



OpenSIPS'16

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Janus

Modules and APIs

What about SIP?


A few examples

Next steps

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

Lorenzo Miniero lorenzo@meetecho.com

OpenSIPS Summit 2016

11th May 2016, 



Outline

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- 3 Writing a WebRTC gateway from scratch
Programmable Real-time Media Components
- 4 Janus: a general purpose WebRTC gateway
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- 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...)

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

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- 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



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

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WebRTC reference architecture

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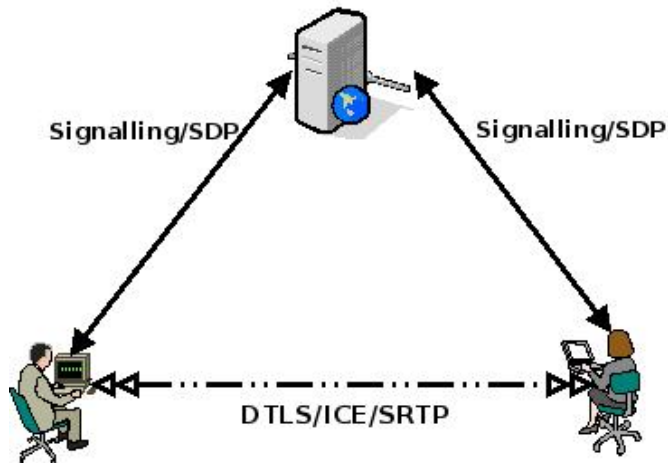
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Involving a gateway (and applications)

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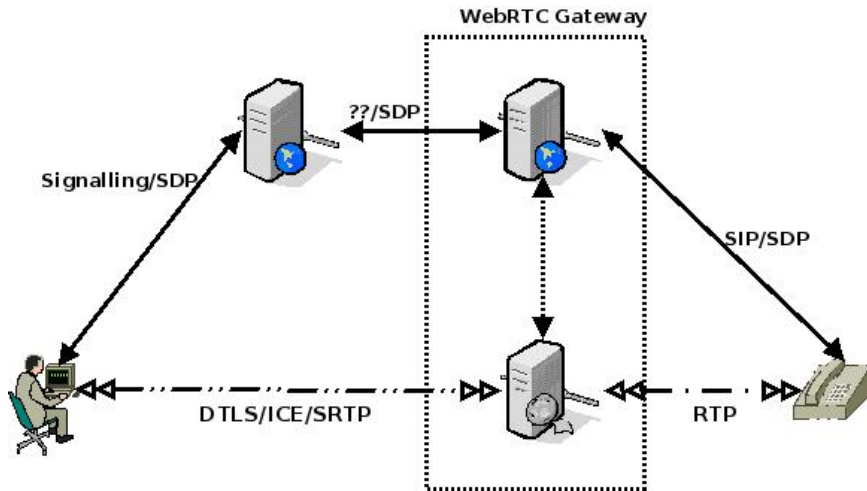
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Involving different technologies as well

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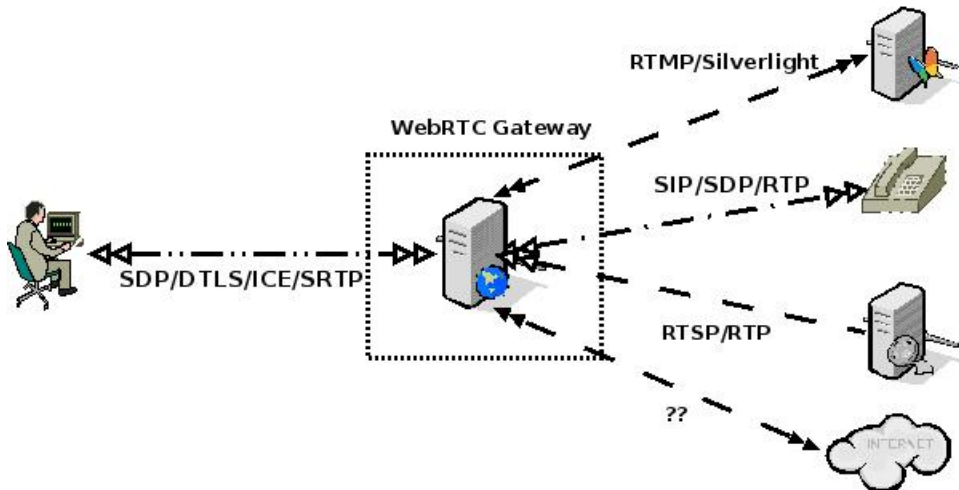
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Do we really need a gateway?

