



KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?


A few examples

Next steps

A Tale Of Two Worlds: Bridging SIP And WebRTC With Janus

Lorenzo Miniero

 @elminiero

Kamailio World
19th May 2016, 



Outline

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

① A brief introduction

② Some context

WebRTC and standardization activities

③ Janus: a general purpose WebRTC gateway

Modular architecture

A few words on Janus and SIP

What is Janus used for today, and by whom?

④ Next steps



What's Meetecho?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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, ...)



What's Meetecho?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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What's Meetecho?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps





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

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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WebRTC reference architecture

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Intro

WebRTC

Standardization

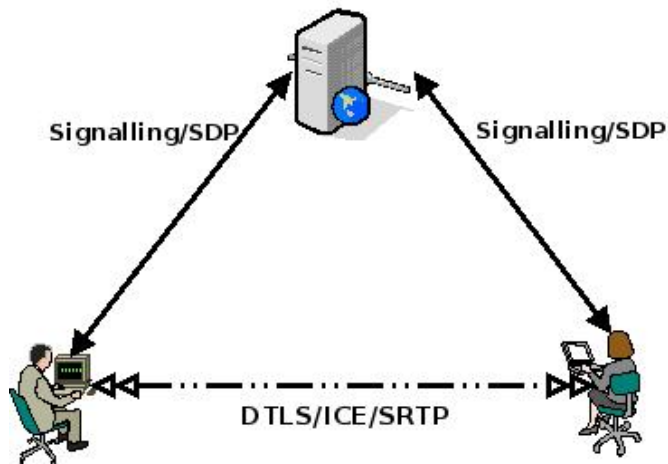
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Modules and APIs

What about SIP?

A few examples

Next steps





Involving a gateway (and applications)

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Intro

WebRTC

Standardization

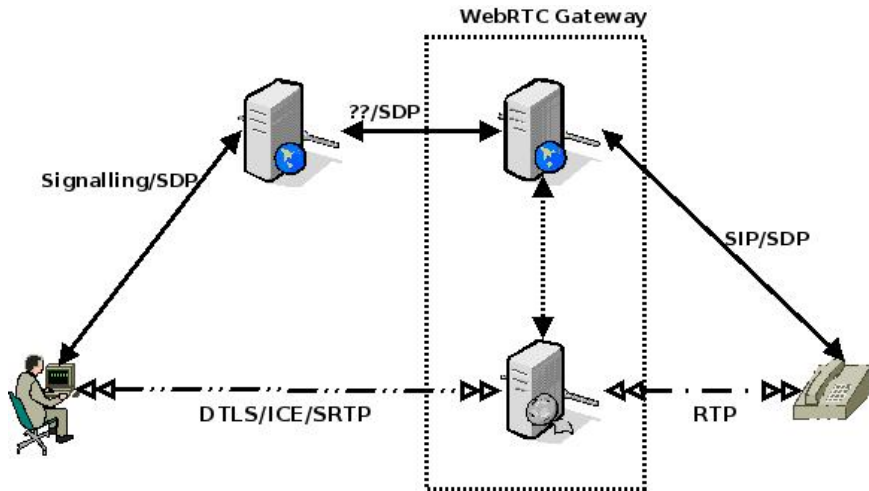
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Modules and APIs

What about SIP?

A few examples

Next steps





Involving different technologies as well

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Intro

WebRTC

Standardization

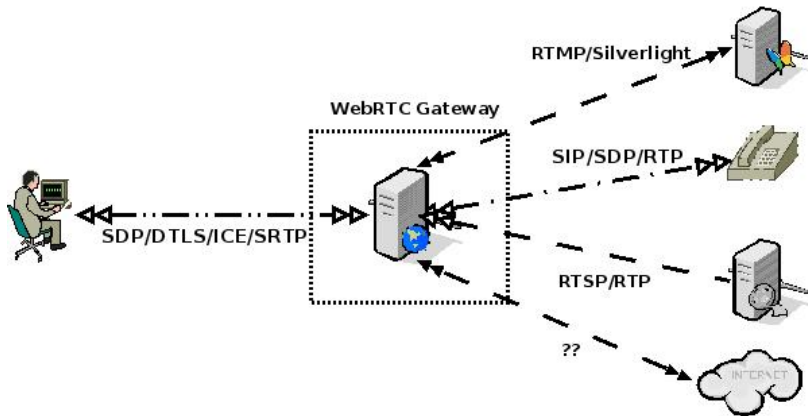
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Modules and APIs

What about SIP?

