Elaborate WebRTC Media with Artificial Intelligence

Let Janus, OpenCV and A.I. work together!

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FOSDEM

JANUS WEBRTC SERVER

WebRTC and Artificial Intelligence?

- **Absolutely fascinating matter**
  Real Time Computer Vision & Sound Recognition
  Data collection and live elaboration
  Broadcasting experience customized by users

- **A lot of scenarios with different requirements**
  Massive broadcasting, small videorooms, device networks
  Device constraints, Time constraints...
  Must choose!

- **Make it simple, it's a project so young!**
  Remove constraints
  Focus on video elaboration
  Explore the results

What we want from this project

- **Handle the video media in a comfortable way**
  Receive and elaborate the Janus RTP stream without (too much) manipulation

- **Minimize the client side effort**
  Avoid client-side elaboration
  Do not overload the client's bandwidth

- **Keep an eye on scalability**
  Put everything in a nutshell and replicate it, if needed

- **Play in the home pitch using known languages and environments**

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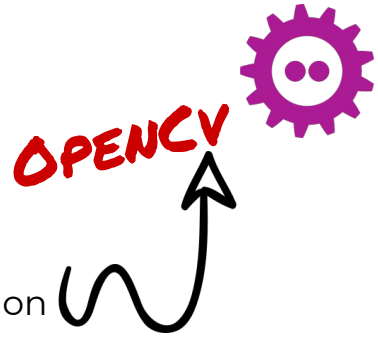

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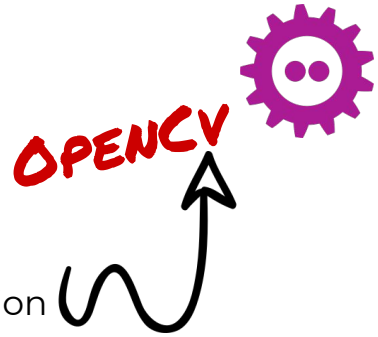

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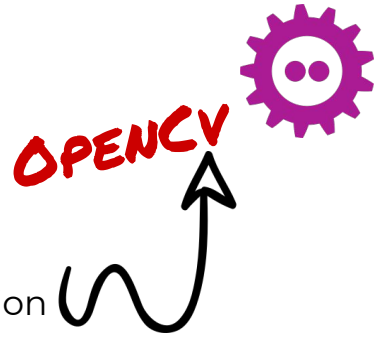

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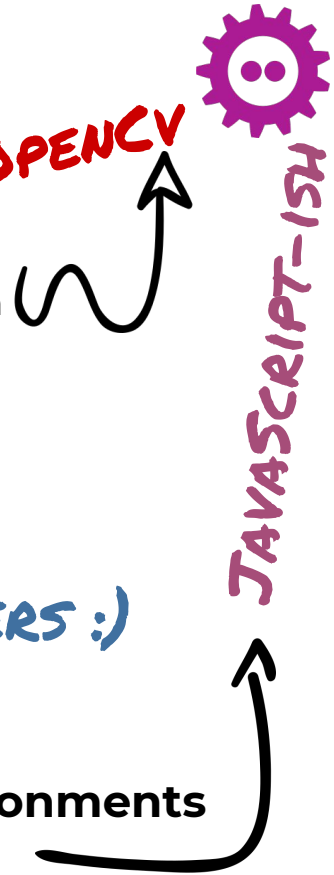

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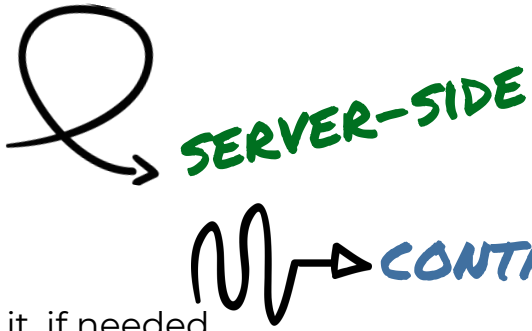
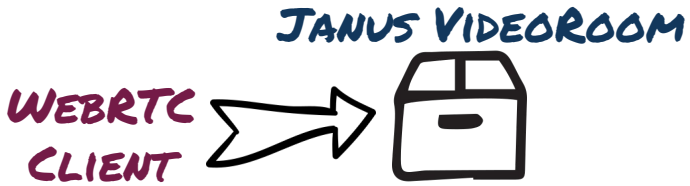