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- 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://webrtcchacks.com/webrtc-gw/>



Do we really need a gateway?

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Real-time Media Components

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- 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?



Real-time Media Components

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The WebRTC protocol suite

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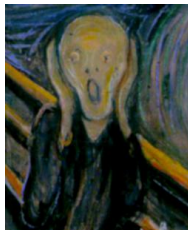
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What about SIP?

A few examples

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- 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

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“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

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

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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

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- 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 (→ Live events!)
 - Video Room (→ Selective Forwarding Unit!)
 - SIP Gateway (→ “Legacy” SIP!)
 - ...



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Extensible Architecture and API

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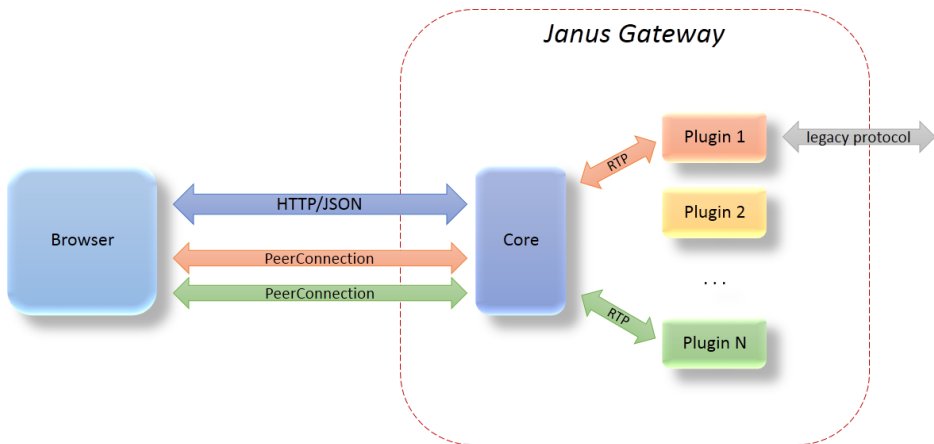
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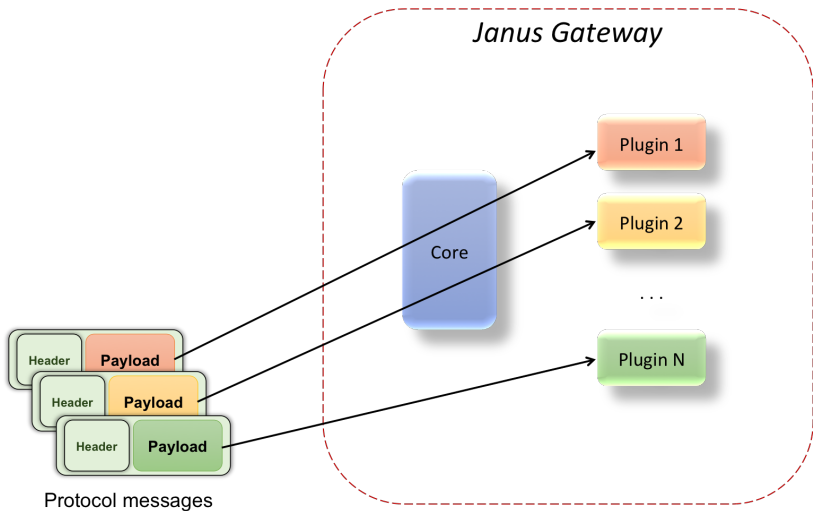
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Plugins as “bricks”

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- 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)



Plugins as “bricks”

