WebRTC broadcasting with WHIP

Lorenzo Miniero
@elminiero

FOSDEM 2022 Real Time Communications
5th February 2022, ~~Brussels~~ My couch

Who am I?

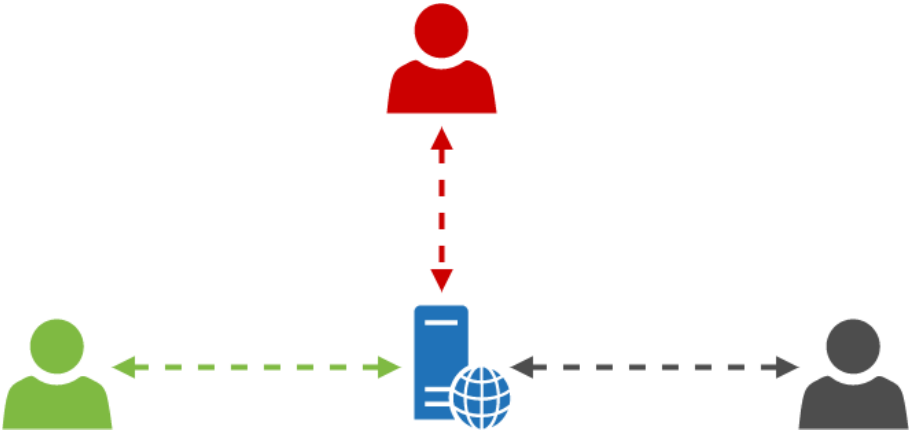
Lorenzo Miniero

- Ph.D @ UniNA
- Chairman @ Meetecho
- Main author of Janus

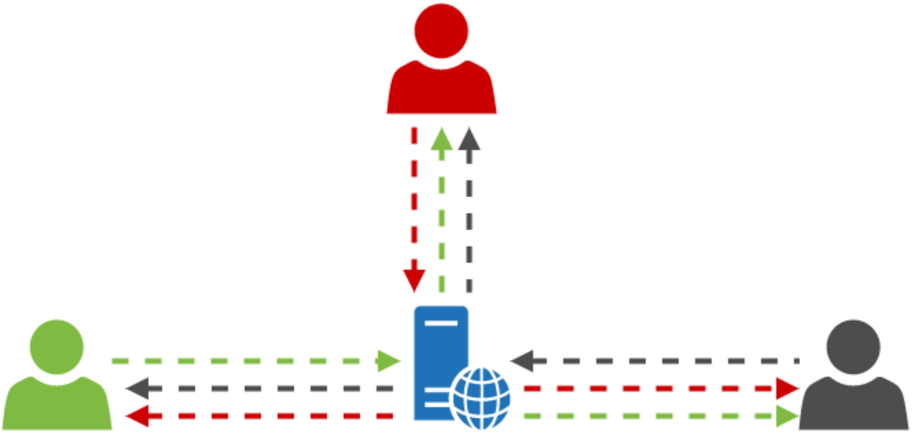
Contacts and info

- lorenzo@meetecho.com
- https://twitter.com/elminiero
- https://www.slideshare.net/LorenzoMiniero
- https://lminiero.bandcamp.com

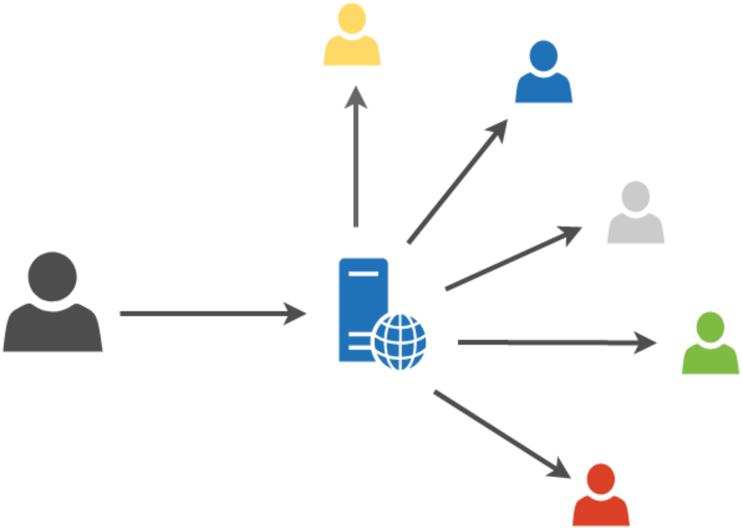
MCU as a WebRTC topology



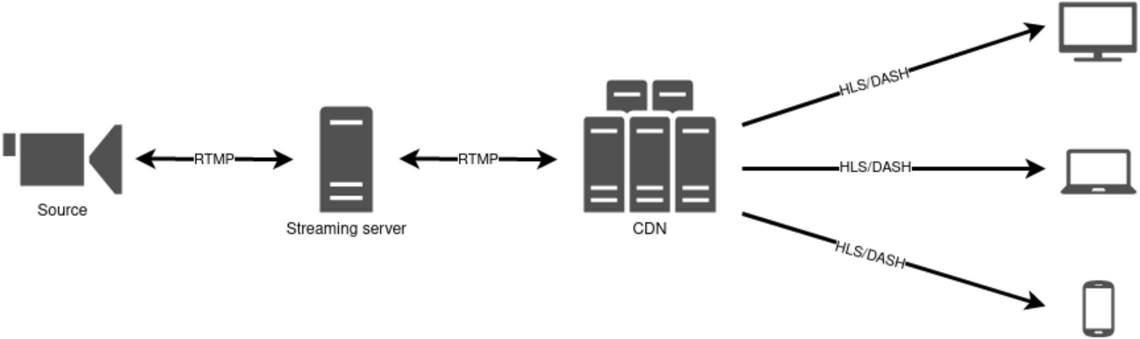
SFU as a WebRTC topology



A slightly variation on the theme



How “traditional” broadcasting typically works



What if we want more interactivity, though?

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Why not WebRTC instead?

- Traditional broadcasting efficient but higher latency
  - At best (live), delay will typically be in the range of a few seconds
  - Besides, different users may experience different delays (buffering)
- WebRTC natively conceived for very low latency, instead
  - Born for conversational audio/video/data
  - Can be (and often is) easily used for monodirectional streaming as well
- Strangely not really considered by the industry up until recently, though
  - Topic of my Ph.D years ago ("Streaming Of Large scale Events over Internet cLouds")
  - Clearing the industry FUD: https://webrtcbydralex.com/index.php/2020/04/14/
- Tooling an important aspect to foster WebRTC adoption, here
  - e.g., a standard way to send media, and tools à la OBS, XSplit, or others

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WebRTC audio: Opus

- WebRTC mandates Opus, and it's a good thing
  - High quality audio codec designed for the Internet
  - Very flexible in sampling rates, bitrates, FEC, etc.
- Different profiles for voice and music/other
  - Both encoding and decoding vary, in case
  - Can be mono and stereo, with dynamic sampling rates and bitrates
- Multiopus (5.1 and 7.1)[1]
  - Each packet is basically OGG with multiple stereo Opus streams
  - Number of streams determines number of channels
  - Not documented, but used by Google for Stadia

[1] https://webrtcbydralex.com/index.php/2020/04/08/surround-sound-5-1-and-7-1-in-libwebrtc-and-chrome/

