The Galène videoconferencing server

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Galène

Galène is a videoconferencing server.

`https://galene.org`

You're using it now.

Galène UI

UI designed for lectures:

- you start with the camera/microphone off, switch on with *Ready*;
- you can share multiple windows;
- breakout groups created automatically;
- video mirroring (mirror in front of webcam).

Compromises were made:

- you can play videos
  (watching videos during lockdown)
  (currently broken in Chrome/Chromium);
- can switch camera on at login
  (apparently essential for meetings).

Galène UI (2)

UI still incomplete:

- some functions are only available as commands
  - type `/help` in the chat
  - students love `/msg`;
  - administator has extra commands (`/kick`, `/mute`);
- only one layout for now
  - use full screen;
  - use picture-in-picture.

Accepted by students, but not by non-CS lecturers.

(Some administrators hate it!)

Galène

Galène is a videoconferencing server:

- designed for teaching, but useful for meetings;
- easy to compile and deploy
  (15 minutes according to Dave);
- small: 7000 lines of Go and 4000 lines of JS;
- minimal server resources:
  - 5€/month VPS for 100-person lectures;
  - runs fine on a 50€ ARM board;
  - multicore scaling

  (teaching is eternally underfunded);
- can run off a read-only filesystem.

The first French lockdown

First French lockdown:

- 17 March through 11 May 2020;
- stuck in 40 m^2
  (Mayor of Paris forbids public parks!);
- working on mobility in IP networks
  (no motivation, cannot test);
- need to lecture from home.

Contradictory orders:

- University buys a Zoom licence
  (for an undisclosed sum);
- national CNRS forbids Zoom;
- local CNRS uses Zoom.

But some of us prefer self-hosted software!

The first French lockdown (2)

Need to lecture from home.

- University buys a Zoom licence;
- some of us prefer self-hosted software.

Finally, a self-hosted instance of BigBlueButton:

- often overloaded;
- only allowed for work-related purposes.

BigBlueButton:

- full of features;
- requires serious server-side resources.

The first French lockdown (3)

BigBlueButton:

- full of features (great for teaching!);
- requires serious server-side resources.

At that point, I should have:

- installed an instance of *Jitsi*; or
- written a frontent for *Janus*; or
- looked in more detail at *Ion*.

But I didn’t. I wrote my own.

Never do that!

(I mean, seriously, don’t.)

Videoconferencing is difficult

At first sight, videoconferencing is difficult:

- signalling:
  - codec negotiation;
  - NAT traversal;
- media flow distribution:
  - loss recovery;
  - congestion control;
- video quality:
  - jitter compensation;
  - lipsynch;
- video and audio codecs
  - oh my!

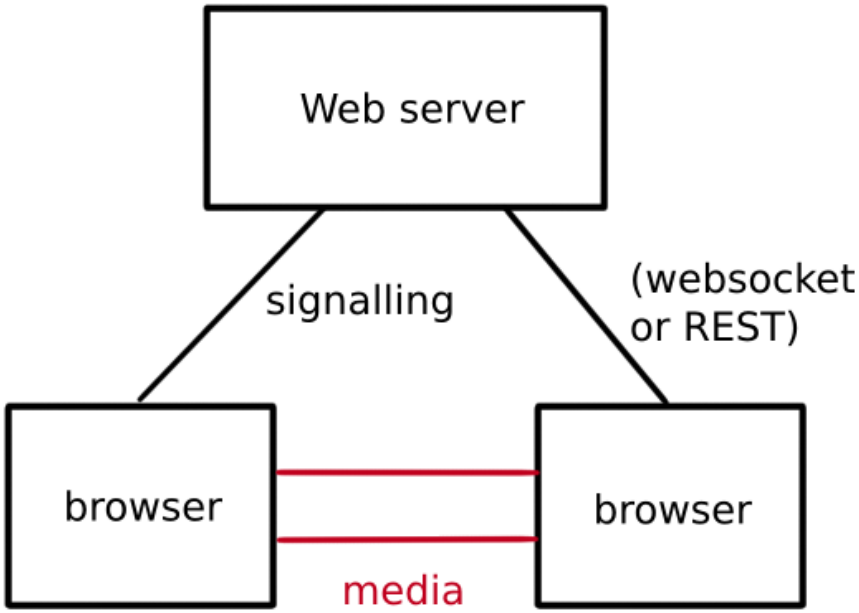
Videoconferencing made tolerable: WebRTC

At first sight, videoconferencing is difficult.

It has recently become tolerable: WebRTC:

- a complete videoconferencing stack;
- implemented in major browsers;
- finally agreed on common codecs:
  - everyone implements Opus and VP8 (even Apple!).

