

GStreamer for audio distribution at Sveriges Radio

Karl Johannes Jondell

Christofer Bustad

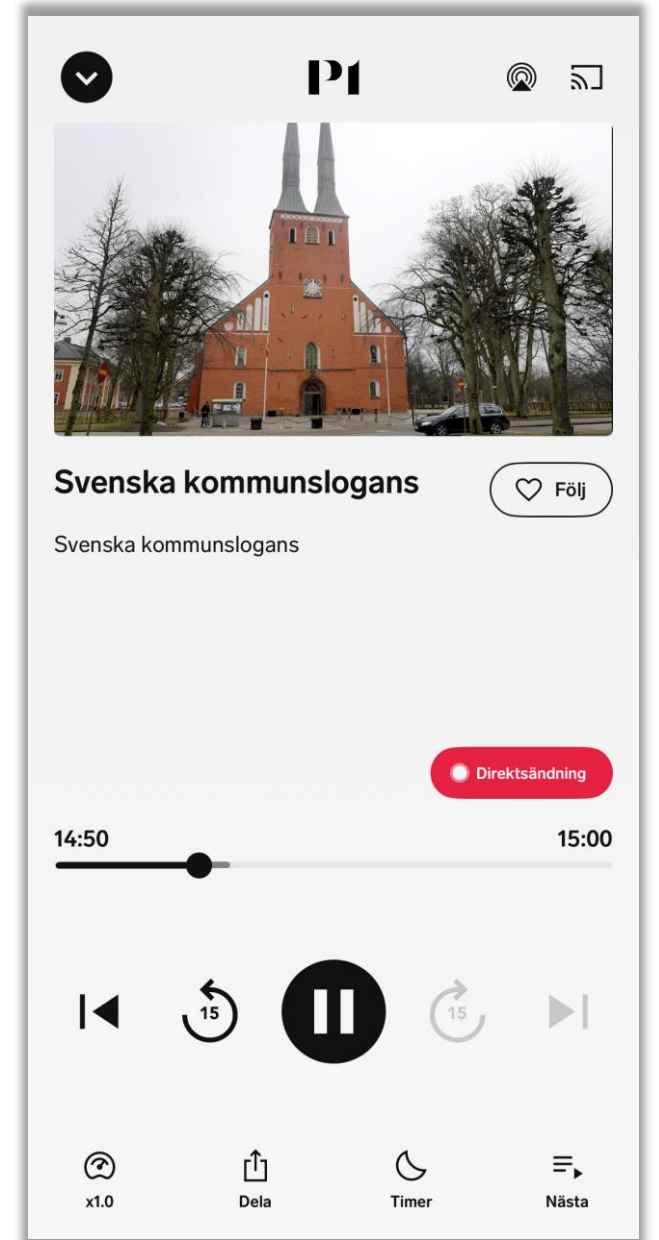


About Sveriges Radio (SR)

- SR is one of three public broadcasting companies in Sweden: SR, SVT, UR.
 - This year celebrating 100 years on the air.
- Our primary medium is radio: terrestrial FM & DAB, IP based-distribution.
- 7 million listeners every week. Sweden's population is 10.5 million.
- Radio channels: 4 national, 25 local, 7 special/niched channels (primarily digital-only).
- Public trust in SR is foundational and consistently measured high.
- Commercial and political independence.
- Mission to be innovative.
- Emergency preparedness mission.

