

The Tandberg-GStreamer Connection



Or how we stopped worrying, and started using GStreamer



Us:

- Håvard Graff (hgr)
- Ole André Vadla Ravnås (oleavr)

The beginning

- TANDBERGs first PC-client
 - A bought software-project
 - Mainly for driving sales of infrastructure
 - Based on ActiveX and JavaScript (!)
 - Limited video quality, H.263 only
 - Hopeless to maintain, impossible to advance

The birth of Movi 2.0

- Proprietary code versus Open Source
- A few people knew of GStreamer, and showed it off
- Quickly getting popular with developers, management no-so-much
- Lots of fighting!

We won!

- Decided to use GStreamer
- GObject for application code
 - PIDL
 - (Vala)



Movi

Business quality HD Video on your PC or Mac...

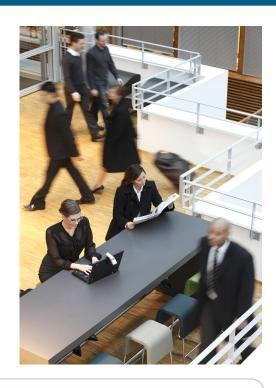
Anywhere. Anytime.



- Superior business quality video and audio communications.
- Connect to any standards based video system from anywhere

Scalable & Interoperable

- PC or Mac video capable of scaling to the tens of thousands.
- Seamless functionality with the Total TelePresence Solution



Easy, Secure, Managed

- Easy to use with little or no training or direction.
- Secure and standards based communications
- Centrally managed solution.

Movi Key Features and Benefits

Superior Quality

- Business quality HD video (up to 720p 30 frames-per -second) on the desktop
- PC or Mac Video Industry's best audio performance (MPEG4 AAC-LD, G.722.1/G.711)
- Acoustic echo cancellation
- Rich presence awareness so you know who's available

Scalable & Interoperable

- Capability to deploy to thousands of users
- SIP Registrar (VCS) and Provisioning (TMS)
- Interoperability with any standards-based SIP endpoint and H.323 systems



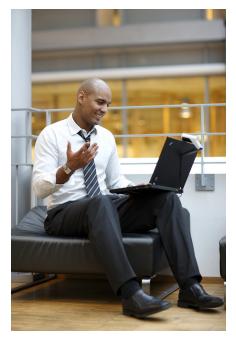
Easy, Secure, Managed

- Enterprise Ready
- Simple installation and management via TMS
- Firewall friendly
- AES and TLS Encryption

Mobile Collaboration with Movi



Extends video communications to the PC or Mac supporting work/life balance programs and green initiatives Remain connected when traveling with high quality and dependable communications



Build more effective teams across corporate boundaries



Brings speed and precision to any organization

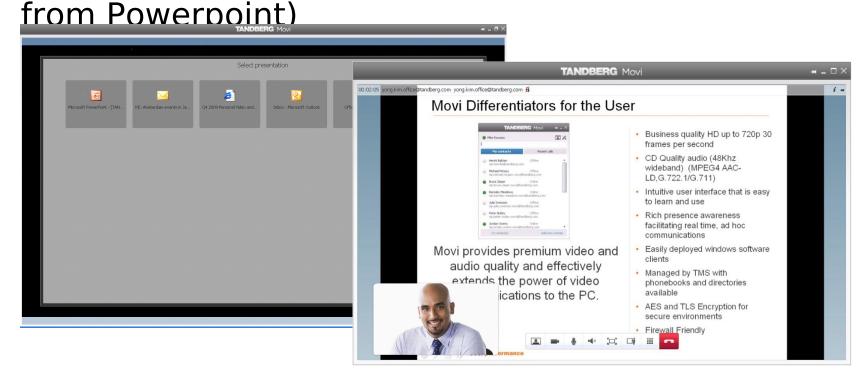
Increase productivity



Collaborate and share

Select application and share content from the PC or Mac with any other standards compliant devices