What we will use

- **Video Capture (WebRTC)**
  Janus WebRTC Server (http://github.com/meetecho/janus-gateway)
- **Video elaboration and Sampling**
  opencv4nodejs (https://github.com/justadudewhohacks/opencv4nodejs)
- **Server**
  NodeJS (https://nodejs.org)
- **Containerization**
  Docker (https://www.docker.com)
- **Language**
  TypeScript (https://www.typescriptlang.org)

Architecture

- **The Janus Videoroom Plugin**
  WebRTC entrypoint for media producers
  Forwards received stream as RTP stream



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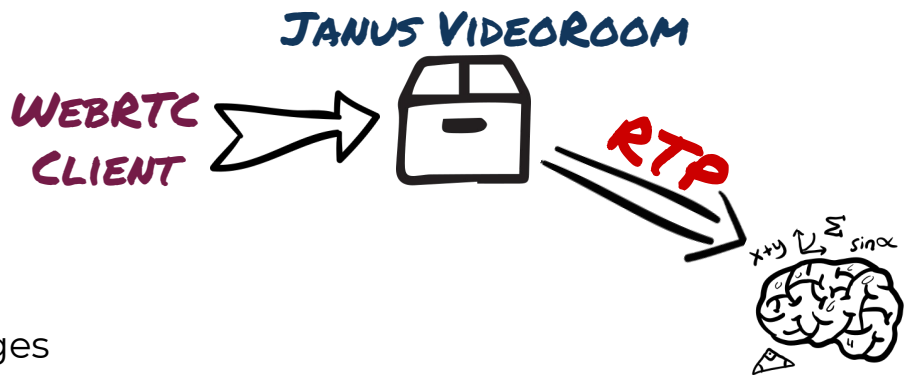

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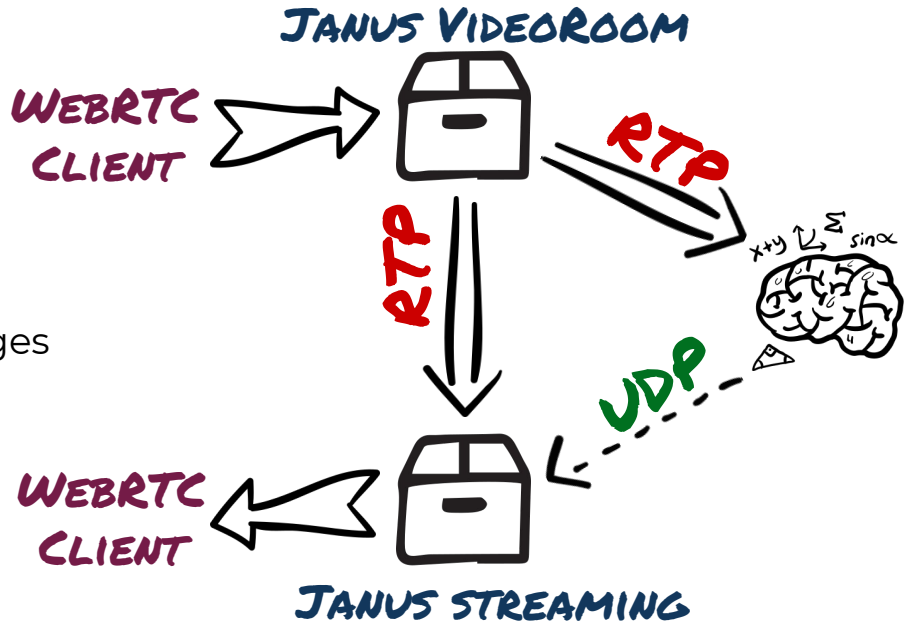
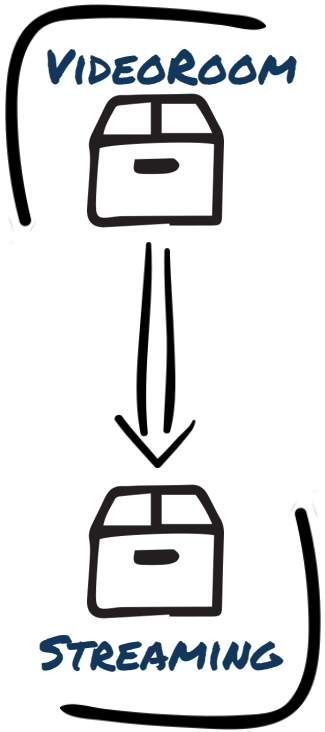
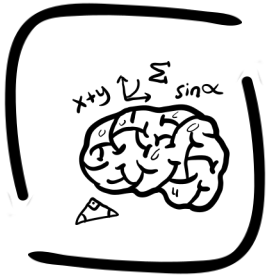
- **The Janus Streaming Plugin**
  WebRTC entrypoint for media receivers
  Forwards received streams (RTP & UDP)
  to WebRTC clients through media stream
  or datachannels