A few examples

Next steps



“What is a WebRTC Gateway anyway?”

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



Involving different technologies as well

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Intro

WebRTC

Standardization

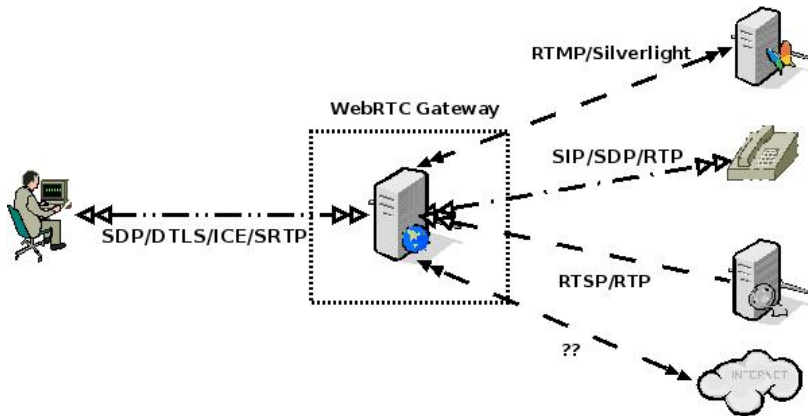
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Modules and APIs

What about SIP?

A few examples

Next steps



“What is a WebRTC Gateway anyway?”

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Janus: a general purpose WebRTC gateway

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L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps



“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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L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- The core only implements the WebRTC stack
 - JSEP/SDP, ICE, DTLS-SRTP, Data Channels, ...
- Plugins expose Janus API over different transports
 - Currently HTTP / WebSockets / RabbitMQ / Unix Sockets
- “Application” logic implemented in plugins too
 - 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 (→ Plain RTP to WebRTC!)
 - Video Room (→ Selective Forwarding Unit!)
 - SIP Gateway (→ “Legacy” SIP!)
 - ...



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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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Modular architecture

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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Modular architecture

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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Extensible Architecture and API

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Intro

WebRTC

Standardization

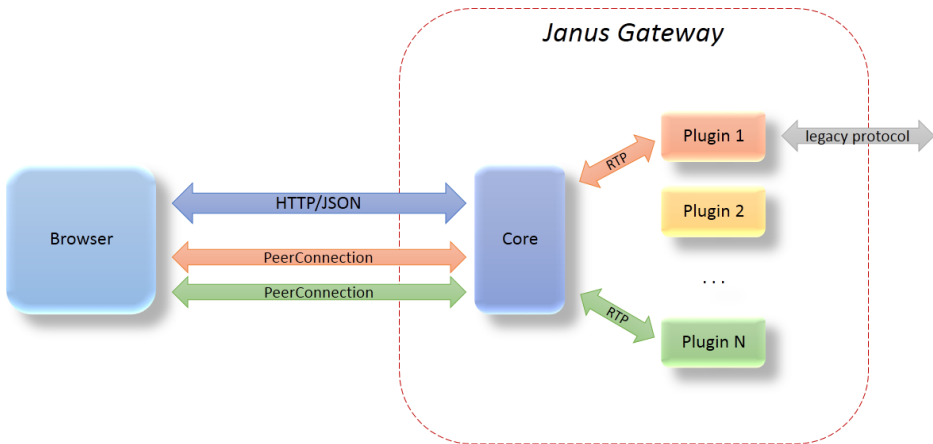
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Modules and APIs

What about SIP?