Optimized for the most common use case (share



Cisco PrecisionHD USB

- High quality optics
- Glass-only lenses
- Large aperture (F1.7)
- Large sensor → Low noise
- Capable of 720p 30f
- Wideband omnidirectional microphone



Movi for the PC or Mac



Details about Movi

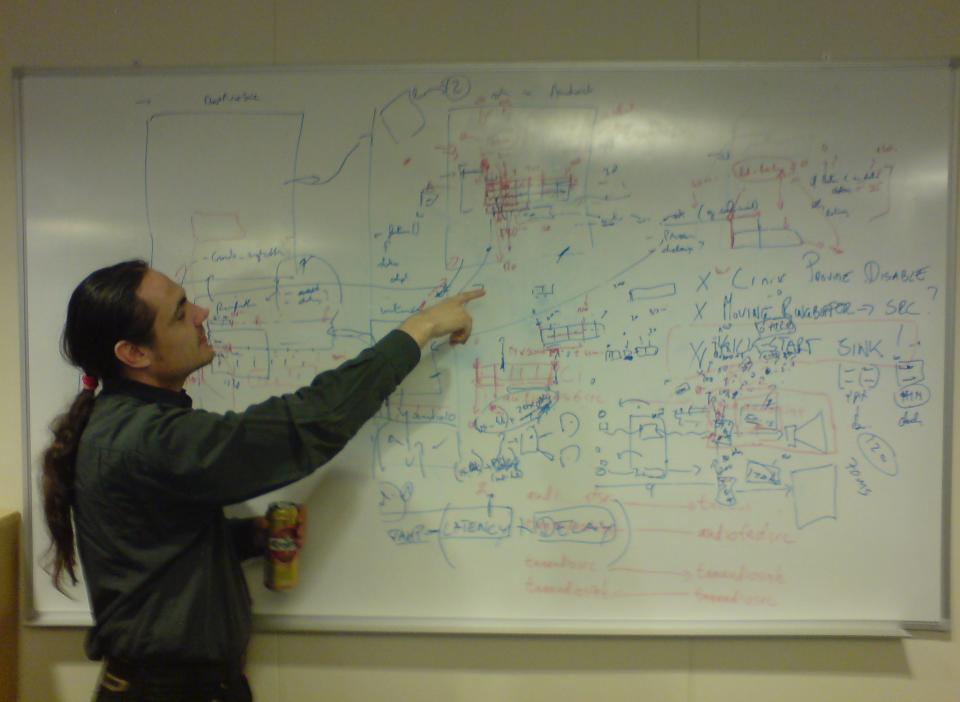
- A lot of Intel Optimization
 - Among the fastest MJPEG sw decoders in the world
 - Fractal in-house developed and optimized H.264 codec
 - Does HD videoconferencing on a standard dual-core PC (competitors require quad core)
 - Can run 720p ~25fps on a single-core Atom!
- On the Business Side
 - Movi is used to sell infrastructure (call control, conferencing etc)
 - Shipped about 150 000 licenses
 - Selling more Movi licenses than appliance-based endpoints, effectively doubling demand for network infrastructure products

The future...

- CPVE in CSF
 - Cisco Precision Video Engine based on the Movi media engine
 - CSF Client Services Framework enabling soft clients with common services for network, call control and media
- Potentially GStreamer running on millions of Cisco-clients