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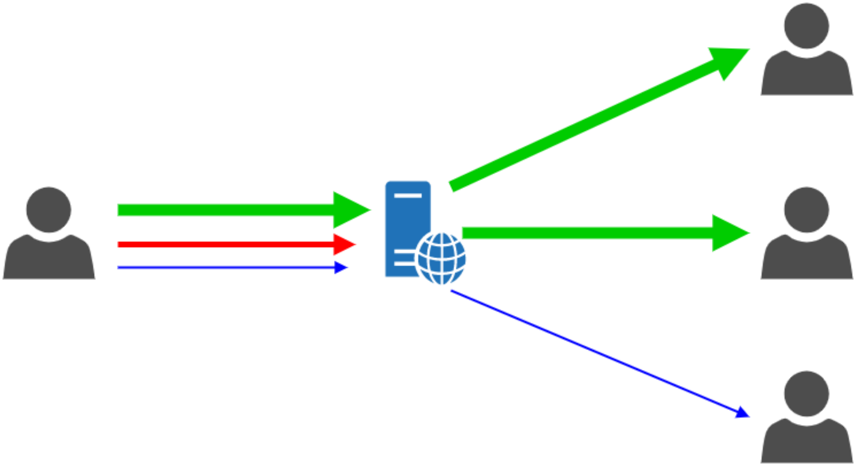
WebRTC video: Simulcasting & SVC



https://webrtchacks.com/sfu-simulcast/

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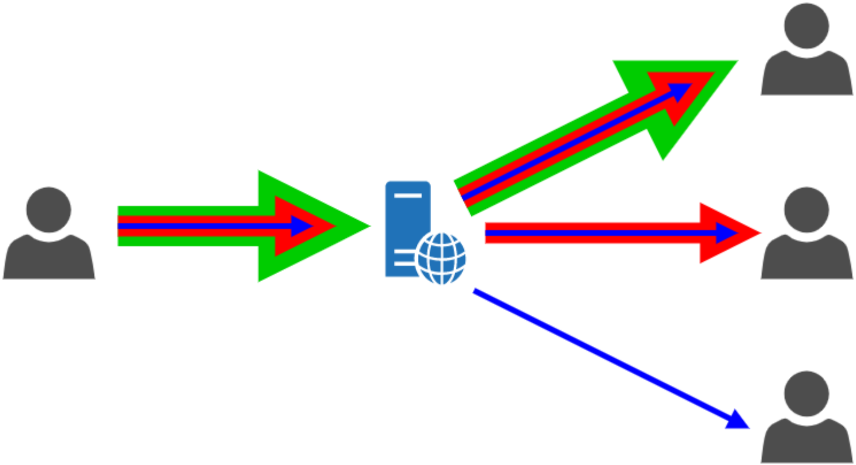
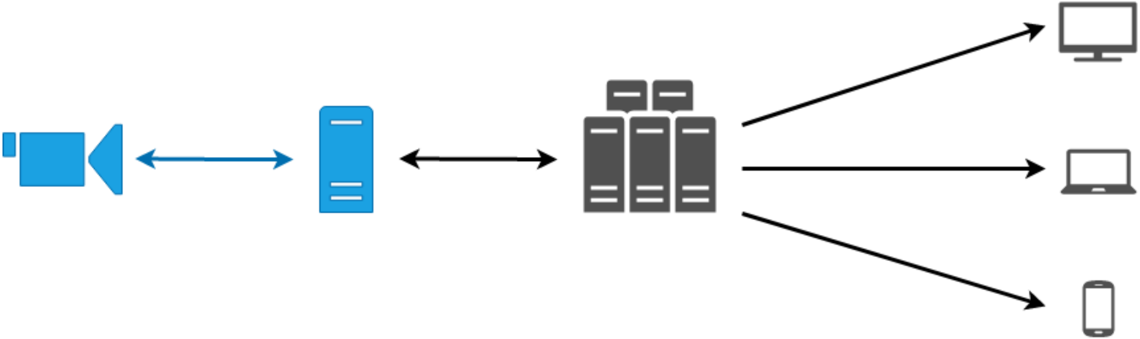
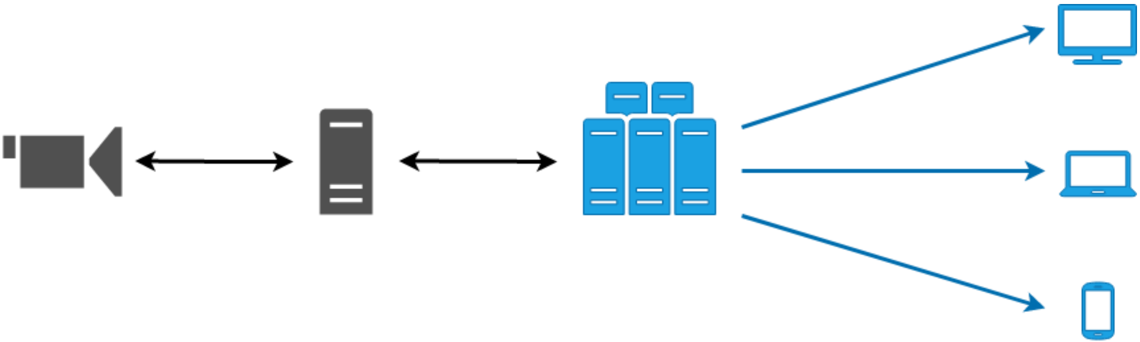


https://webrtchacks.com/chrome-vp9-svc/

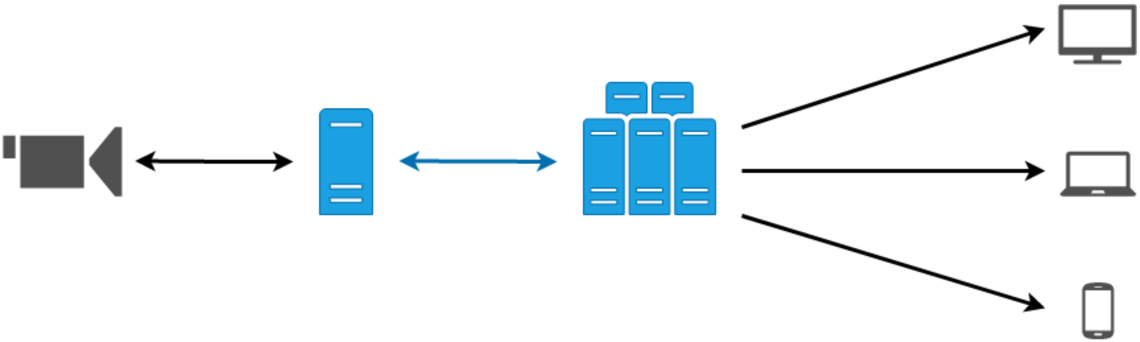
Two main challenges: ingestion...



Two main challenges: ... and distribution



(and yeah, maybe scaling too! 😉)



Making WebRTC ingestion easy: WISH / WHIP!

https://www.meetecho.com/blog/whip-janus/ (September 2020)

Making WebRTC ingestion easy: WISH / WHIP!

https://www.meetecho.com/blog/whip-janus-part-ii/

There would be no WHIP without Dr. Alex ♥

A new Working Group in the IETF...

Datatracker Groups Documents Meetings Other User Document search

WebRTC Ingest Signaling over HTTPS (wish)

About | Documents | Meetings | History | Photos | Email expansions | List archive » | Tools »

WG	**Name**	WebRTC Ingest Signaling over HTTPS
	Acronym	wish
	Area	Applications and Real-Time Area (art)
	State	Active
	Charter	charter-ietf-wish-01 Approved
	Dependencies	Document dependency graph (SVG)
	Additional Resources	- Github
Personnel	**Chairs**	Nils Ohlmeier Sean Turner
	Area Director	Murray Kucherawy
Mailing list	**Address**	wish@ietf.org
	To subscribe	https://www.ietf.org/mailman/listinfo/wish
	Archive	https://mailarchive.ietf.org/arch/browse/wish/
Jabber chat	**Room address**	xmpp:wish@jabber.ietf.org?join
	Logs	https://jabber.ietf.org/logs/wish/

Charter for Working Group

The WISH working group is chartered to specify a simple, extensible, HTTPS-based signaling protocol to establish one-way WebRTC-based audiovisual sessions between broadcasting tools and real-time media broadcast networks.

https://datatracker.ietf.org/wg/wish/about/

... and a new draft for the WHIP specification!