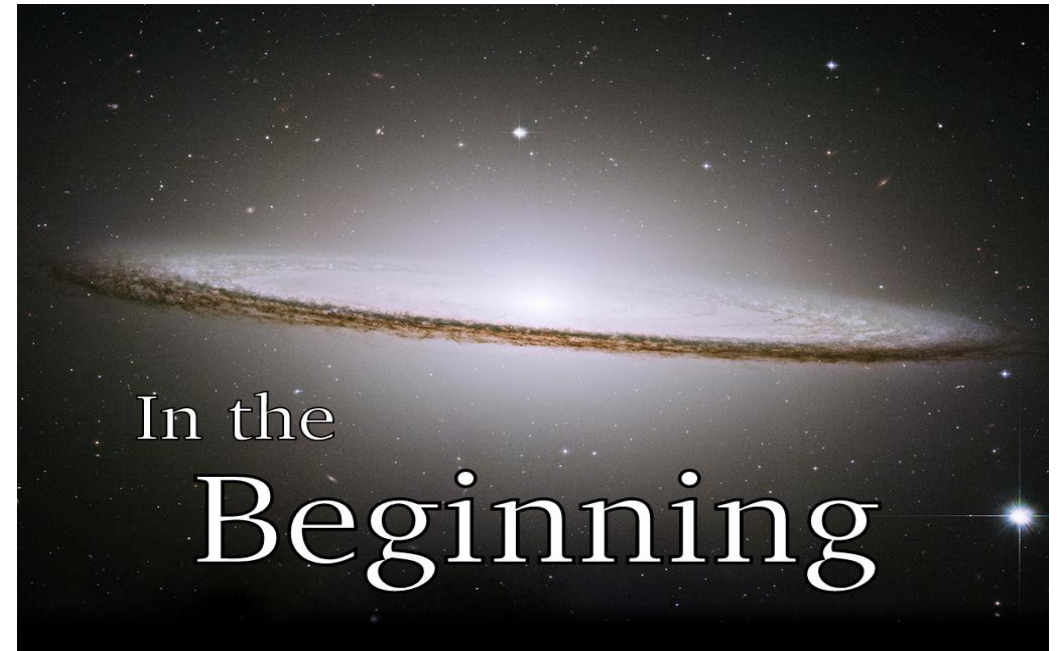
Our IP offerings

- Our platforms
 - iOS/Android app
 - sverigesradio.se
 - 3rd party platforms
- Linear
 - Using HLS for our own platforms & some 3rd party platforms.
 - Using ICY for backwards compatibility and most 3rd party platforms.
 - **HLS:** 52 live streams with 24/7 content encoded in: AAC
 - **ICY:** 37 live streams encoded in: AAC, mp3, FLAC, Opus
- On-demand
 - Podcasts, audio clips, catch up radio
 - 3rd party platforms
 - Formats used: AAC, mp3



In the beginning...

