Implementation Experience
WHIP in Janus (and GStreamer)

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Detailed blog post on integration

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

IIT RTC 2021 Presentation

https://www.youtube.com/watch?v=b_QBd3WnGgY

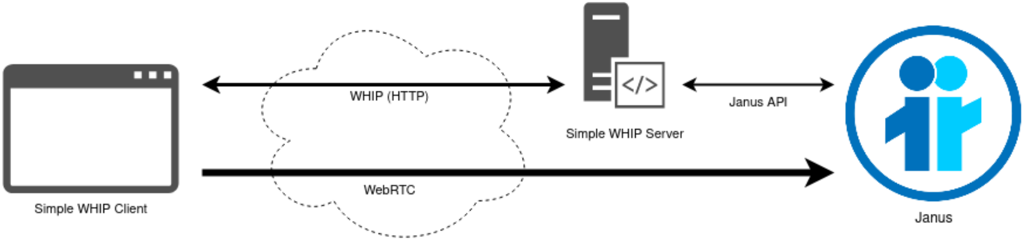
A WHIP server based on Janus

- Janus is a popular WebRTC server, so a 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

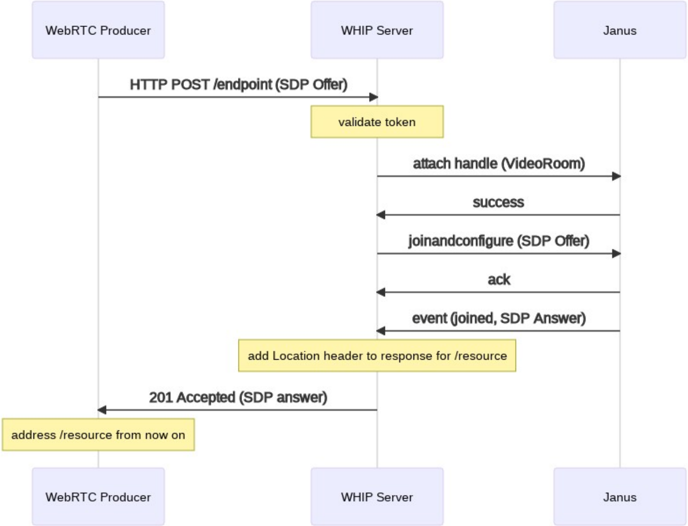
Simple WHIP Server

https://github.com/lminiero/simple-whip-server/

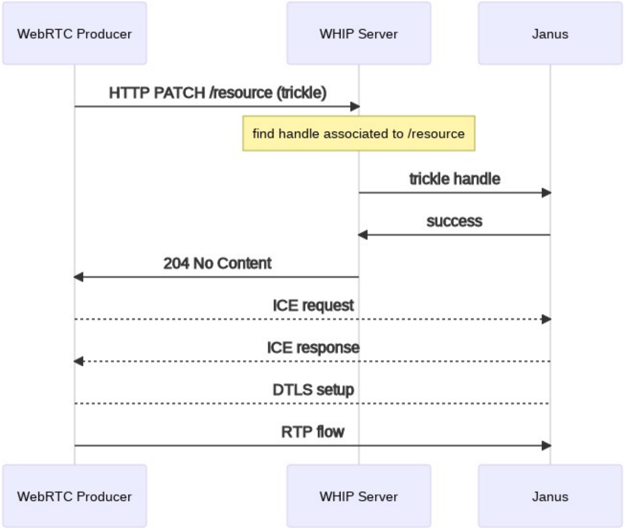
WHIP client and server + Janus



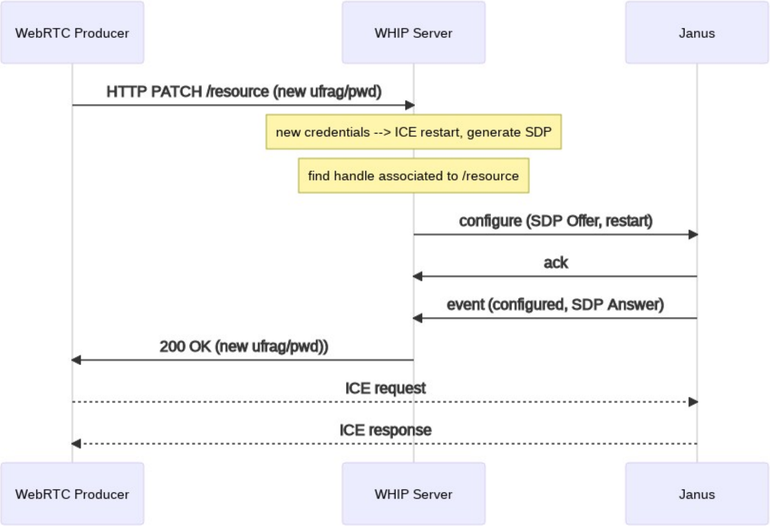
Mapping WHIP interactions to the Janus API



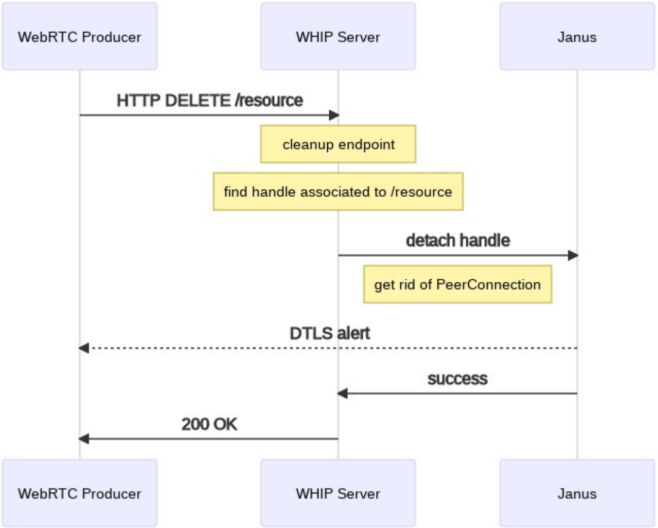
Mapping WHIP interactions to the Janus API