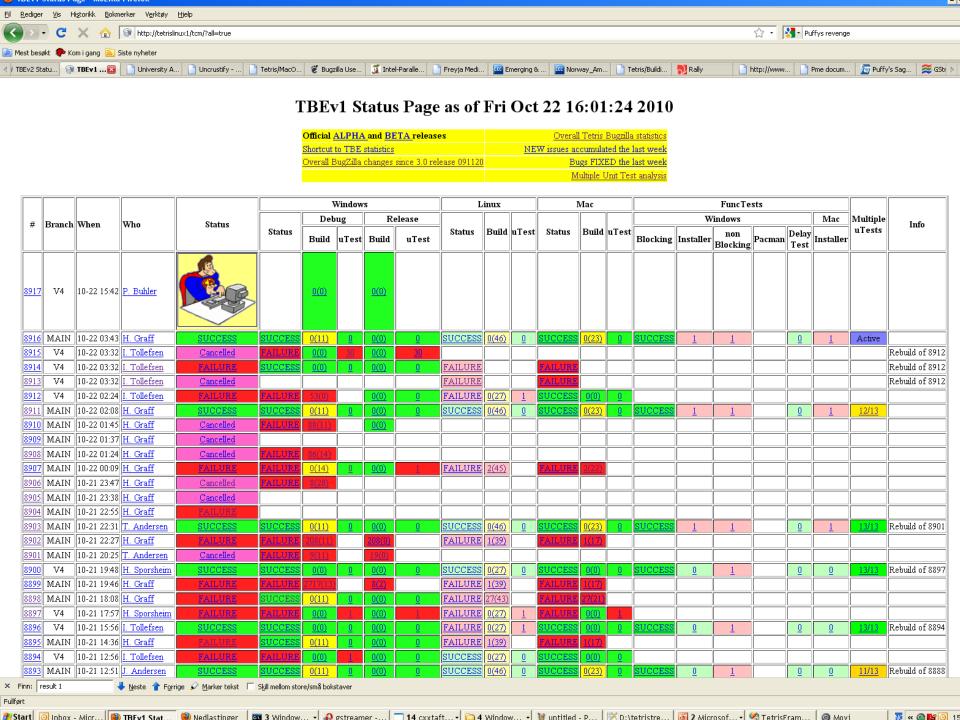
The GStreamer symbiosis

- Quite high threshold to go past simple playback-cases
- Live pipelines poses many challenges
- We have invested a lot of time and effort into GStreamer
 - Directly related to how successful it has been
- Give and take
 - Communication (#IRC)
 - Patches...
 - Memory-leaks
 - Tricky race-conditions
 - New functionality
 - Windows-stuff
 - Sponsored development
 - Dynamic BaseTransform
 - Thread priority in tasks
- Belgian Support ™



TBE

- Our buildsystem
- Windows, Linux and OSX
- Continuous integration
- Runs roughly 12.000 tests related to GStreamer for every build
 - Reveals race-conditions
- Memory-leaks not allowed!