A container to rule them all

- **Two containers**
  Create a more flexible architecture





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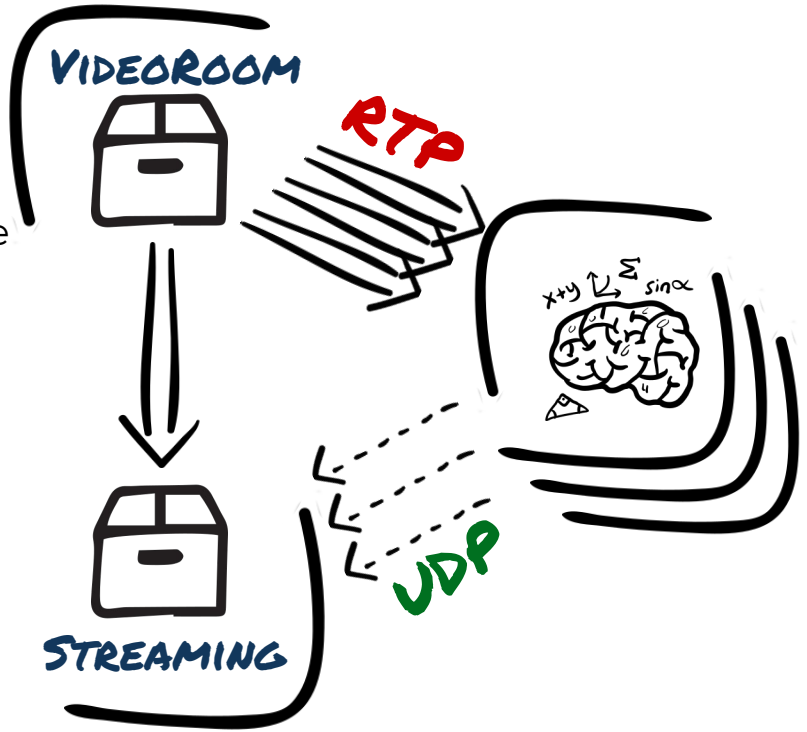
  Each of them elaborated by a dedicated instance
- **Single stream, multiple elaborations**
  Different elaborations on same stream in parallel
  Let users display results in a selective way



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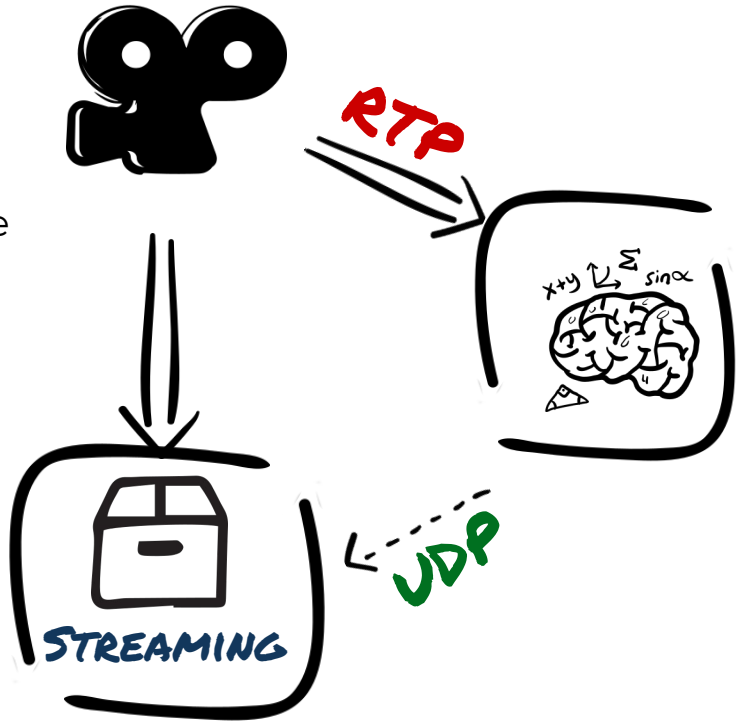
  Replace containers according to our needs
  Use an external RTP source