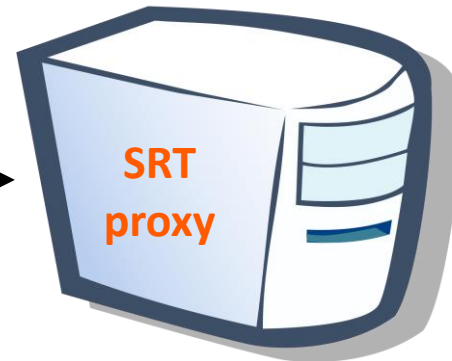
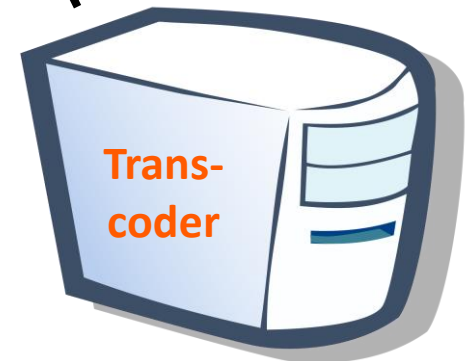
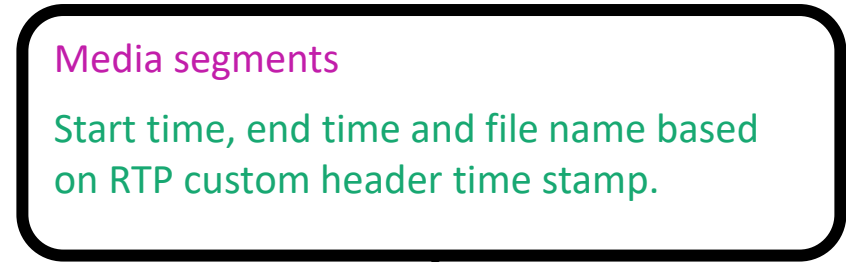
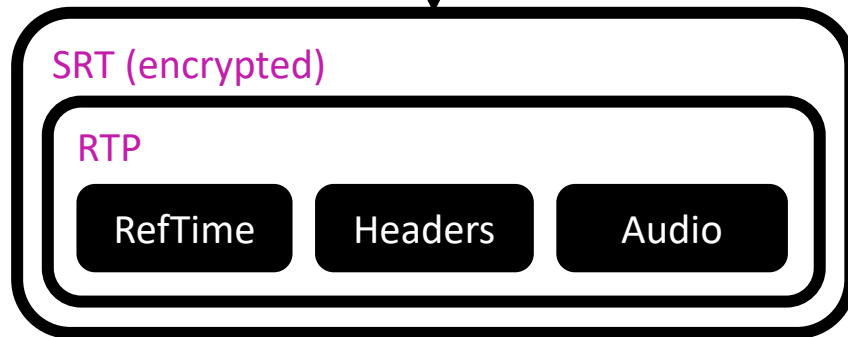
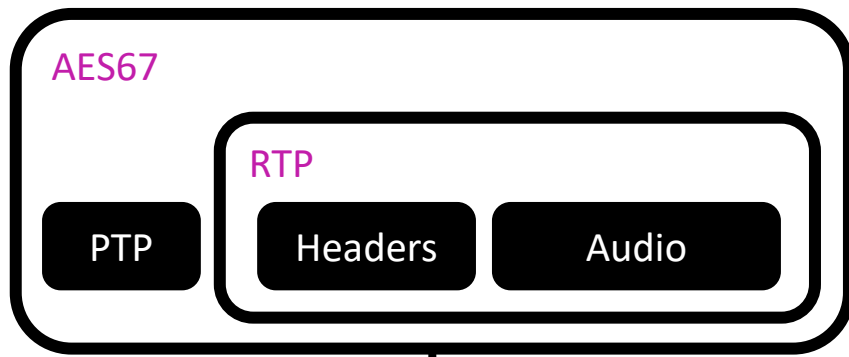
- There was a need for a renewed IP based distribution system.
- At the crossroads: modernising existing solution or building new from open software.
- Requirements
 - Containerised
 - Hybrid cloud
 - Redundancy
 - Reusability
 - Lower latency
 - Audio watermarking
 - Broadcast processing
- We decided to use GStreamer (and Rust)!



Pilot: HLS

- Containerised setup
- Three pilot channels
 - Knattekanalen, P4 Plus, P2
- HLS is a segmented media-based protocol.
- To achieve optimal redundancy, identical media segments produced independently by several transcoders are required.
- We decided to use GStreamer for transport, encoding, packaging and metadata, but not for HLS manifests, playlists, and peripheral services.
 - GStreamer application built in Rust for stability. Peripheral services built in Perl and Python.
- The foundation of the pilot was built by Centricular and is available at:
<https://github.com/centricular/aes67-relay-chunker>





Pilot: Lessons learnt

- Initially we had a gradual drift between our redundant pathways. The media segments created were identical, but the creation time diverged slowly.
- Large overhead in low bit rates. Large number of PES.
- LATM framing caused client playback issues for HE-AAC v2. We now ADTS use instead.
 - Bug in Chrome playing HE-AAC v2 with ADTS framing.
 - We only serve HE-AAC v2 to our iOS and Android apps.
- Other issues and shortcomings not directly related to media.