Workgroup:	wish	
Internet-Draft:	draft-ietf-wish-whip-01	
Published:	20 October 2021	
Intended Status:	Standards Track	
Expires:	23 April 2022	

Authors:	S. Murillo	A. Gouaillard
	CoSMo Software	*CoSMo Software*

WebRTC-HTTP ingestion protocol (WHIP)

Abstract

While WebRTC has been very sucessful in a wide range of scenarios, its adoption in the broadcasting/streaming industry is lagging behind. Currently there is no standard protocol (like SIP or RTSP) designed for ingesting media into a streaming service using WebRTC and so content providers still rely heavily on protocols like RTMP for it.

These protocols are much older than WebRTC and by default lack some important security and resilience features provided by WebRTC with minimal overhead and additional latency.

The media codecs used for ingestion in older protocols tend to be limited and not negotiated. WebRTC includes support for negotiation of codecs, potentially alleviating transcoding on the ingest node (wich can introduce delay and degrade media quality). Server side transcoding that has traditionally been done to present multiple renditions in Adaptive Bit Rate Streaming (ABR) implementations can be replaced with simulcasting and SVC codecs that are well supported by WebRTC clients. In addition, WebRTC clients can adjust client-side encoding parameters based on RTCP feedback to maximize encoding quality.

Encryption is mandatory in WebRTC, therefore secure transport of media is implicit.

This document proposes a simple HTTP based protocol that will allow WebRTC based ingest of content into streaming servics and/or CDNs.

Table of Contents ▲

https://www.ietf.org/archive/id/draft-ietf-wish-whip-01.html

WebRTC-HTTP ingestion protocol (WHIP)

- HTTP-based signalling to create `sendonly` PeerConnections
  - HTTP POST to send SDP offer, and get an SDP answer in the response
  - Teardown of sessions using HTTP DELETE
- Authentication and authorization via Bearer tokens
  - https://www.rfc-editor.org/rfc/rfc6750.html
- Trickle and ICE restart via HTTP PATCH and SDP fragments
  - https://www.rfc-editor.org/rfc/rfc8840.html
- Everything else is your usual WebRTC!
  - ICE, DTLS, etc.

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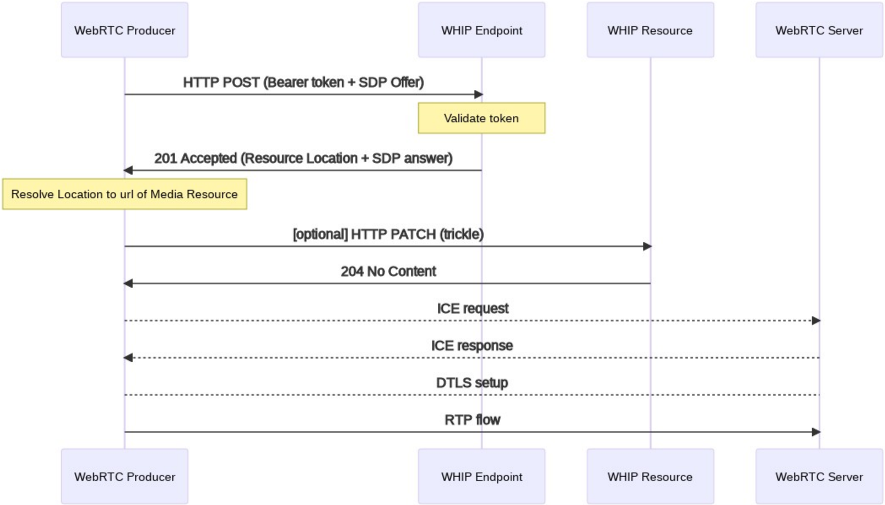
A few sequence diagrams



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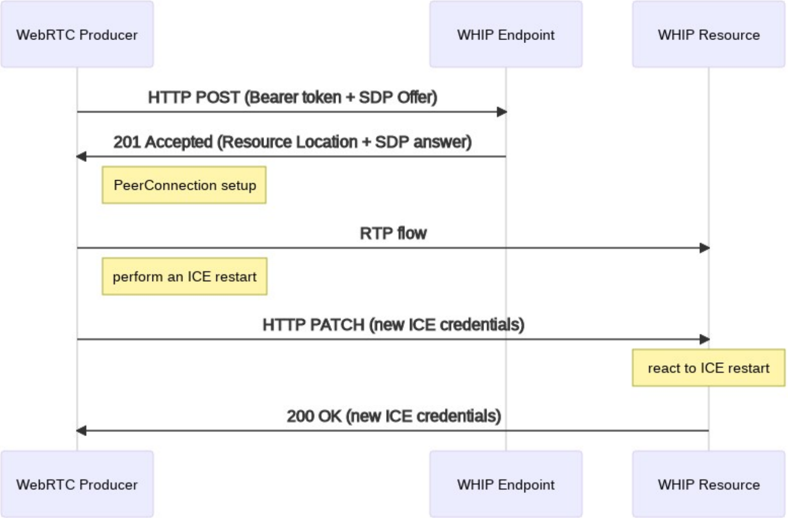

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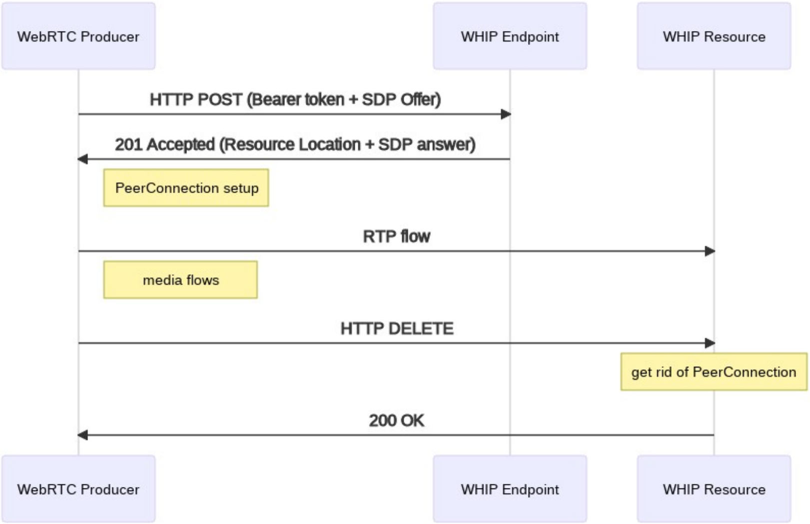

A WHIP server based on Janus

- Janus is a popular WebRTC server, so good option for WHIP
  - It implements its own JSON-based API, though (Janus API)
- Simple (and transparent) solution: basic API translator in front of Janus
  - WHIP API maps quite simply to set of Janus API primitives
  - No need to change anything in the WebRTC stack
- Implemented simple prototype using node.js and Express
  - REST server that implements the WHIP API, and talks to Janus accordingly
  - Only takes care of ingest: distribution out of scope (e.g., via SOLEIL)