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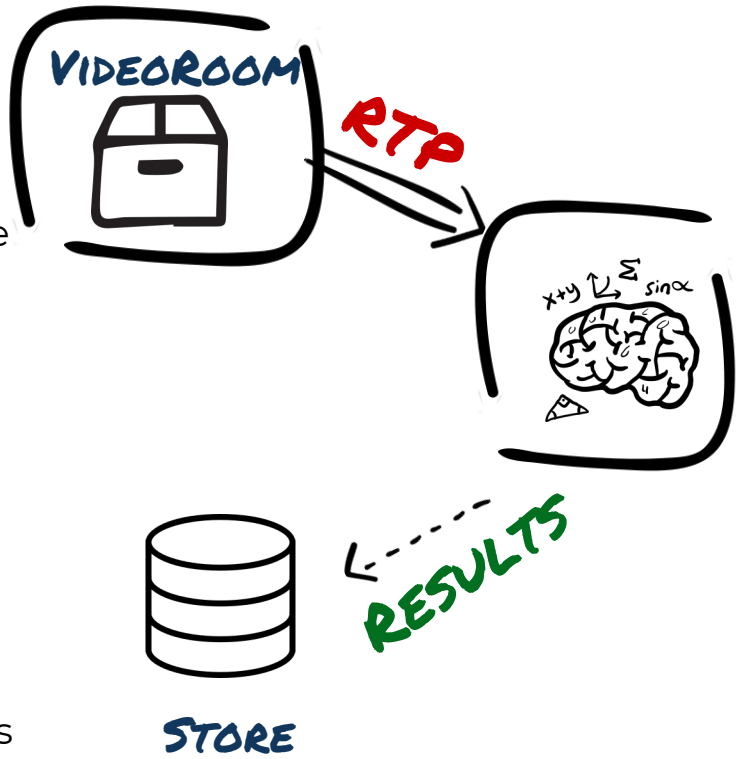
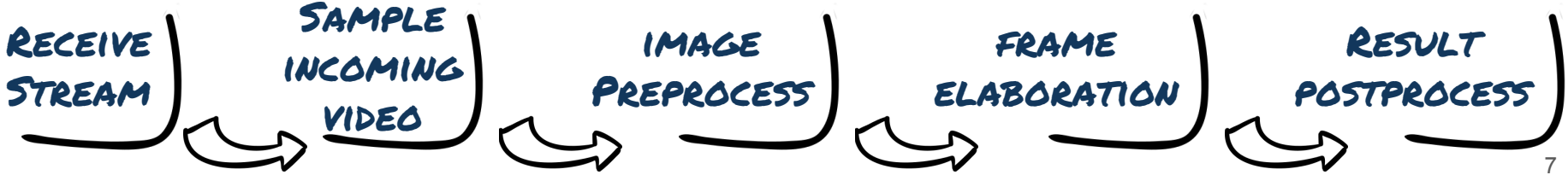
  Save elaborated data without live UDP feedbacks



Let’s Code!

- **The idea**
  Break the elaboration flow in several steps
  Create a superclass that takes care of all the under the hood tasks
  Let programmers implement only the target-specific code using a inherited class

- **The code flow of our video elaboration**



Let's Code!

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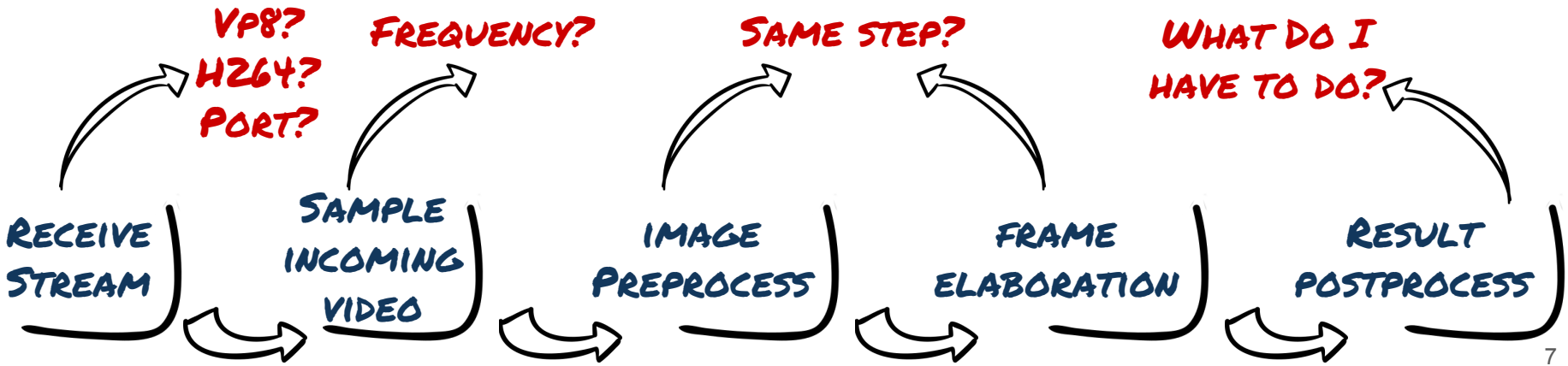