WebRTC: a peer-to-peer protocol



WebRTC implemented in the browser:

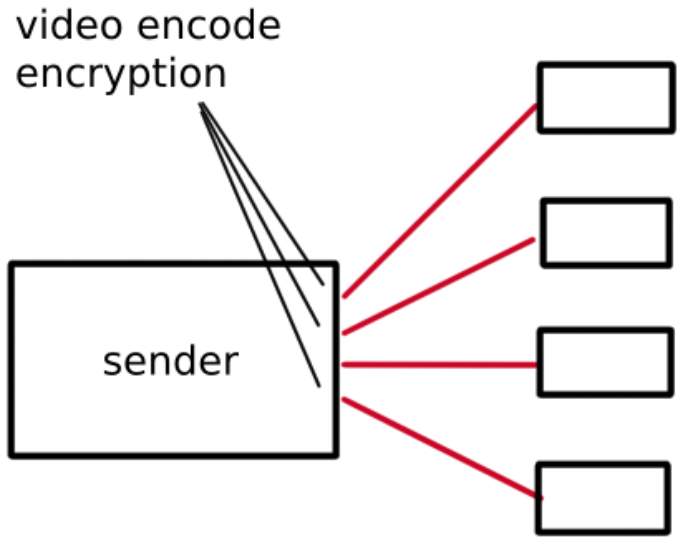
- client-server signalling (WebSocket, REST);
- media is peer-to-peer (RTP+RTCP+SRTP);
- optional peer-to-peer data (SCTP+DTLS).

The media traffic is encrypted end-to-end, keys negotiated over the signalling channel.

The JavaScript API is simple but inflexible (leading to “SDP munging”) (where art thou, ORTC?).

Peer-to-peer WebRTC doesn’t scale

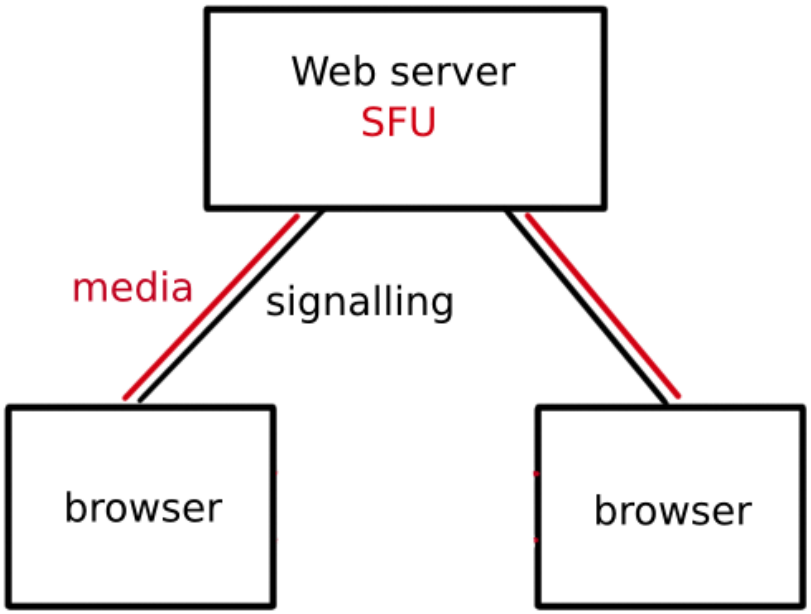
What happens if you try to broadcast over Webrtc ?



Every p2p flow is encoded, encrypted and sent separately.

Doesn’t scale beyond 4 or 5 peers.

Client-server WebRTC



The solution is client-server:

- client-server signalling;
- client-server media.

No need to reencode the media.

The server decrypts and reencrypts the video: there is no end-to-end confidentiality.

(Yes, I know about insertable streams.)

Digression: Pion

In the peer-to-peer case, WebRTC is in the browser.

Client-server, you need server-side WebRTC:

- RTP and RTCP;
- SRTP;
- SCTP;
- STUN, TURN, ICE…

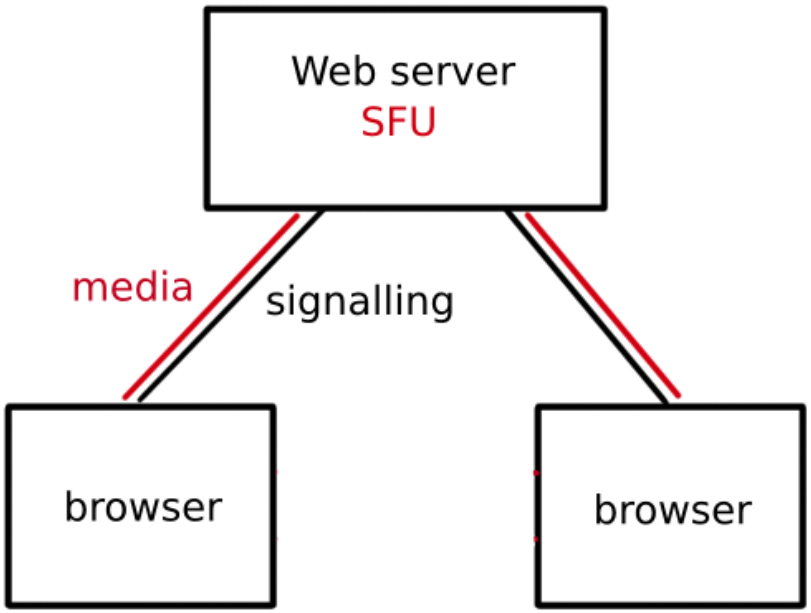
Pion is a Go implementation of WebRTC:

- pure Go (easy to cross-compile);
- lower layers fairly complete, upper layers in progress;
- reactive and friendly maintainer (Sean DuBois).

Galène uses Pion. Excellent experience.

Loss handling

Once you do client-server, where do you handle packet loss ?



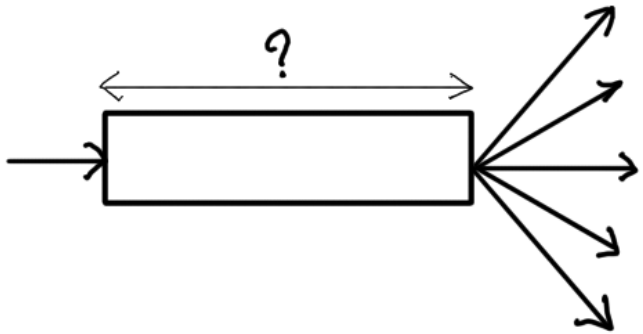
In Galène, we handle packet loss locally:

- reduces latency;
- requires buffering at the server.

This buffering does not cause bufferbloat:
packets are forwarded or dropped, never queued;
the buffer is only used to serve NACKs from the client.

Buffer management

Buffers in Galène : what size?



Packets are not queued, the size doesn't matter much:

- if too small, we won't be able to serve NACKs locally (we forward to the sender, increasing latency);
- if too large, we'll waste memory.

Currently sized proportionally to

$$\text{rate} \cdot (\text{maxRTT} + 4 \cdot \text{jitter}).$$

More experimentation is needed.

Congestion control

How fast can we send data over a given link ?
That's the problem of congestion control.

WebRTC doesn't define congestion control.
Browsers implement Google Congestion Control (GCC), which combines two congestion controllers:

- a traditional loss-based controller (useless in the presence of bufferbloat);
- a novel delay-based controller.

In Galène, we terminate congestion control at the SFU.
Galène acts as an application-layer proxy.

Congestion control (2)

We terminate congestion control at the SFU.

The resulting data rate is the minimum of the data rates acceptable for all clients:

- for small meetings, high rate ;
- during large lectures,
  the rate falls down to the minimum.

Potential solutions: simulcast or SVC.

Right now, congestion control in Galène is incomplete:

- complete in the server ⟶ client direction ;
- loss-based in the client ⟶ server direction.

Due to the prevalence of bufferbloated routers, this needs fixing.

Current status

Galène is good enough for lectures with 100 students:

- robust server (doesn't crash or deadlock);
- robust NAT traversal (many students are on 4G) (thanks to Pion and ICE);
- robust loss recovery.

Congestion control:

- state of the art in the server⟶client direction (loss- and delay-based);
- loss-based in the client⟶server direction (requires manual tweaking on bufferbloated networks).

Current status(2)

With a fascist firewall, Galène keeps trying:

- difficult to determine when to give up;
- UI issue: how to indicate that there is a problem?

Good video quality:

- NACKs served locally in a timely manner;
- PLIs aggregated and forwarded to the sender.

Audio quality issues:

- browsers don’t implement (enough) audio FEC
  please implement flexfec in the browsers!

Future plans

Improve the UI:

- Ready/Panic is not obvious;
- multiple layouts;
- contextual menus and mouse-over text;
- alternate frontends?

Vary quality per client:

- simulcast;
- scalable video coding (SVC).

Improve congestion control:

- many networks are bufferbloated!

Conclusion

Galène is a videoconferencing server:

- easy to deploy;
- easy to understand and improve;
- requires minimal server resources.

`https://galene.org`

Please install your own instance!