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Webinar with Q/A

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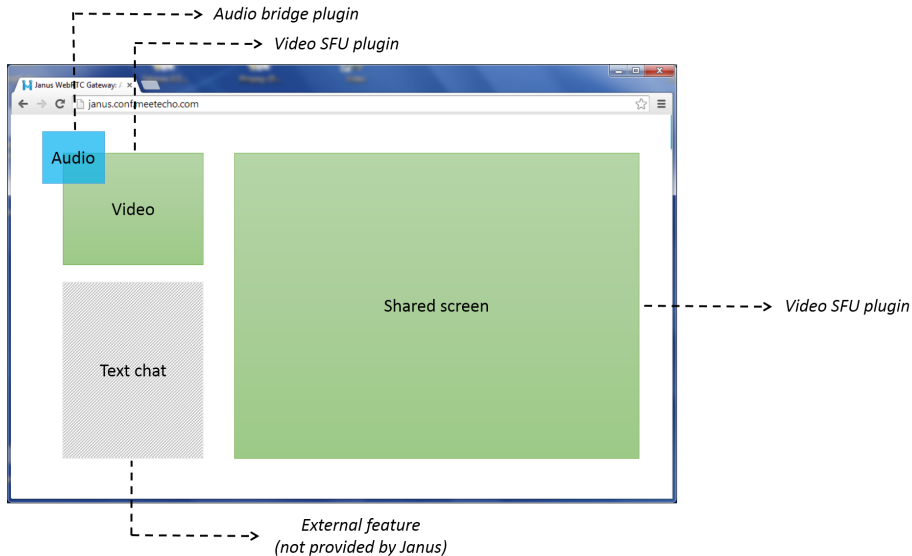
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Social TV

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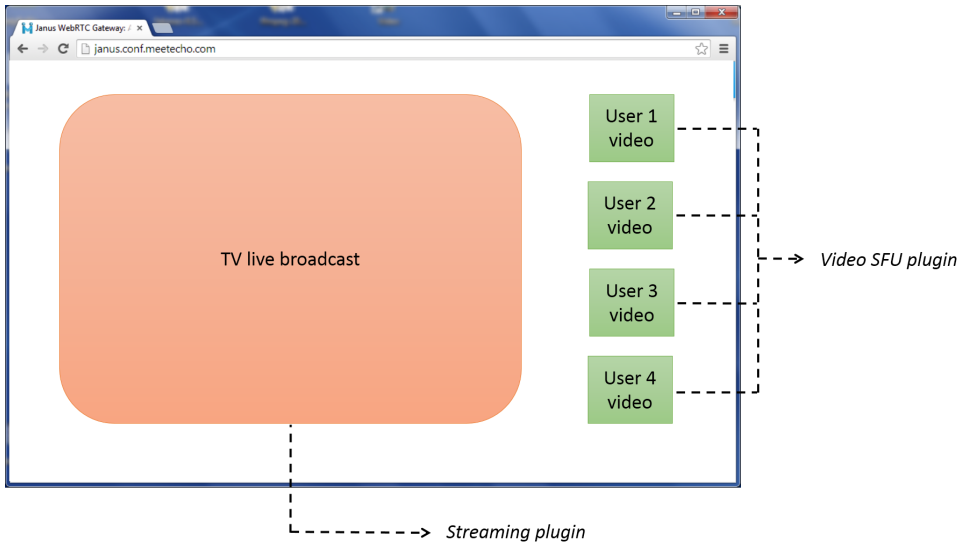
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Anything wrong? Check the Admin API!

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

Sessions (1) ↻

1489448365

Handles (1) ↻

783422373

Handle Info ↻

```
{
  "session_id": 1489448365,
  "handle_id": 783422373,
  "plugin": "janus.plugin.echotest",
  "plugin_specific": {
    "audio_active": "true",
    "video_active": "true",
    "bitrate": 0,
    "slowlink_count": 0,
    "destroyed": 0
  },
  "flags": {
    "processing_offer": 0
  }
}
```

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



Yeah, yeah, but what about SIP?

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- 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?

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- 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



But what if you DON'T want it simple?

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Another idea: Homer/HEP support!