The superclass interface

```
export interface ModelMethods<T extends Detection> {
  train(trainSet: any[], label?: any[]): void,
  classify(image: cv.Mat, frame_counter: number): Promise<any[]>,
  configure(config?: object): void,
  getVideoInput(pipe?: string): object,
  stopElaboration(): void,
  startElaboration(pipe?: string, elaborating_rate?: number): Promise<void>,
  elaborateVideo(sampling_rate: number): Promise<T[]>,
  elaborateResults(output: T[]): void,
}
```

The superclass interface

- **The under-the-hood stuff**
  Getting video input once pipe is defined
  Sample video every *sampling_rate* frame and provide image to the **classify** function
  Helpers function, like the **bootstrap** and the **teardown** of the component

- **What does a programmer have to provide?**

```
abstract train(trainSet?: any[], label: any[]): void; // if needed
abstract classify(object: cv.Mat, frame_counter: number): Promise<any[]>;
abstract configure(config?: object): void;
abstract elaborateResults(output: T[]): void;
```

The *configure* method

- **Sets the GStreamer receiving pipe used by openCV module**
  i.e. *udpsrc port=20010 caps=\"application/x-rtp, media=(string)video, clock-rate=(int)90000\" ! rtpvp8depay ! vp8dec ! videoconvert ! appsink*

- **Load here everything you need for elaboration phase**
  Import a Tensorflow/Caffe model
  Import dataset to train your own model
  Connect with an external cognitive service
  Define custom configurations

```
udp_client = dgram.createSocket('udp4');
pbFile = path.resolve(‘path/to/repo/’, ‘frozen_inference_graph.pb’);
pbtxtFile = path.resolve(‘path/to/repo’, ‘ssd_mobilenet_v2_coco_2018_03_29.pbtxt’);
net = Net.loadFromTensorFlow(pbFile, pbtxtFile);
```

The *elaborateResults* method

- **Receives an array of extended *Detection* object**
  Extending the Detection class allows programmers to handle elaboration results, marshalling the received data

- **Sends data to the Janus Streaming Plugin**
  Detection results could be sent to Streaming Plugin in order to be forwarded to receivers through datachannels

- **No boundaries**
  Store data
  Define a protocol between elaborator and clients to enrich the video they are receiving
  Involve external services

The *classify* method

A lot of things could be done!