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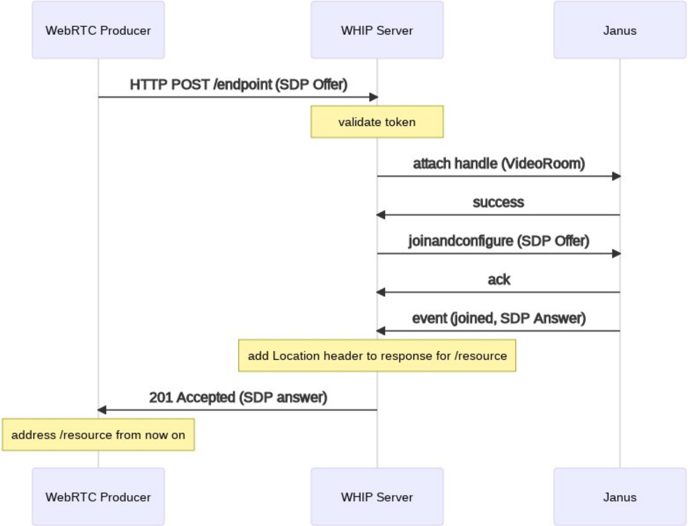

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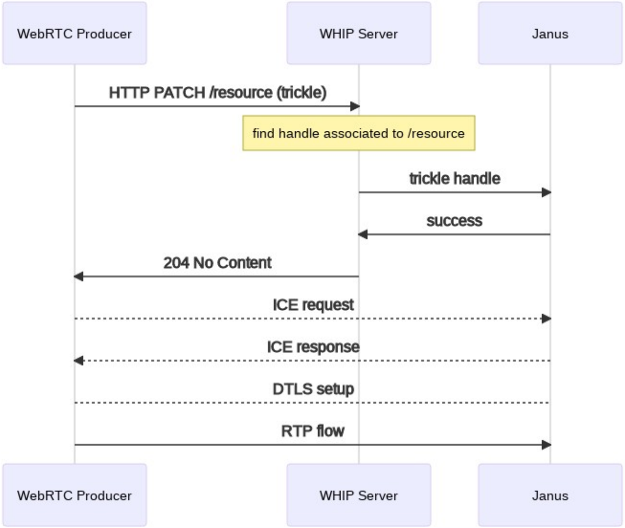

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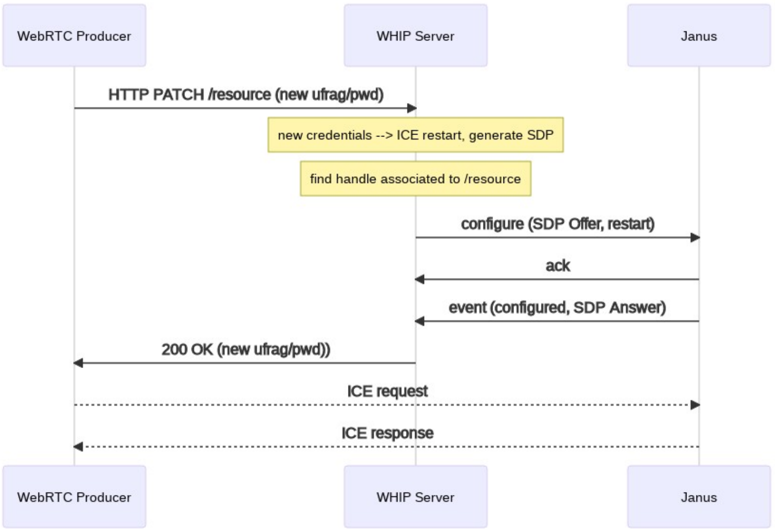

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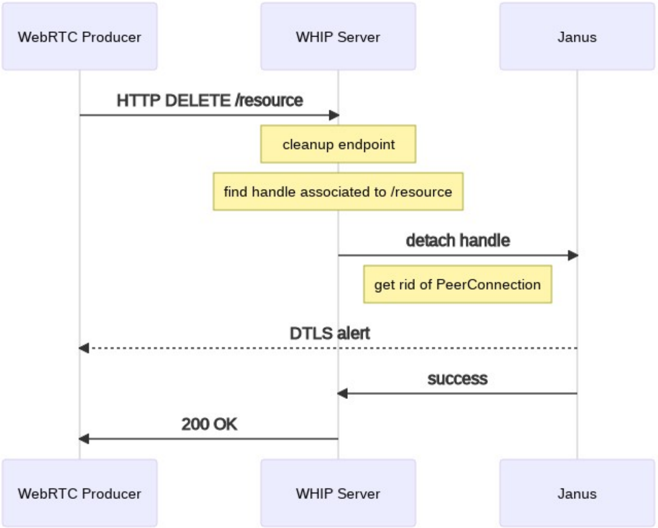

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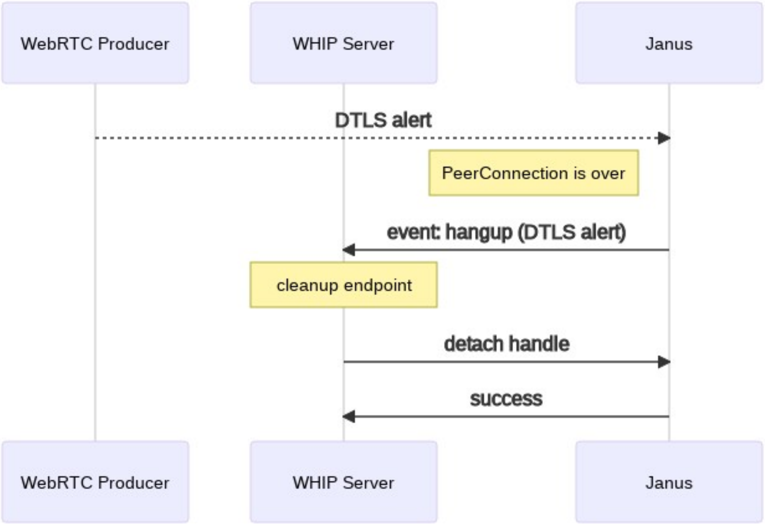


Simple WHIP Server in action 😎

WHIP server

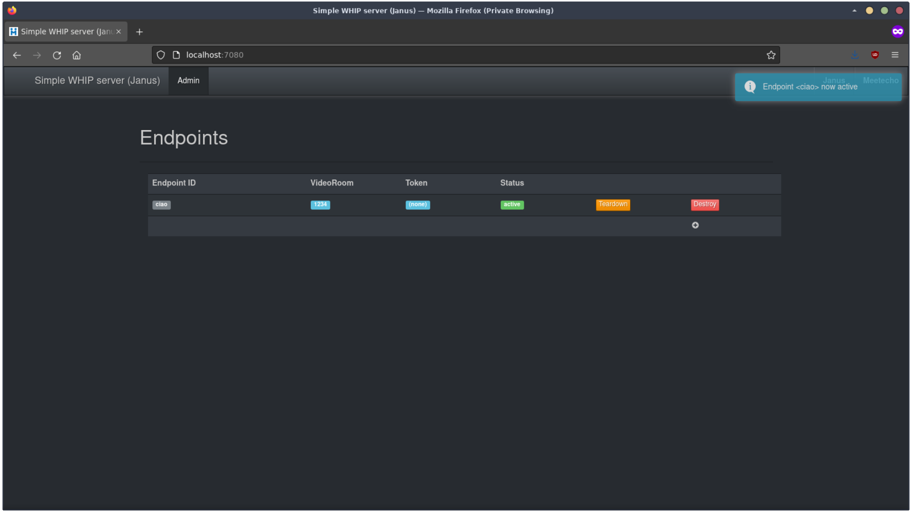
File Edit View Terminal Tabs Help

```
[lminiero@lminiero server]$ npm start

> janus-whip-server@0.0.1 start /home/lminiero/Work/code/services/whip/server
> DEBUG=whip:*,-whip:debug,janus:*,-janus:debug,-janus:vdebug node src/server.js

[1. Janus]
Connecting to Janus: { address: 'ws://127.0.0.1:8188' }
  janus:info Connecting to ws://127.0.0.1:8188 +0ms
  janus:info Janus WebSocket Client Connected +7ms
  janus:info Janus session ID is 1252536092417283 +2ms
  whip:info Connected to Janus: ws://127.0.0.1:8188 +0ms
[2. WHIP REST API]
WHIP REST API listening on *:7080
WHIP server prototype started!
[ 'Janus OK', 'WHIP REST API OK' ]
  whip:info [ciao] Created new WHIP endpoint +32s
  whip:info [ciao] Publishing to WHIP endpoint +5s
  whip:info [ciao] Terminating WHIP session +12s
  whip:info [ciao] Publishing to WHIP endpoint +23s
  whip:info [ciao] PeerConnection detected as closed +8s
```