A few examples

Next steps





Extensible Architecture and API

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Intro

WebRTC

Standardization

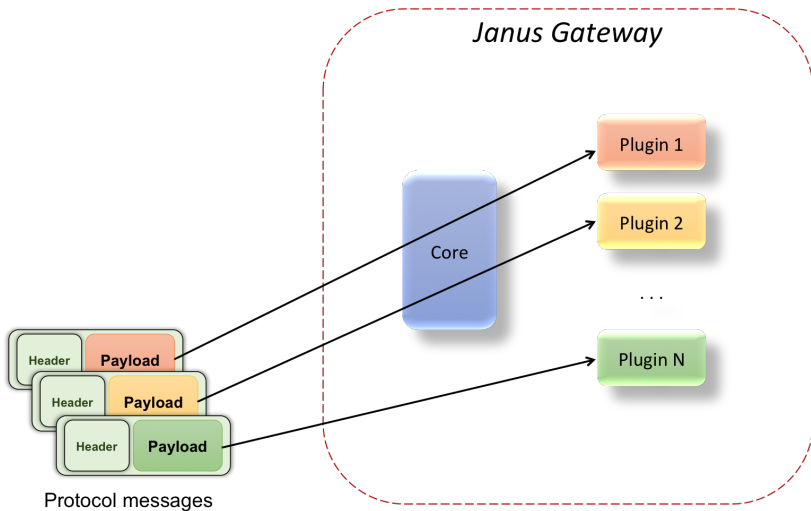
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Modules and APIs

What about SIP?

A few examples

Next steps





Plugins as “bricks”

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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Webinar with Q/A

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Intro

WebRTC

Standardization

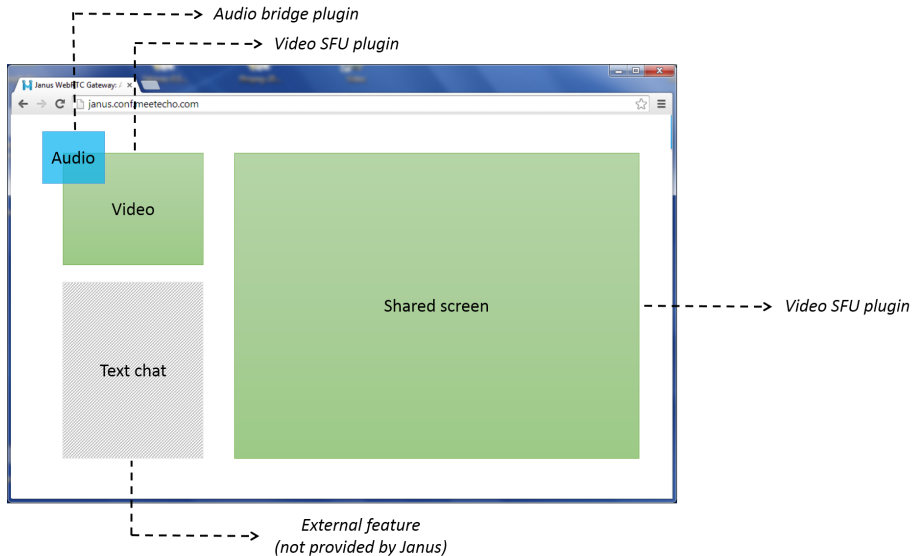
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Modules and APIs

What about SIP?

A few examples

Next steps





Social TV

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Intro

WebRTC

Standardization

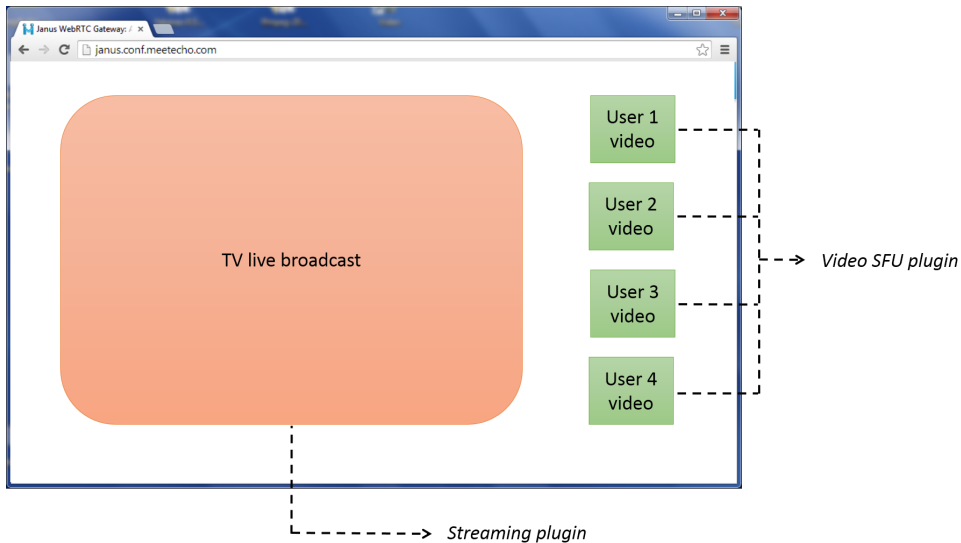
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Modules and APIs

What about SIP?