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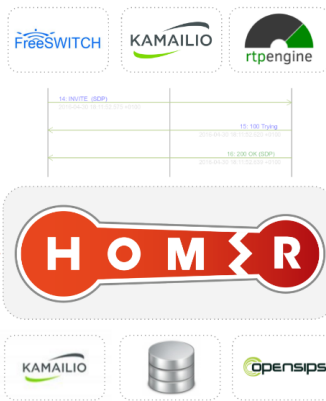
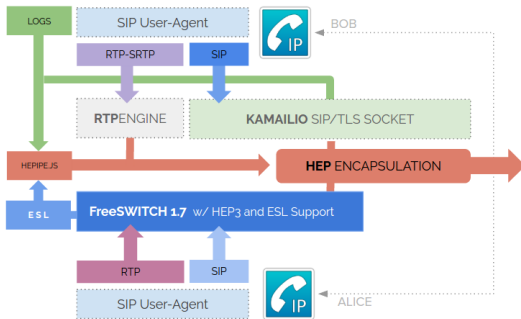
Next steps

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

HOMER 5: FreeSWITCH + Kamailio

Example Illustration SIP + RTCP via Load Balancer w/ Correlation





What is Janus used for today, and by whom?

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- 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
 - <https://janus.conf.meetecho.com/docs/resources>
 - New plugins for ad-hoc requirements
 - Server-side API wrappers (node.js, .NET, ...)



What is Janus used for today, and by whom?

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- We use it ourselves for many things (obviously)
 - Web conferencing and Webinars
 - WebRTC-to-SIP gateway
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“Director” room @ IETF meetings

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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

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The screenshot displays a Meetecho interface for an IETF meeting. On the left is a chat window for the session 'Simon Romano Passivo' with 19 participants. The chat history shows a message from Adam Montville at 16:53 stating 'This room is not anonymous', followed by a discussion about WebEx audio and WebEx listening at 16:54 and 16:55. The main area features a presentation slide titled 'XMPP-Grid: Enabling the Potential of Network-Wide Information Sharing'. The slide contains a central diagram of a grid labeled 'XMPP-Grid Context Sharing' with 'Single Framework' and 'Direct, Secured Interfaces'. Surrounding the grid are various information types and their needs: 'I have reputation info! I need threat data...' (SIO), 'I have application info! I need location & auth-group...' (APP), 'I have NBAR info! I need identity...' (NBAR), 'I have location! I need identity...' (Location), 'I have MDM info! I need location...' (MDM), 'I have app inventory info! I need posture...' (App Inventory), 'I have identity & device-type! I need app inventory & vulnerability...' (Identity), 'I have firewall logs! I need identity...' (Firewall), 'I have threat data! I need reputation...' (Threat Data), 'I have NetFlow! I need entitlement...' (NetFlow), and 'I have sec events! I need reputation...' (Sec Events). On the right, there are two video feeds: the top one shows two people at a table, and the bottom one is a close-up of a man with glasses. The Meetecho logo is in the bottom right corner of the interface.

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



Meetecho: IETF recordings

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**IETF92
PPSP**

THE PPSP PEER PROTOCOL (PPSP)

Arno Bakker
Riccardo Petrocco (Spotify/TU Delft)
Victor Grishchenko (Citrea LLC)

VU
AMSTERDAM
UNIVERSITY
LOOKING FURTHER

I E T F®

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



A “silly” use case: The Jumping Sumo!

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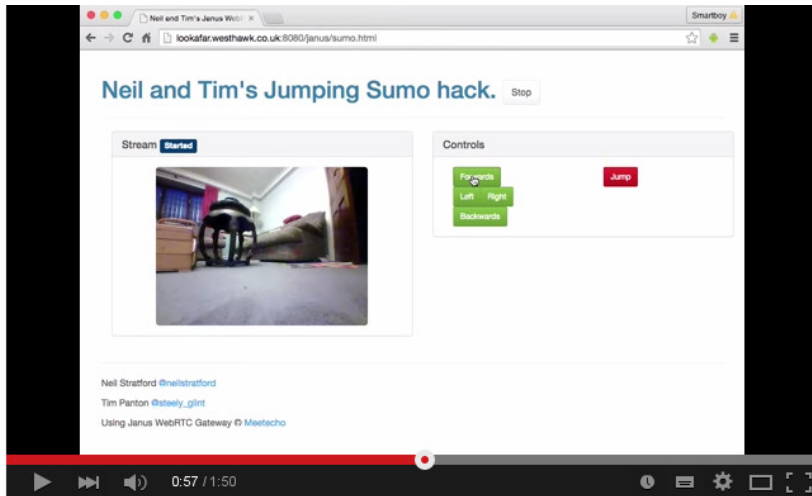
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<https://www.youtube.com/watch?v=isGSnMIKcss>