Basic UI to create/manage endpoints



Writing a WHIP client for testing

- Needs to support HTTP (WHIP API) and have a WebRTC stack
  - Browsers are the obvious choice, but what about a native solution?
  - Many broadcasters today use custom tools (e.g., OBS)
- Unfortunately OBS-WebRTC is not currently an option
  - Used legacy WHIP API, and currently only supports Millicast ingestion
- Chose GStreamer's `webrtcbin`[2] for the purpose
  - Used it already with success in other applications (e.g., JamRTC)
  - Modular and very powerful, so easy to feed with external sources

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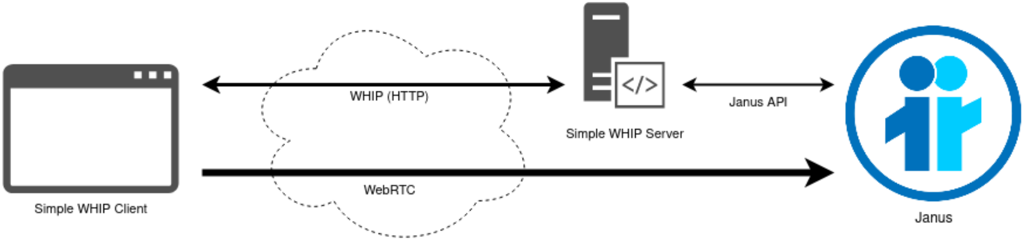
Simple WHIP Client options

```
Usage:
  whip-client [OPTION?] -- Simple WHIP client

Help Options:
  -h, --help         Show help options

Application Options:
  -u, --url          Address of the WHIP endpoint (required)
  -t, --token        Authentication Bearer token to use (optional)
  -A, --audio        GStreamer pipeline to use for audio (optional, required if audio-only)
  -V, --video        GStreamer pipeline to use for video (optional, required if video-only)
  -n, --no-trickle Don't trickle candidates, but put them in the SDP offer (default: false)
  -f, --follow-link Use the Link headers returned by the WHIP server to automatically configure STUN/TURN servers to
          use (default: false)
  -S, --stun-server STUN server to use, if any (stun://hostname:port)
  -T, --turn-server TURN server to use, if any; can be called multiple times
          (turn(s)://username:password@host:port?transport=[udp,tcp])
  -F, --force-turn In case TURN servers are provided, force using a relay (default: false)
  -l, --log-level  Logging level (0=disable logging, 7=maximum log level; default: 4)
```

Simple WHIP Client example

```
./whip-client -u http://localhost:7080/whip/endpoint/abc123 \
    -t verysecret \
    -A "audiotestsrc is-live=true wave=red-noise ! audioconvert !
       audioresample ! queue ! opusenc ! rtpopuspay pt=100 ! queue !
       application/x-rtp,media=audio,encoding-name=OPUS,payload=100" \
    -V "videotestsrc is-live=true pattern=ball ! videoconvert ! queue !
       vp8enc deadline=1 ! rtpvp8pay pt=96 ! queue !
       application/x-rtp,media=video,encoding-name=VP8,payload=96" \
    -S stun.l.google.com:19302
```