A few examples

Next steps





Anything wrong? Check the Admin API!

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- Requests/response API to interrogate Janus
 - Query server capabilities
 - Control some aspects (e.g., enable/disable debugging)
 - Inspect handles and WebRTC “internals”
- What about asynchronous events? More on that later!

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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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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But what if you DON'T want it simple?

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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What is Janus used for today, and by whom?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- 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
 - E-health
 - 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>



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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

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“Director” room @ IETF meetings

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

The screenshot shows a Meetecho interface for an IETF meeting. On the left, a sidebar contains a chat window with messages from Simon Romano and Adam Montville, and a participant list showing 19 people. The central video area displays a presentation slide titled "XMPP-Grid: Enabling the Potential of Network-Wide Information Sharing". The slide features a central diagram of a grid labeled "XMPP-Grid Context" with "Single Framework" and "Direct, Secured Interfaces". Surrounding the grid are various icons and text boxes representing different types of information sharing, such as "I have reputation info!", "I have application info!", "I have NetFlow!", "I have threat data!", "I have firewall logs!", "I have identity & device-type!", "I have MDM info!", "I have app inventory info!", "I have location!", "I have NBAR info!", and "I have sec events!". On the right, a sidebar shows two video feeds of participants.

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



Meetecho: IETF recordings

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

**IETF92
PPSP**

THE PPSP PEER PROTOCOL (PPSPP)

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

VU
Vrije Universiteit Amsterdam
LOOKING FURTHER

I E T F®

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



A “silly” use case: The Jumping Sumo!

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Intro

WebRTC

Standardization

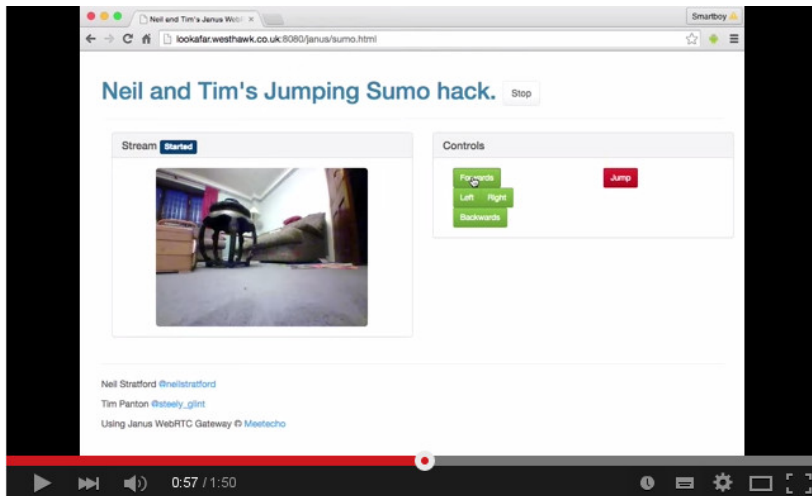
Janus

Modules and APIs

What about SIP?

A few examples

Next steps



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



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Intro

WebRTC

Standardization

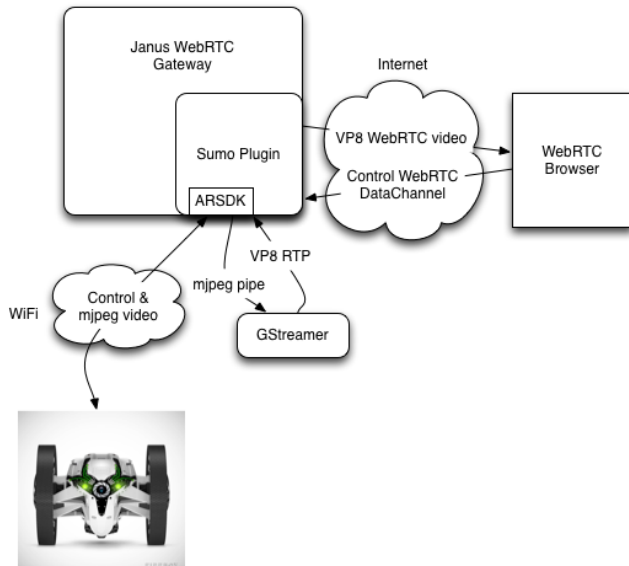
Janus

Modules and APIs

What about SIP?