Mapping WHIP interactions to the Janus API



Mapping WHIP interactions to the Janus API



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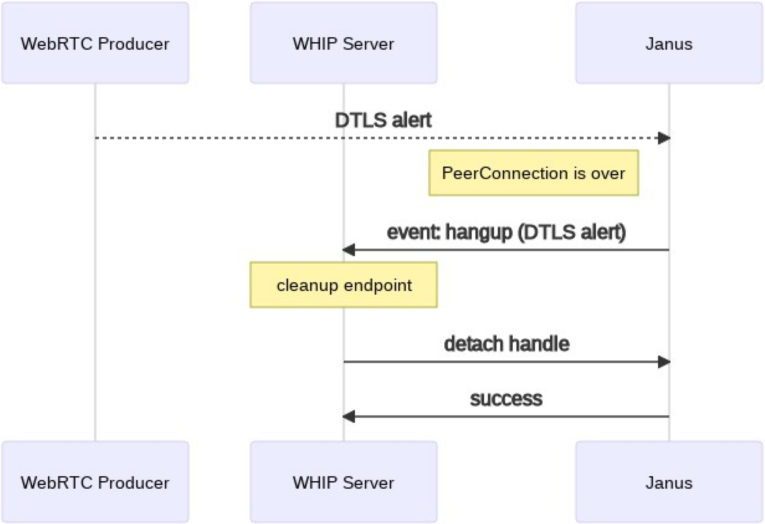


Writing a WHIP client for testing

- 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`**[1] 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

Simple WHIP Client

https://github.com/lminiero/simple-whip-client/

[1] https://gstreamer.freedesktop.org/documentation/webrtc/

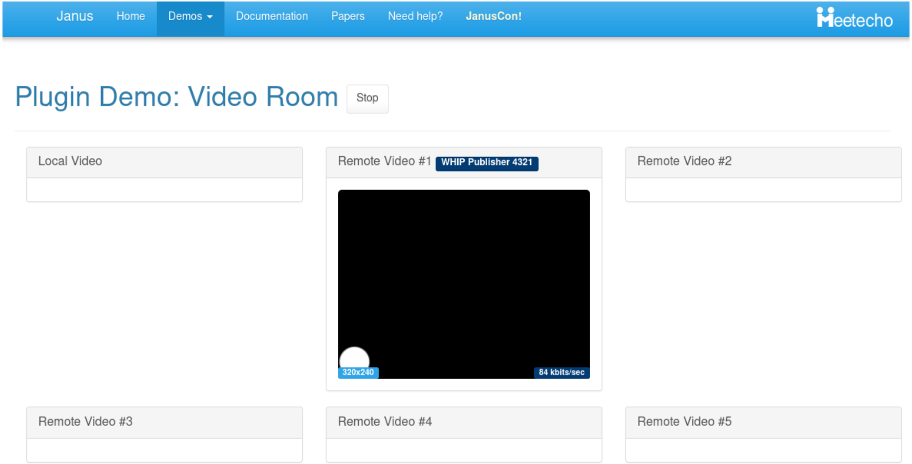
WHIP client features (and limitations)

- Almost everything supported
  - Trickle (PATCH), STUN/TURN (via `Link` too), tokens, DELETE, etc.
  - Added support for non-trickle too (manual addition of candidates to SDP)
  - Option to force TURN (`iceTransportPolicy:"relay"` equivalent)
- Customizable audio/video pipelines
  - Easy to experiment with different sources and codecs
- A couple of things not supported in `webrtcbin` yet, though
  - ICE restarts (there seems to be a PR, though)
  - `Link` support in POST (we currently only do it via OPTIONS in the client)

Simple WHIP Client example

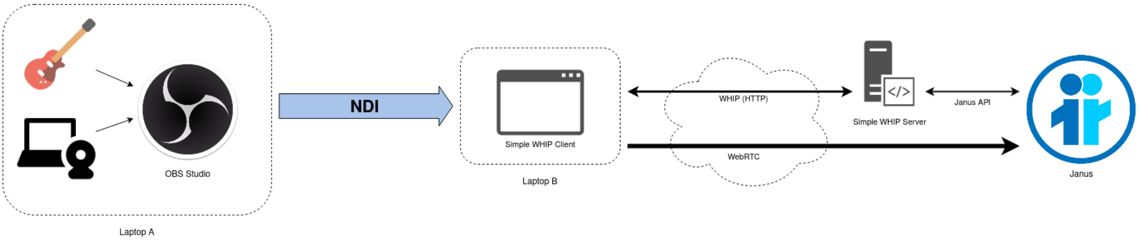
```
./whip-client -u
   https://mercury.conf.meetecho.com:8443/whip/endpoint/test \
   -t hackathon \
   -A "audiotestsrc is-live=true wave=red-noise ! audioconvert !
      audioresample ! queue ! opusenc ! rtpopuspay pt=100 ! queue !
      application/x-rtp,media=audio,encoding-name=OPUS,payload=100" \
   -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
```

Simple WHIP Client example



https://janus-legacy.conf.meetecho.com/videoroomtest.html?room=4321&subscriber-mode=true

Having fun with WHIP ☺



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

Having fun with WHIP ☺

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