WHIP client and server + Janus



Simple WHIP Client in action 😎

WHIP client

File Edit View Terminal Tabs Help

```
[WHIP] Initializing the GStreamer pipeline:
webrtcbin name=sendonly bundle-policy=3   videotestsrc is-live=true pattern=ball
 ! videoconvert ! queue ! vp8enc deadline=1 ! rtpvp8pay pt=96 ssrc=2 ! queue ! a
pplication/x-rtp,media=video,encoding-name=VP8,payload=96 ! sendonly. audiotests
rc is-live=true wave=red-noise ! audioconvert ! audioresample ! queue ! opusenc
! rtpopuspay pt=100 ssrc=1 ! queue ! application/x-rtp,media=audio,encoding-name
=OPUS,payload=100 ! sendonly.
[WHIP] Starting the GStreamer pipeline
[WHIP] Creating offer
[WHIP] Offer created
[WHIP] Setting local description
[WHIP] Sending SDP offer (1167 bytes)
[WHIP] Resource URL: http://localhost:7080/whip/resource/ciao
[WHIP] Received SDP answer (1385 bytes)
[WHIP] Setting remote description
[WHIP] ICE gathering started...
[WHIP] PeerConnection connecting...
[WHIP] ICE connecting...
[WHIP] ICE completed
[WHIP] DTLS connecting...
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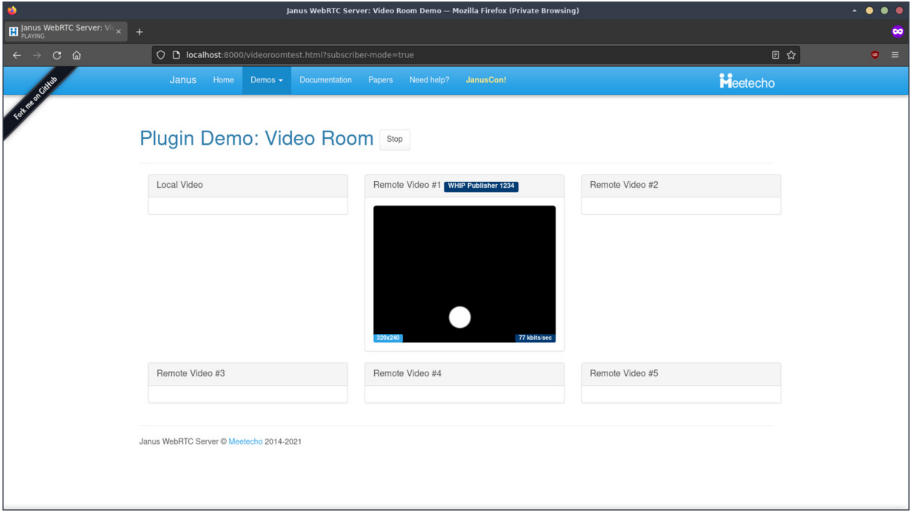
Testing my WHIP client with Janus



Other WHIP implementations: servers

- Lorenzo Miniero – Simple WHIP Server (Janus)
  - https://github.com/lminiero/simple-whip-server
- Juliusz Chroboczek – Galene
  - https://github.com/jech/galene/tree/whip
- Sergio Garcia Murillo – Millicast integration
  - https://millicast.com/
- Cameron Elliott – Deadsfu
  - https://github.com/x186k/deadsfu

Other WHIP implementations: clients

- Lorenzo Miniero – Simple WHIP Client (GStreamer)
  - https://github.com/lminiero/simple-whip-client
- Sergio Garcia Murillo – whip-js (JavaScript)
  - https://github.com/medooze/whip-js/
- Gustavo Garcia – whip-go (Go)
  - https://github.com/ggarber/whip-go/
- Tim Panton – whipi (Java / Raspberry Pi)
  - https://github.com/pipe/whipi
- Alberto Gonzalez Trastoy – free-whip (Python)
  - https://github.com/agonza1/free-whip/
- Cameron Elliott – whip-whap-js (JavaScript)
  - https://github.com/x186k/whip-whap-js

WHIP interop tests @ IETF 112 Hackathon!

WHIP Interoperability Tests

Lorenzo Miniero, on behalf of the group

IETF 112 Hackathon
November 5th, 2021

https://github.com/IETF-Hackathon/ietf112-project-presentations/blob/main/ietf112-hackathon-whip.pdf

WHIP interop tests @ IETF 112 Hackathon!

	Simple WHIP Server	Galene	Millicast	deadsfu
Simple WHIP Client	🙂	🙂	🙂	😐
whip-js	🙂	🙂	🙂	😕
whip-go	🙂	🙂	🙂	☹️
whipi	🙂	😐	😐	☹️
free-whip	😐	😕	☹️	☹️
whip-whap-js	😐	😕	☹️	🙂

Integration WHIP in broadcasting workflows

- Old version of OBS-WebRTC did have WHIP support
  - Tested for my blog post from last year
  - Implemented legacy WHIP API, and used libwebrtc
- No popular streamer tool supports WHIP yet, though
  - WHIP will make it easy for the signalling part...
  - ... but they'll still need a working WebRTC stack
- Why not start with a more “loose” integration then?
  - Keeping on using existing tools as they work today
  - Somehow get them to work with my GStreamer-based WHIP client

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Enter NDI!

https://www.meetecho.com/blog/webrtc-ndi/
https://www.meetecho.com/blog/webrtc-ndi-part-2/

What is NDI?

- Network Device Interface (NDI)
  - Royalty-free software standard developed by NewTek
  - https://www.ndi.tv/
- Live exchange of multimedia streams within the same LAN
  - Multichannel and uncompressed media streams (high quality)
  - mDNS used for service discovery
- Easy to use (and integrate) native SDK
  - Available on Windows, Linux, MacOS, Android, etc.
  - VLC team working on an alternative implementation
- Widely used in the broadcasters industry
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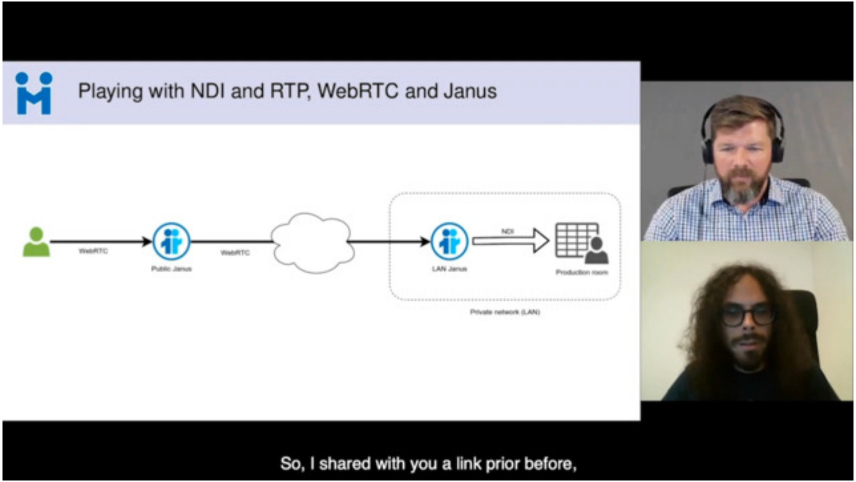
We’ve talked about WebRTC-to-NDI before...



What about NDI-to-WebRTC, though?

- There's a cool NDI plugin for GStreamer
  - Makes it easy to use NDI sources in GStreamer pipelines
  - https://github.com/teltek/gst-plugin-ndi
- Hey, our WHIP client is based on GStreamer too!
  - Audio and video pipelines are customizable (command line)
  - NDI plugin as source for the media $\longrightarrow$ encoders/WebRTC will do the rest
- Of course, we need something that generates NDI
  - OBS has an NDI plugin, for NDI input *and* output
  - https://github.com/Palakis/obs-ndi

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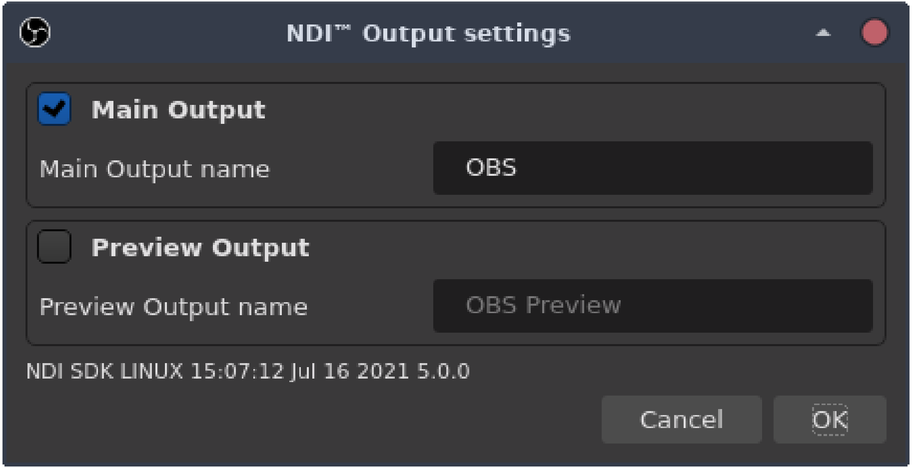
1. Configure NDI output in OBS



2. Create your scenes in OBS

3. Setup the WHIP client to capture NDI

```
Terminal - lminiero@lminiero:~/Work/code/services/whip/client
File  Edit  View  Terminal  Tabs  Help
[WHIP] Initializing the GStreamer pipeline:
webrtcbin name=sendonly bundle-policy=3   ndisrc ndi-name="lminiero (OBS)" ! ndi
srcdemux name=n n.video ! videoconvert ! vp8enc deadline=1 target-bitrate=500000
 ! rtpvp8pay pt=96 ssrc=2 ! queue ! application/x-rtp,media=video,encoding-name=
VP8,payload=96 ! sendonly. n.audio ! audioconvert ! audioresample ! audiobuffers
plit output-buffer-duration=2/50 ! queue ! opusenc ! rtpopuspay pt=100 ssrc=1 !
queue ! application/x-rtp,media=audio,encoding-name=OPUS,payload=100 ! sendonly.
[WHIP] Starting the GStreamer pipeline
[WHIP] Creating offer
[WHIP] Offer created
[WHIP] Setting local description
[WHIP] Sending SDP offer (1163 bytes)
[WHIP] Resource URL: http://localhost:7080/whip/resource/test
[WHIP] Received SDP answer (1497 bytes)
[WHIP] Setting remote description
[WHIP] ICE gathering started...
[WHIP] PeerConnection connecting...
[WHIP] ICE connecting...
[WHIP] ICE completed
[WHIP] DTLS connecting...
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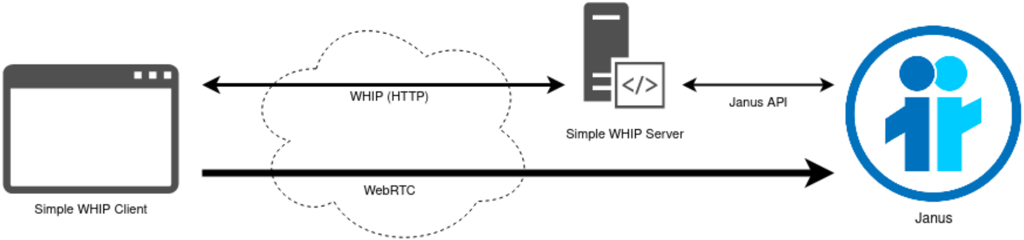
Ready for our demo!