Pilot: ICY



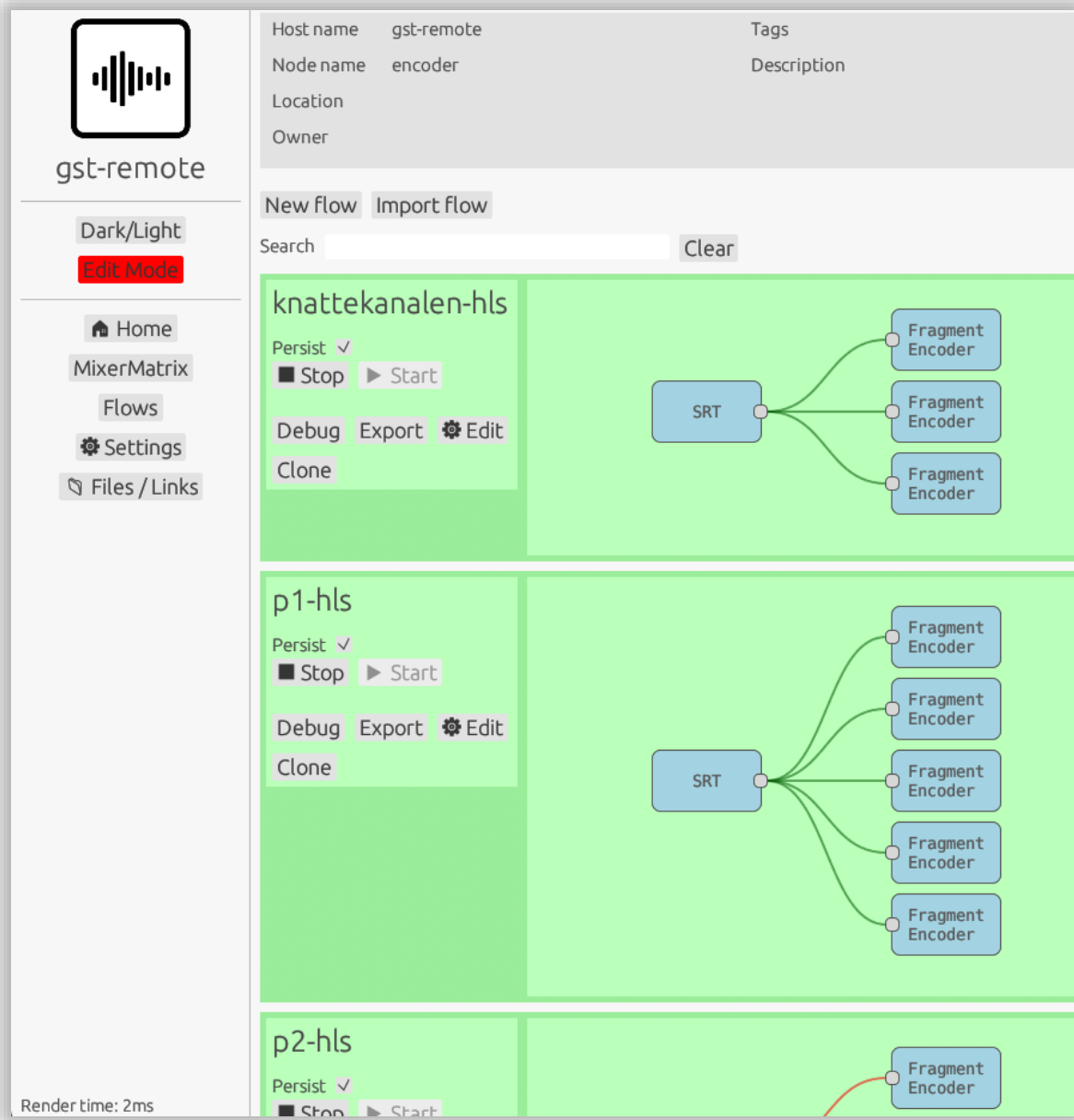
- ICY streaming protocol: Continuously growing file. Origins from WinAmp's SHOUTcast.
- We wanted to offer more formats than previously, more specifically the open formats FLAC and Opus.
- We have a long history and reputation of focusing on audio quality.
 - We both take pride in and enjoy fulfilling the high demands of our audiophilic audience to provide high quality distribution formats.
- For our music channel P2, we provide an ICY FLAC stream, that has been well received.
- Initially we used the shout2send plug-in, but now we use icecastsink (can be found in a branch on Tim-Philipp Müller's gst-plugin-rs fork: <https://gitlab.freedesktop.org/tpm/gst-plugins-rs/-/tree/icecastsink>).
- An ICY shortcoming: libvlc at 48 kHz sample rate.