A few examples

Next steps





“Matrix wins Best of Show at WebRTC World!”

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps



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



“Matrix wins Best of Show at WebRTC World!”

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps



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



Jangouts (for "Janus Hangouts" 😊)

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Intro

WebRTC

Standardization

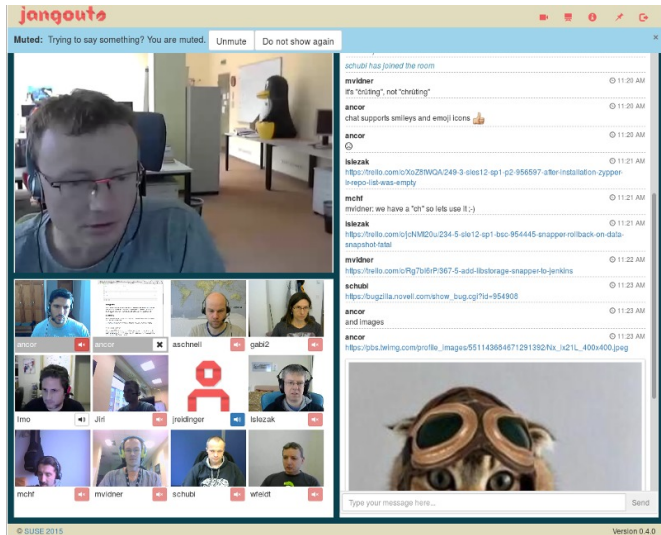
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Modules and APIs

What about SIP?

A few examples

Next steps



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



SylkServer (SIP/XMPP Application Server)

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Intro

WebRTC

Standardization

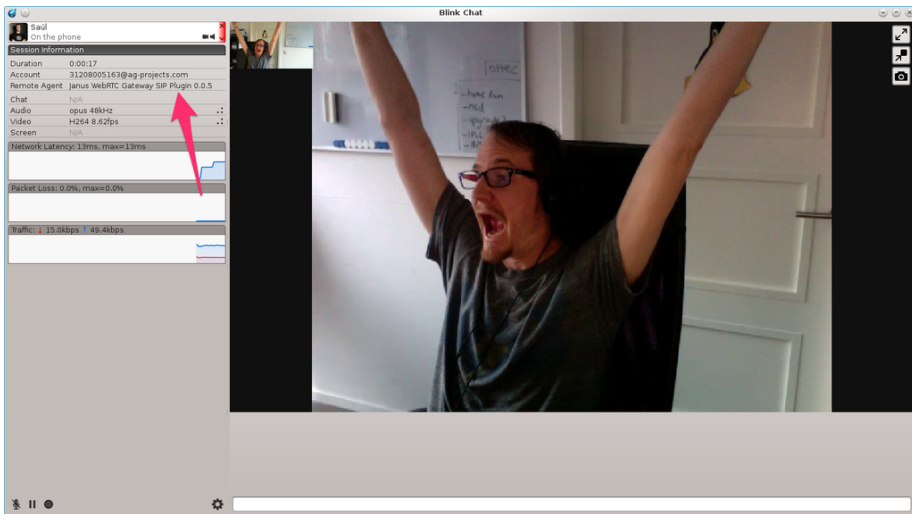
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Modules and APIs

What about SIP?

A few examples

Next steps



<http://sylkserver.com/>



Slack? (team co-working)

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Intro

WebRTC

Standardization

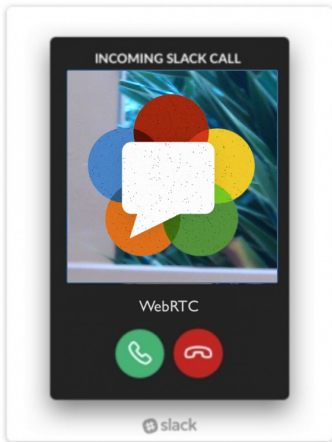
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Modules and APIs

What about SIP?