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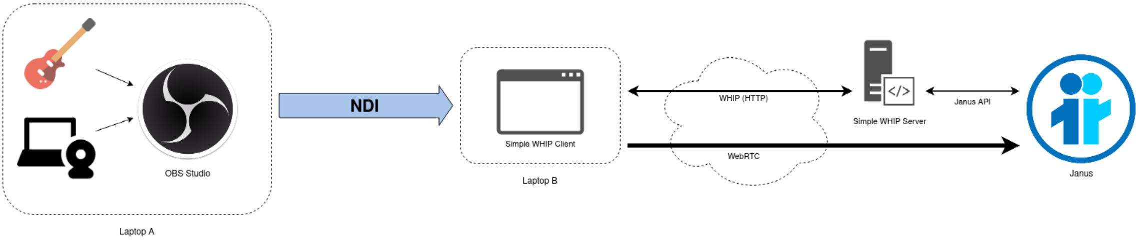

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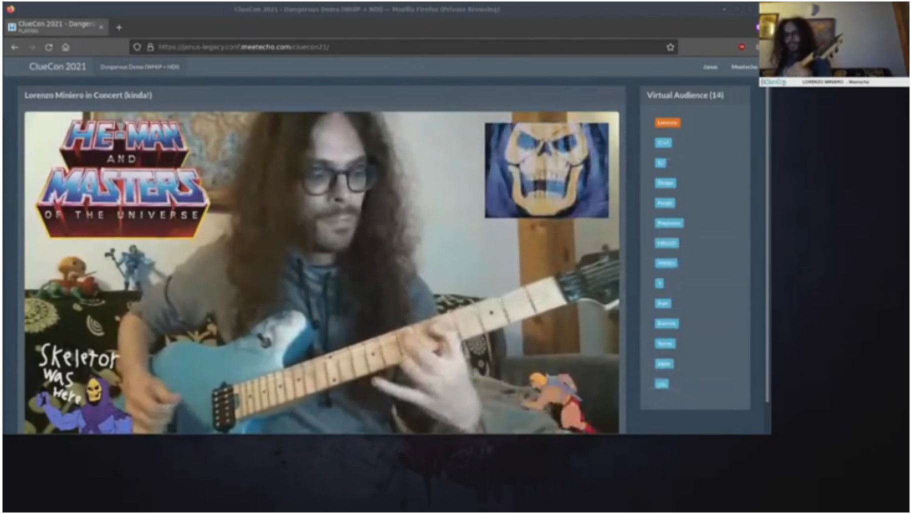

More details in a recent CommCon talk

https://2021.commcon.xyz/talks/whip-ndi-and-janus-genesis-of-a-broadcasting-demo

Next step: broadcasting the stream

- WHIP server + Janus get you in a VideoRoom, and it’s a good starting point
  - That’s the whole point of WebRTC ingest!
  - Already “consumable” via VideoRoom itself (SFU)
- Janus VideoRoom plugin not really optimized for broadcasting, though
  - Conceived for videoconferencing use cases
  - Will not work well if you have to feed, e.g., 100’s or 1000’s
- Janus Streaming plugin a much better choice
  - Natively optimized for doing one-to-many
  - Can receive media from VideoRoom (and so WHIP) via “RTP forwarders”
  - Even better, multiple Janus instances can work together

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Ph.D Thesis on WebRTC broadcasting (2015)

UNIVERSITÀ DEGLI STUDI DI NAPOLI FEDERICO II

Facoltà di Ingegneria

Dottorato di Ricerca in Ingegneria Informatica ed Automatica
XXVII Ciclo
Dipartimento di Ingegneria Elettrica e delle Tecnologie dell'Informazione

SOLEIL: STREAMING OF LARGE SCALE EVENTS OVER INTERNET CLOUDS

LORENZO MINIERO

Ph.D. Thesis

Using SOLEIL for the purpose

- "Streaming Of Large scale Events over Internet cLouds" (Ph.D Thesis)
  - In a nutshell, tree-based distribution of WebRTC feeds
  - Ingest and edges are WebRTC (Janus), everything in the middle just RTP
- Working with RTP in intermediate layers has many advantages
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  - You can even take advantage of multicast, here
- Just needs RTP forwarding to start everything
  - PR available in WHIP server to do RTP forwarding (merged)
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To learn more about RTP forwarders...

FOSDEM 20

Janus as a WebRTC “enabler”

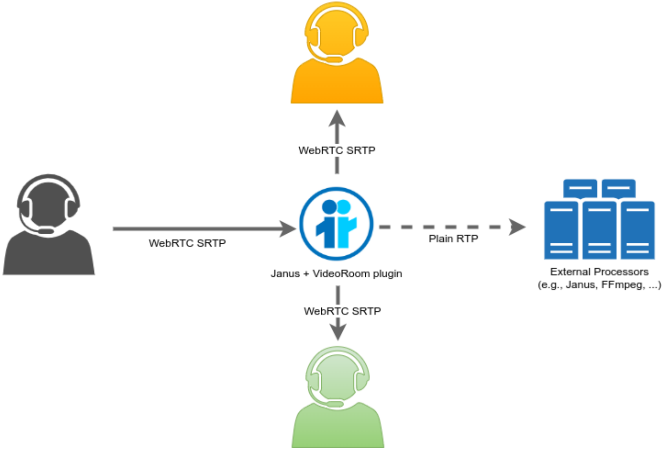
Having fun with RTP and external applications

Lorenzo Miniero
@elminiero

FOSDEM 2020 Real Time devroom
2nd February 2020, Brussels

https://archive.fosdem.org/2020/schedule/event/janus/

To learn more about RTP forwarders...



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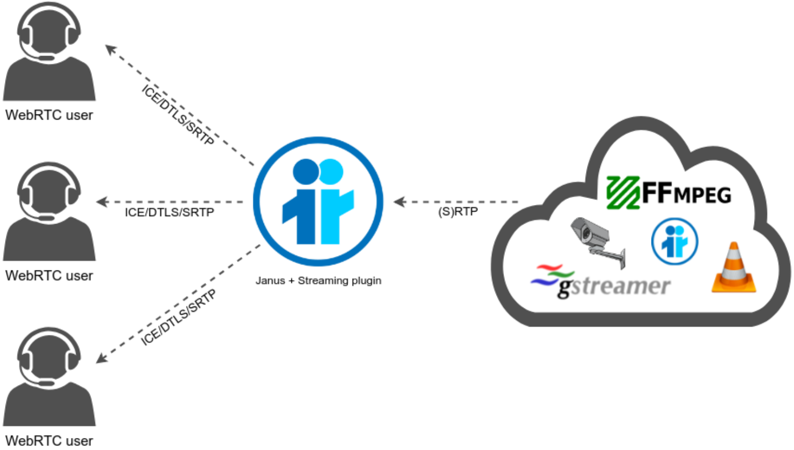

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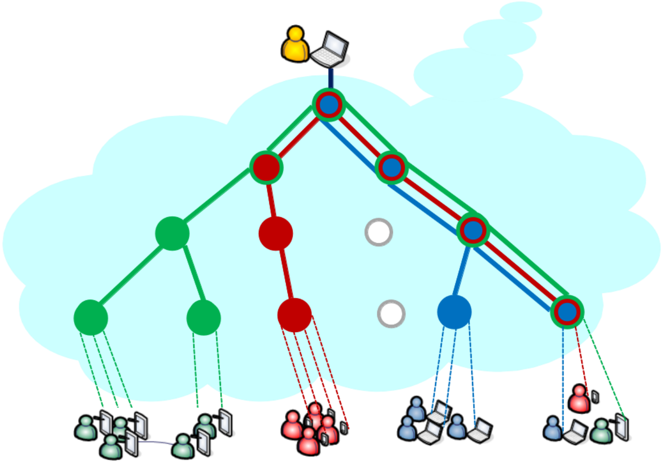
Distributing WHIP Janus streams via SOLEIL