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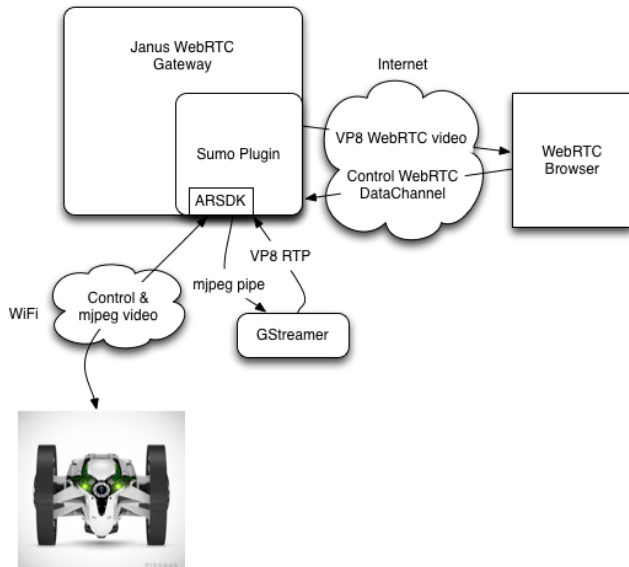
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“Matrix wins Best of Show at WebRTC World!”

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



“Matrix wins Best of Show at WebRTC World!”

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



Jangouts (for "Janus Hangouts" 😊)

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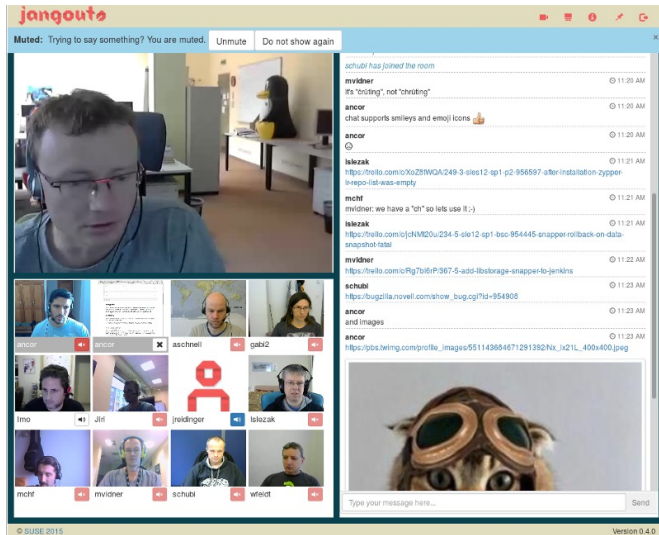
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<https://github.com/jangouts/jangouts>



SylkServer (SIP/XMPP Application Server)

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Saul
On the phone

Session Information

Duration	0:00:17
Account	31208005163@ag-projects.com
Remote Agent	Janus WebRTC Gateway SIP Plugin 0.0.5
Chat	N/A
Audio	opus 48kHz
Video	H264 8.62fps
Screen	N/A

Network Latency: 13ms, max=13ms

Packet Loss: 0.0%, max=0.0%

Traffic: ↓ 15.0kbps ↑ 49.4kbps

Blink Chat

http://sylkserver.com/

<http://sylkserver.com/>



Slack? (team co-working)

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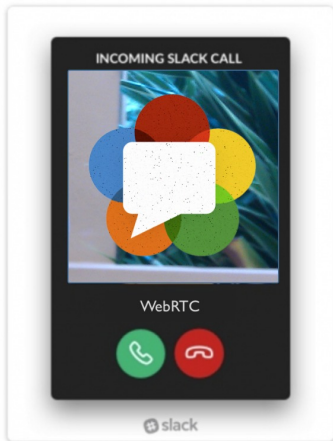
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<https://webrtcchacks.com/dear-slack/>
<https://webrtcchacks.com/slack-webrtc-slacking/>



Lenovo's AirClass (e-learning)

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LIVE

Dan Verwolf
Available

Session 1
Continue the discussion on policies

Mute All Unmute All

Broadcasting - Video

Look who's here

Kate Andrews
Muted

Isabella Marion
Muted

Aiden Johnson
Muted

Gavin Jameson
Muted

00:05:21

<https://www.airclass.com>



Sqwiggle / Speak.io (team co-working)