Disaster Recovery

- A refinement of the pilot. Still hybrid-cloud. Redundancy completely independent of our main HLS distribution. Used if needed, for all linear channels.
- Used alongside both our old streaming platform and our current platform.
- No longer any need for an SRT proxy.

Our production systems

- A lot refinements based on the pilot, but the structure is basically the same.
- Completely on-prem.
- Refined pipelines
 - no dependency on PTP for media segment synchronisation
 - watermarking in custom GStreamer plug-in
- Running 52 channels on HLS and 37 channels on ICY (no pop-up channels)



gst-remote

- We needed an application to manage our GStreamer pipelines.
- Consists of a WebSocket API, reusable modular GStreamer components (bins), and a GUI. Written in Rust.
- We also use GStreamer for other applications at SR.
- gst-remote gives us a unified way of developing, managing and operating different products at the organisation.

gst-remote



gst-remote

Dark/Light

Edit Mode

Home

MixerMatrix

Flows

Settings

Files / Links

Host name conference-test

Node name conference-node

Location London

Owner Sveriges Radio

Controller Standalone

Tags conference, gstreamer, testing

Description A demo of gst-remote for the 2025 GStreamer Conference.

New flow Import flow

Search Clear

Name test-flow

Description A test flow with WebRTC and RTSP output.

Flow ID d3e96596912e86ba

Min Latency 21 ms

Persist ☒

Save Cancel

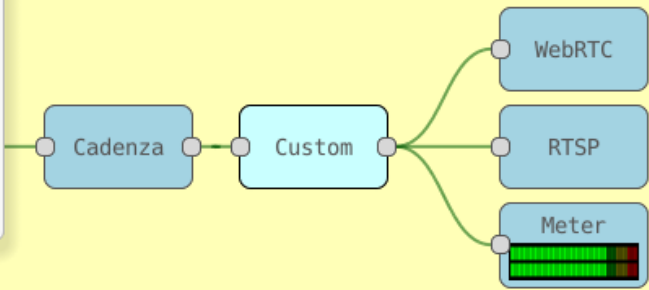
Delete

Source

- AES67
- SRT
- JACK
- RTSP
- AudioTest
- WASAPI
- WebRTC
- ASIO

Filter/Process

Destination



>>

Delete block

Custom Block settings

Block ID 51e158db42294dfa

Graph

```
queue name=queue_0 ! tee name=tee_0
```

GStreamer plug-ins

- Our model/starting point for developing plug-ins was:
<https://gitlab.freedesktop.org/gstreamer/gst-plugins-rs/-/tree/main/audio/csound>
- Our first plug-in was audio watermarking
 - All our linear audio is watermarked regardless of distribution.
 - Channel specific code & time stamp
 - For audience measurements
 - Perceptually transparent
 - Should be late in the signal chain, but might cause extra peaks in the waveform

GStreamer plug-ins

- Currently working on a plug-in for broadcast audio processing.
- What is broadcast audio processing?
 - Audio processing, after the radio production, but before the audio reaches listeners.
 - Similar to mastering.
 - EQ, AGC, compression, limiting, stereo widening, multi-/single band, etc
- Why do broadcast audio processing?
 - To adapt the audio for a technical platform.
 - Loudness, peak levels, SNR, audio bandwidth, etc
 - Regulatory requirements & best listener experience within the technical platform.
 - To adapt the audio for the typical listener & listening environment.
 - To achieve a certain “sound”: Listeners will recognise the radio channel, and for aesthetics.
 - To adjust any mistakes from earlier in the signal chain: levels, frequency balance, etc

GStreamer plug-ins

- Audio quality has always been a top priority at SR: Listening tests, broadcast processing, etc
- 25 years ago, our colleague Torbjörn Wallentinus developed the ideas of an advanced one-band broadcast processor in collaboration with the company Factum (now Factum Radioscape)
→ The Cadenza was born!
- The hardware units has been working great for many years.
- SR recently bought a software version of the Cadenza, including the source code.
- Working towards a pilot for FM in the near future.



Wrapping up...

- Working towards getting gst-remote and our plug-ins open source to the public.
- Streaming platform foundation:
<https://github.com/centricular/aes67-relay-chunker>
- Questions?