We implemented a few examples found online

Emotion recognition using third party service

Streaming Controls

Status Info

Detections

ANGER	0
CONTEMPT	0.002
DISGUST	0
FEAR	0
HAPPINESS	0.977
NEUTRAL	0.022
SADNESS	0
SURPRISE	0

Use a trained LBPHFaceRecognizer

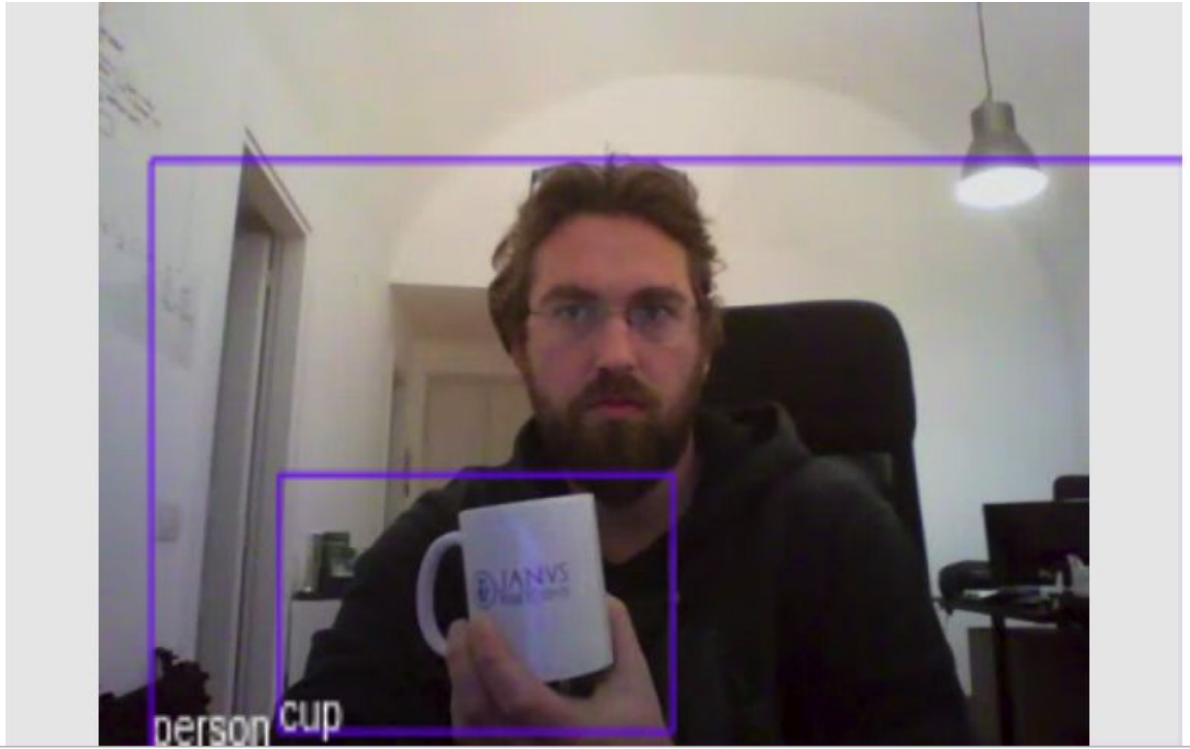
Streaming Controls

Initialize | Destroy

Fetch Streams | Stop Stream

Status Info

Object Detection loading a Tensorflow model



Streaming Controls

Initialize | Destroy

Fetch Streams | Stop Stream

Status Info

What's Next?

- **Measurements**
  Provide the capability to evaluate delays introduced by elaboration

- **Scalability**
  Make software ready-to-scale, taking advantages of Docker

- **Handle the audio and data media**
  Implement the necessary tools to treat other media

- **Human Intelligence**
  Improve our knowledge on A.I. and machine learning in order to match as many use cases as possible


ALEX
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IMPACT
IMPACT
33
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Fuzzing the Janus WebRTC Server

And why you should fuzz too

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JANVS WEBRTC SERVER

The infamous Project Zero's post

- Natalie Silvanovich's post series [1] on Google Project Zero blog
- Aiming at RTC services, focusing on **End-To-End**, **RTP** testing
- Malicious endpoint generating randomized input
- *"fairly time intensive" and "required substantial tooling"* [2]

> *"Our research found a total of 11 bugs in WebRTC, FaceTime and WhatsApp. The majority of these were found through less than 15 minutes of mutation [...] We were surprised to find remote bugs so easily in code that is so widely distributed."*

- Is our WebRTC Server safe? [3]