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<https://www.sqwiggle.com>

<https://speak.io>



Sqwiggle / Speak.io (team co-working)

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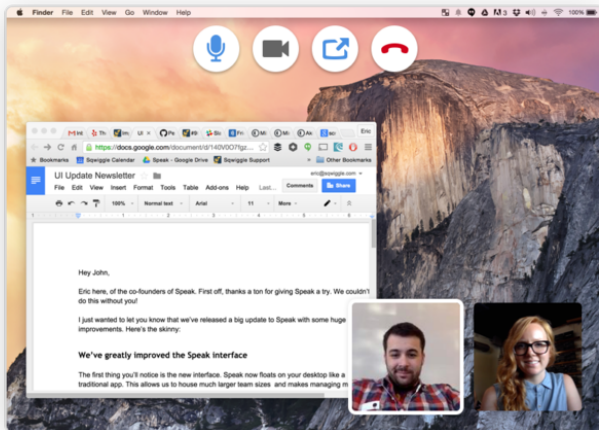
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<https://www.sqwiggle.com>

<https://speak.io>



Veeting rooms (web conferencing)

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The screenshot displays the Veeting rooms web interface. At the top, the logo "veeting rooms" is visible, followed by navigation links: HOME, FEATURES, PRICING, SCHEDULE MEETING, and JOIN MEETING. Below the logo, there are tabs for Agenda, Slide decks (selected), Documents, Chat, Minutes, and Private notes. The main content area shows a presentation slide titled "How Are We Different" with the following bullet points:

- ▶ All servers are **hosted in Switzerland**, we don't use the cloud.
- ▶ Our business customers know where their data is, can choose the **jurisdiction they have most trust in**.
- ▶ Strong focus on **privacy, data protection and user experience**.
- ▶ All data **communication is end-to-end encrypted** and runs either peer-to-peer or through Swiss servers.
- ▶ No **software installation** is required, it runs directly in most web browsers, on **all major platforms** including Android.
- ▶ No **account** required for guests.

On the right side of the interface, there are buttons for "Audio call", "Video call", and "Close call". Below these is a video call grid showing several participants. At the bottom right, there is a profile card for "Christian" with the email "christian@veeting.com" and buttons for "Audio", "Video", and "Leave".

<https://www.veeting.com>



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- 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?
- **Help us improve this!**
 - Play with it, more testing is important
 - Write your own applications/wrappers/plugins!



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Last month's events

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- IETF 95
 - April 3-8, Buenos Aires (Argentina)
 - More than 140 interactive (WebRTC-enabled) sessions in a week
 - ... and without being bashed! (too much ☺)
- IEEE INFOCOM 2016
 - April 10-15, San Francisco (USA)
 - Innovation Challenge Panel/Pitchfest
 - ... and guess what? we won!!
 - (despite the fact I spilled a glass of water on the slides laptop...)
- **Next:** Berlin (twice!)
 - Kamailio World 2016, May 18-20
 - IETF96, July 17-22



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Pitchfest: souvenirs from San Fran ☺

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Questions? Comments?

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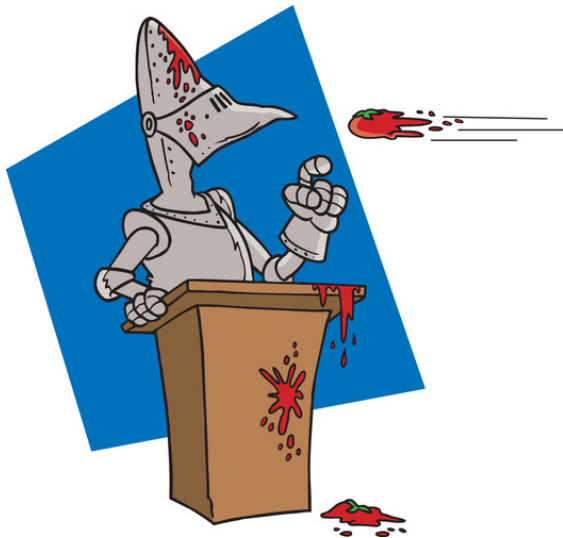
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<https://twitter.com/elminiero>