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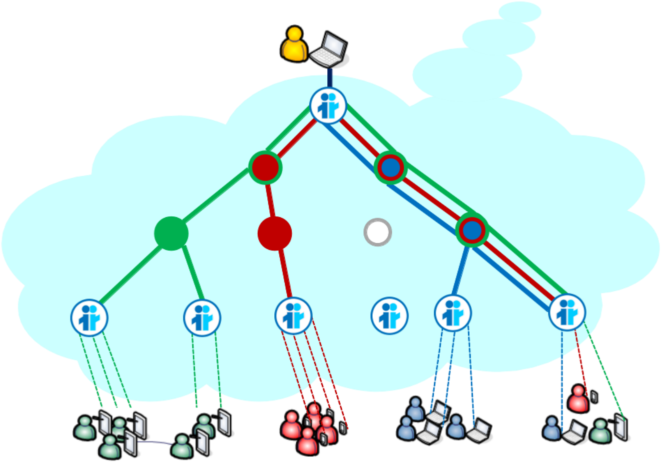

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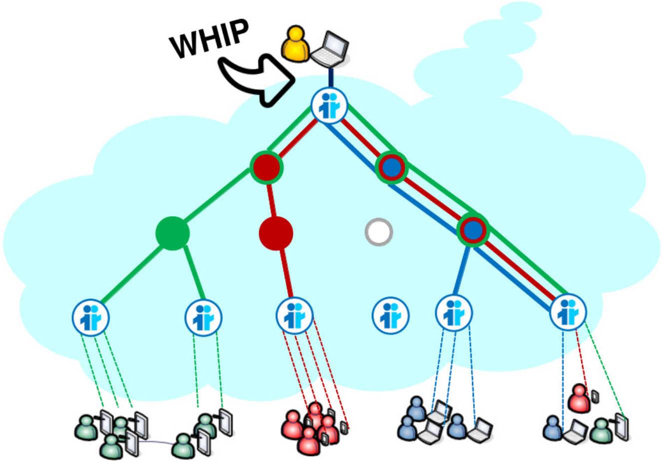

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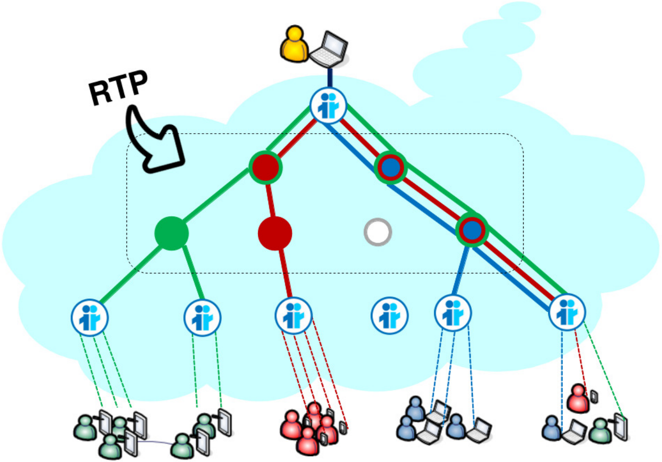

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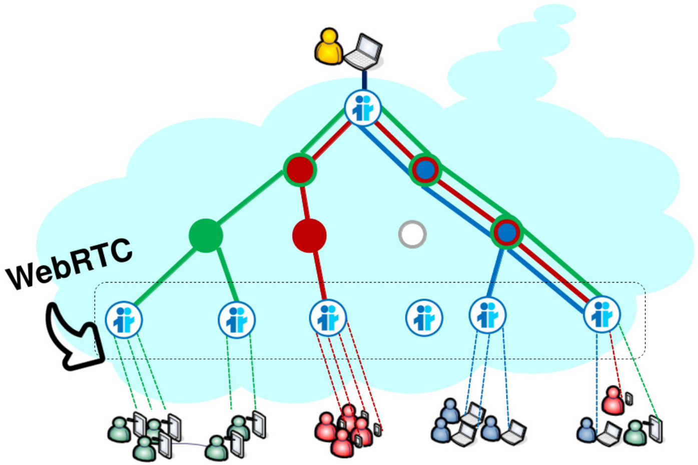
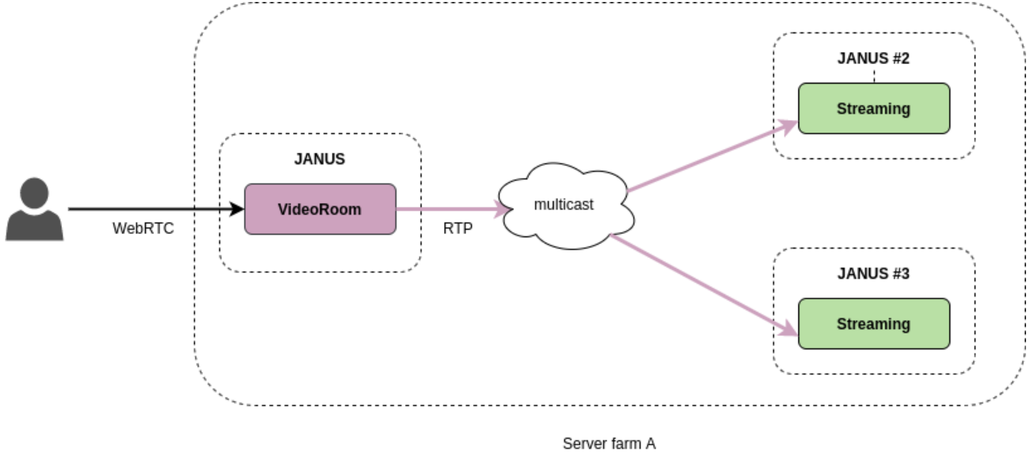


Leveraging multicast internally for RTP



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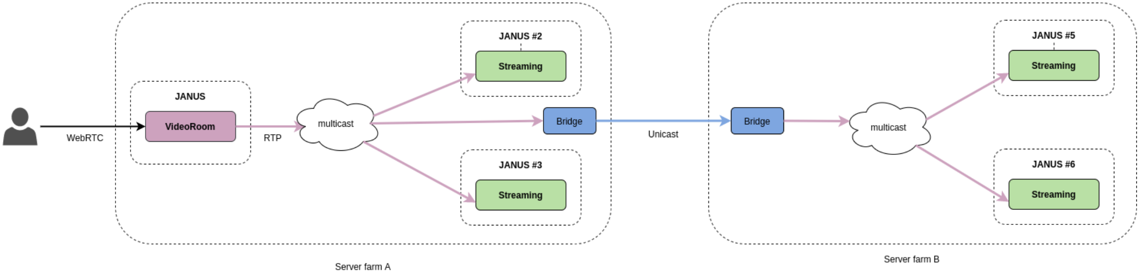
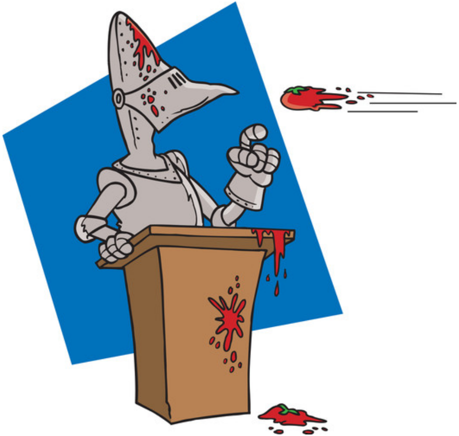


Thanks! Questions? Comments?



Contacts

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