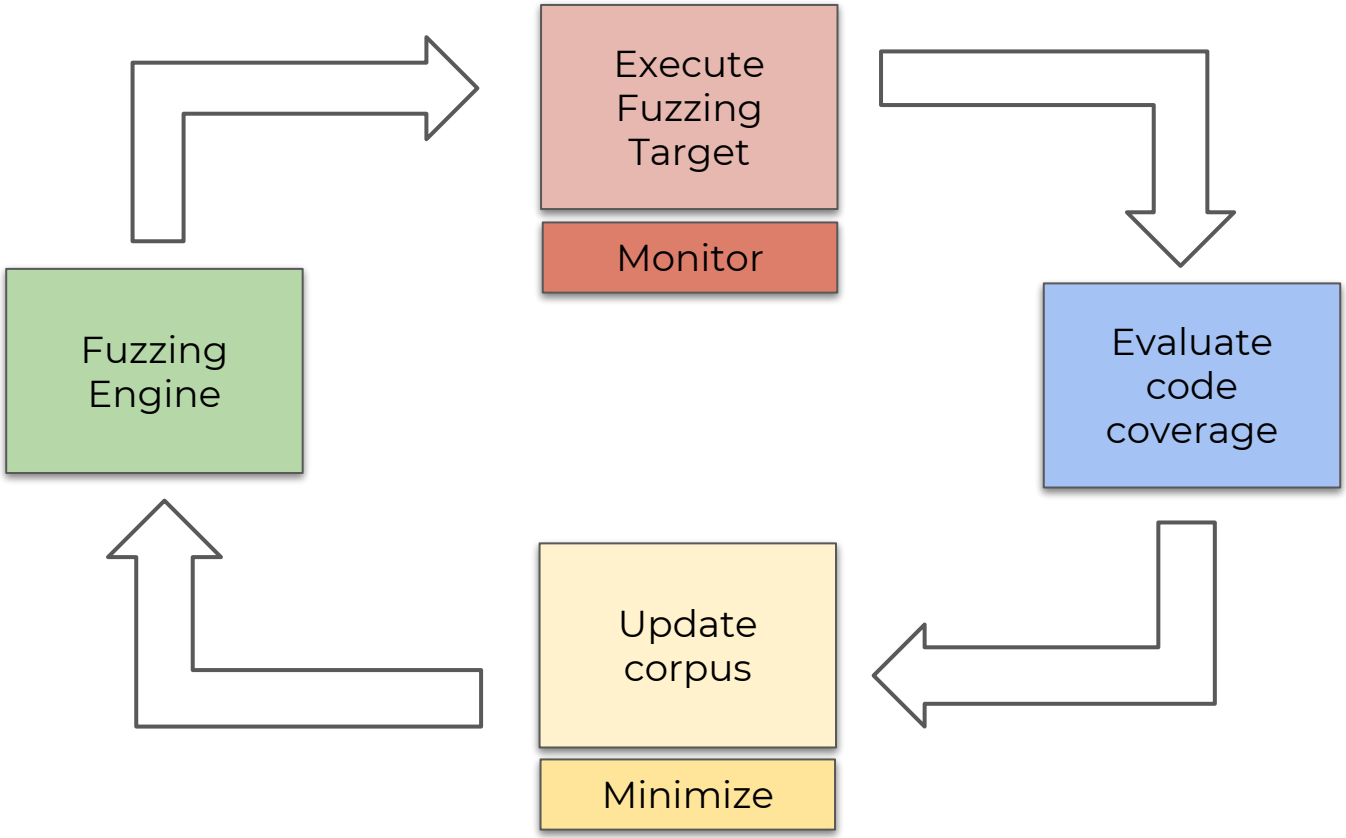
[1] https://googleprojectzero.blogspot.com/2018/12/adventures-in-video-conferencing-part-1.html
[2] https://github.com/googleprojectzero/Street-Party
[3] https://webrtchacks.com/lets-get-better-at-fuzzing-in-2019-heres-how/

Fuzz Testing

- Fuzzing is a software testing technique that consists of automatically submitting unexpected or invalid data to a program and modifying the input pattern according to a defined strategy
  - Coverage guided mutations

- Continuous checking of a Target function, input generated by an Engine
  - At every step, the Engine generates a mutated pattern depending upon
    - The current dataset known as Corpus
  - If new lines of code are covered while executing Target
    - The pattern is added to the Corpus
    - Coverage statistics get updated
  - Target execution is monitored through some tools (e.g. sanitizers)
  - Coverage data are occasionally minimized

Coverage guided fuzzing



libFuzzer

- Coverage-guided fuzzing engine [4]
- It is part of LLVM project, need to compile your sources with **Clang**
- Works **in-process** and linked with the library under test
- Feeds inputs to the target via a **fuzzing entrypoint** (target function)

- The execution of the target function is monitored with **sanitizers** tools
  - AddressSanitizer (ASan) [LLVM /GCC]
  - UndefinedBehaviorSanitizer (UBSan) [LLVM]
  - MemorySanitizer (MSan) [LLVM]

- Openssl, glibc, boringssl, SQLite, ffmpeg ...

[4] https://llvm.org/docs/LibFuzzer.html

libFuzzer

- Prepare a fuzzing target that accepts a bytes array and does something using the API under test

```
// api_fuzzer.c
int LLVMFuzzerTestOneInput(const uint8_t *Data, size_t Size) {
  TargetAPI(Data, Size);
  return 0;
}
```

- Build using **`-fsanitize=fuzzer`** flag
- Combine libFuzzer with ASan and/or UBSan!

```
> clang -g -O1 -fsanitize=fuzzer,address,undefined api_fuzzer.c
```

- `-fsanitize=fuzzer` links in the libFuzzer's main() symbol

libFuzzer

- Create a Corpus folder holding the initial **samples** (may be empty)

```
> ./my_fuzzer CORPUS_DIR
```

