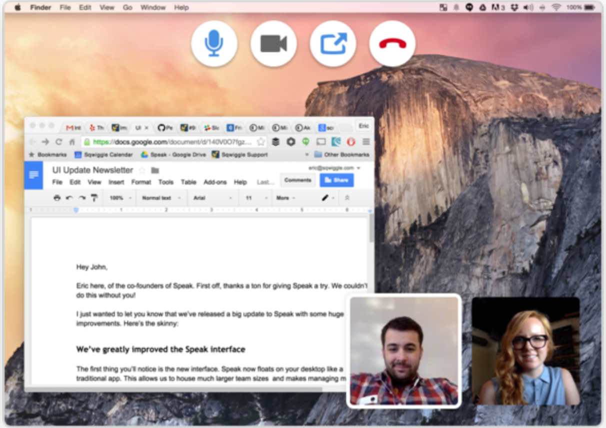

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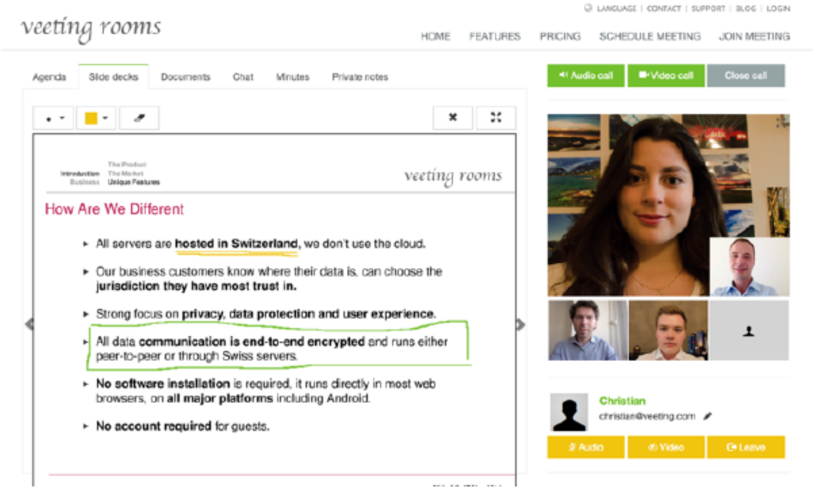
Veeting rooms (web conferencing)



https://www.veeting.com

What to do next?

- Finalize the WebRTC implementation
  - Work on effective renegotiation
  - Implement multistream (Unified Plan)
  - Add octets (besides strings) to DataChannels
  - Keep up-to-date with newest stuff
- Keep on improving and fixing things
  - A datachannel to/from SIP MESSAGE may be cool
  - Implement admin API notifications (subscription)
  - Reference counters (currently in a PR)
  - Why not, some new transport modules (Unix Sockets in PR)
  - Maybe some changes to the pluggable architecture too?
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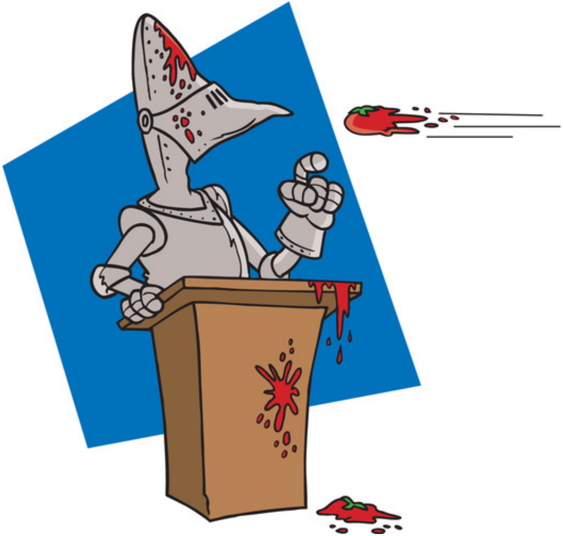
Pitchfest: souvenirs from San Fran ☺

Questions? Comments?



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