A few examples

Next steps



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



Lenovo's AirClass (e-learning)

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

The screenshot displays the Squiggle web application interface. On the left, a sidebar lists team members: Matt Boyd (Sqwiggle Main), Eric Bieller (Sqwiggle Main), Tom Moor (Dev Ops), John Smith (Marketing), Ashley Jamison (Marketing), and Julie Abrahms (Marketing). The main chat area shows messages from Matt Boyd, Eric Bieller, and Tom Moor. Eric Bieller's message includes a GIF of a hot air balloon. Tom Moor's message includes a code snippet for a Python function and a class definition. On the right, a video conference grid shows participants: Eric Bieller (top left), a woman (top right), a man (middle left), a woman (middle right), and a man (bottom left). The bottom of the interface has a 'Share something...' input field.

<https://www.sqwiggle.com>

<https://speak.io>



Sqwiggle / Speak.io (team co-working)

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Intro

WebRTC

Standardization

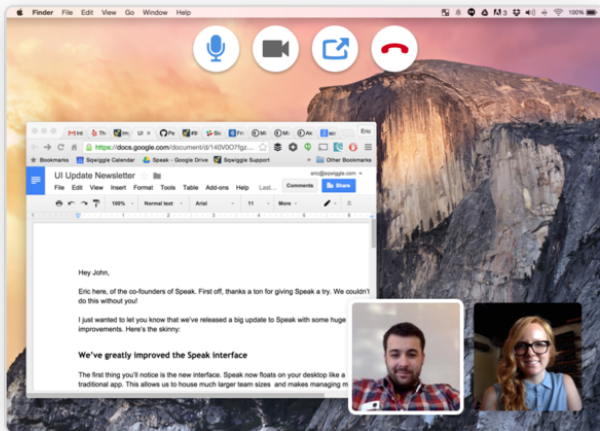
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Modules and APIs

What about SIP?

A few examples

Next steps



<https://www.sqwiggle.com>
<https://speak.io>



Veeting rooms (web conferencing)

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

veeting rooms

HOME FEATURES PRICING SCHEDULE MEETING JOIN MEETING

Agenda Slide decks Documents Chat Minutes Private notes

Audio call Video call Close call

Introduction Business The Product The Market Unique Features

How Are We Different

- ▶ 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.

veeting rooms

Christian
christian@veeting.com

Audio Video Leave

<https://www.veeting.com>



What to do next?

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

- Finalize the WebRTC implementation
 - Renegotiation, multistream, ...
- Keep on improving and fixing things
 - Code cleanup, reference counters, new modules, ...
- A recent idea: modular Events API!
 - Core and plugins generate events
 - Custom modules can subscribe to and handle them
 - e.g., save to DB, send to external service, etc.
 - A couple of potentially interesting integrations
 - Homer/HEP, for SIP calls and their relation to WebRTC
 - Live stats collection via callstats.io

Help us improve this!

- Play with it, more testing is important
- Write your own applications/wrappers/plugins!



What to do next?

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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What to do next?

KamailioWorld

L. Miniero

Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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KamailioWorld

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps

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Shameless bragging! Souvenirs from San Francisco ☺

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Intro

WebRTC

Standardization

Janus

Modules and APIs

What about SIP?

A few examples

Next steps



Innovation Challenge Panel/Pitchfest @ INFOCOM2016... we won!



Questions? Comments?

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Intro

WebRTC

Standardization

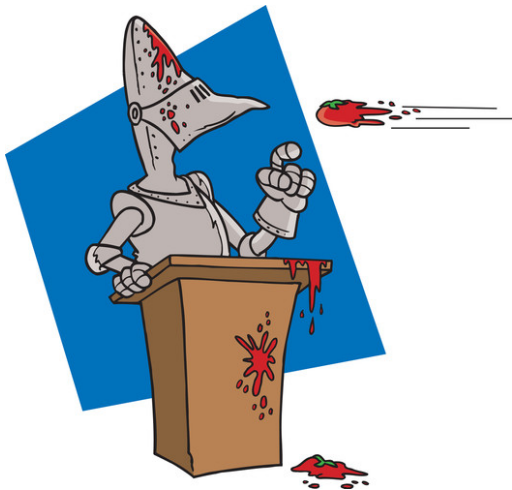
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Modules and APIs

What about SIP?

A few examples

Next steps



@elminiero