- Coverage incrementing test cases will be added to the corpus
- The fuzzing process will **stop** when a bug is found
- The input that triggered the bug will be **dumped** to the disk

- The fuzzer can be re-executed against the offending pattern
- Use to check if a bug has been fixed and for regression testing too!

```
> ./my_fuzzer crash-file-dump
```

Fuzzing integration in Janus

- The codebase has been checked for compilation with Clang
  - Some compilation flags have been updated to better support Clang
  - Clang generated useful warnings that led to some fixes!
- Started off from **RTCP**
- Wrote a meaningful fuzzing target (`fuzz-rtcp.c`)
  - Identify critical functions that handled raw pointers
- Added helper scripts to build (`build.sh`) and run (`run.sh`) the fuzzers
- Source [5]

[5] https://github.com/meetecho/janus-gateway/pull/1492

RTCP fuzzing target

```c
// fuzz-rtcp.c
#include "janus/rtcp.h"

int LLVMFuzzerTestOneInput(const uint8_t *data, size_t size) {
     if (size < 8 || size > 1472) return 0;
     if (!janus_is_rtcp(data, size)) return 0;
     /* Initialize an empty RTCP context */
     janus_rtcp_context ctx;

     janus_rtcp_parse(ctx, (char *)data, size);
     GSList *list = janus_rtcp_get_nacks((char *)data, size);
     ...
     if (list) g_slist_free(list);
     return 0;
}
```

Building the target

- Build needed Janus objects for the fuzzer target

```
> FUZZ_FLAGS="-g -O1 -fno-omit-frame-pointer -fsanitize=fuzzer,address ... "
> ./configure CC=clang CFLAGS="$FUZZ_FLAGS"
> make janus-log.o janus-utils.o janus-rtcp.o
> ar rcs janus-lib.a janus-log.o janus-utils.o janus-rtcp.o
```

- Collect needed Janus dependencies and build the fuzzer

```
> DEPS_CFLAGS="$(pkg-config --cflags glib-2.0)"
> DEPS_LIB="$(pkg-config --libs glib-2.0)"
> clang $FUZZ_FLAGS $DEPS_CFLAGS fuzz-rtcp.c -o fuzz-rtcp janus-lib.a $DEPS_LIB
```

Running the target

- Corpus: reused the *webrtc.org* RTCP corpus [6]
- Created **our own** too, with valid WebRTC RTCP packets and invalid patterns that crashed our server

```
> ./fuzz-rtcp fuzz-rtcp_corpus
```

- Reproduce a crash within the **debugger**

```
> ASAN_OPTIONS=abort_on_error=1 gdb --args ./fuzz-rtcp crash-file
```

[6] https://chromium.googlesource.com/external/webrtc/+/master/test/fuzzers/corpora/rtcp-corpus/

Meet OSS-Fuzz

- OSS-Fuzz [7] is Google's infrastructure dedicated to **continuous** fuzzing of critical Open Source Software (openssl, ffmpeg … Janus?)
  - Multiple fuzzing engines and sanitizers
  - Issue tracker and dashboard for collecting statistics

- To ask for a project integration you need to submit a PR and provide
  - **`Dockerfile`**: prepare the environment (fetch code, install packages)
  - **`build.sh`**: build your library and your fuzzing targets
  - **`project.yaml`**: your project descriptor

- Janus helper script **`build.sh`**
  - Can be seamlessly used for both local and OSS-Fuzz setup
  - Reads all the env vars defined in the OSSF building environment

[7] https://github.com/google/oss-fuzz

Results

- Fixed **dozens** of RTCP parsing bugs in Janus
  - Many of them were memory buffer overflows (DoS, security flaws)
  - Some of the bugs were confirmed in end-to-end testing with a malicious fuzzing browser
- Built useful tools for fuzzing and regression testing
  - Great opportunity to integrate in OSS-Fuzz for continuous fuzzing [8]
- Got in Clang compiler and some LLVM tools
- Contribute to make WebRTC a safer world [9]

[8] https://github.com/atoppi/oss-fuzz/tree/janus-gateway
[9] https://github.com/RTC-Cartel/webrtc-fuzzer-corpora

What's next

- Extend to **other protocols** creating new fuzzing targets
  - RTP, SDP ...
- Submit a PR to OSS-Fuzz
- Investigate other fuzzing engines (AFL)
- Integrate the fuzzer building script in Janus GNU Autotools scripts
  - Create a specific target in Makefile for building fuzzers

Thank You!

Naples, Italy

September 23 - 25

www.janusconf.it