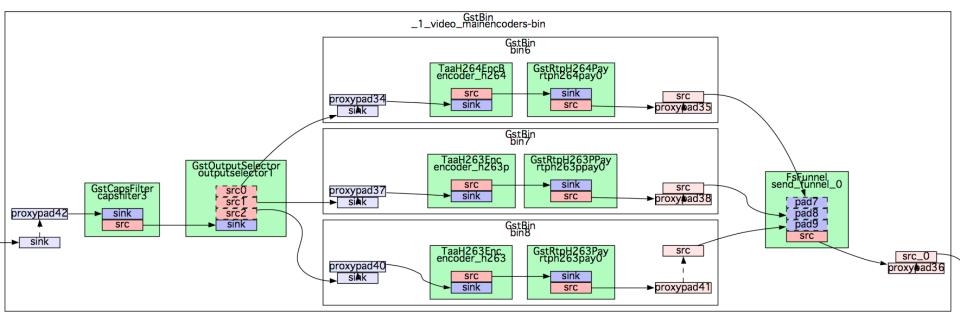


Our pipeline

- a mix of Open Source and proprietary

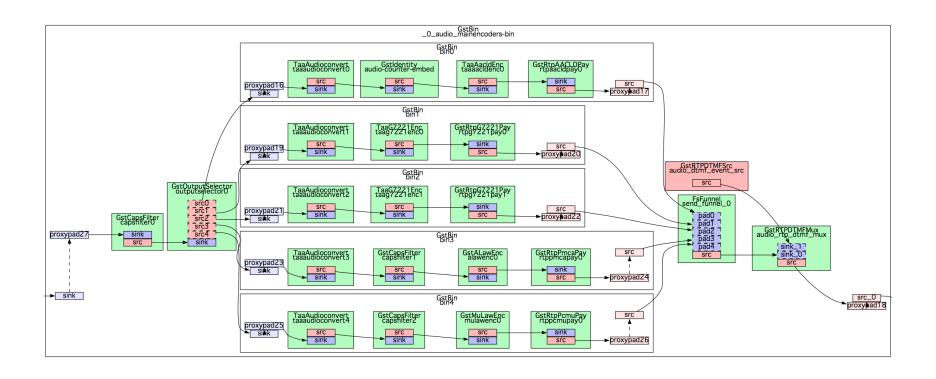
- Proprietary:
 - H.264, H.263, H.263+
 - AAC-LD
 - G.722.1
 - JPEG decoder
 - SRTP (Secure RTP)
 - Colorspace converter
 - Forward Error Correction
 - Far End Camera Control (H.224)
 - Audio packetloss concealer
 - Audio sink and src
- Open source
 - RTP (payloading, rtpbin)
 - Pipe-fittings: Tee / OutputSelector / Funnel (farsight)
 - Capsfilter
 - Level, Volume
 - Queue
 - Fakesink, Appsink, Cluttersink
 - Videosources for Windows and OS X (made by us!)

Encoder Bin (Video)



- Capsfilter forces a standard videoformat
 - YUV 1420
 - Used by all our video encoders
- OutputSelector used for selecting encoder
- Funnel in the opposite end

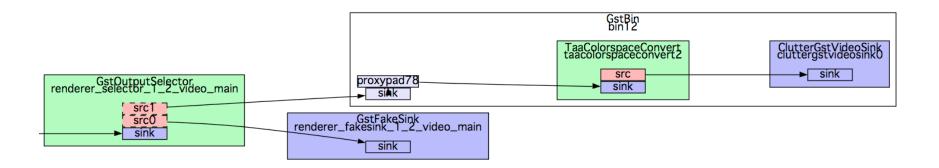
Encoder Bin (Audio)



Dynamic Linking Sinks

- Several Problems
 - The ClutterSink needs to be created inside the GLContext
 - Dangerous to do this synchronously
 - Deadlocks
 - Asynchronously
 - Problem with flow-not-linked
 - The sink needs to be removed dynamically as well
 - Avoiding flow-not-linked
 - Blocking the pad is problematic:
 - Unsafe to unlink until the callback comes
 - No buffer no callback

Our solution



- Linking
 - The OutputSelector start being linked to the fakesink
 - Once the sink arrives, it is linked up, and swapped in
- Unlinking
 - A probe is attached to the src-pad of the selector
 - A custom event is sent to the sink-pad of the selector
 - In the callback of that probe, the sink is unlinked and the probe removed

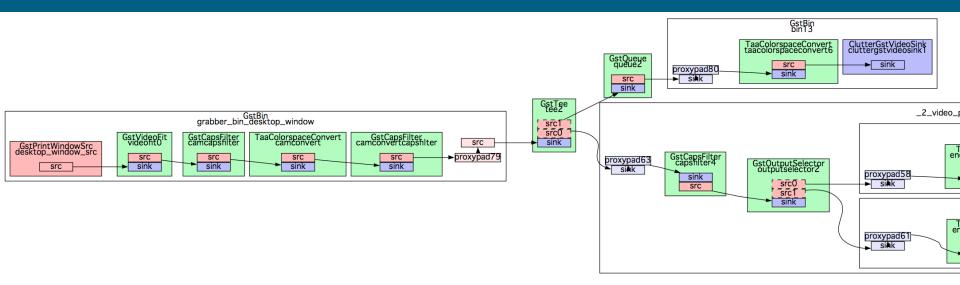
Sources in live pipelines

Sources send events:

- When starting
 - Newsegment
- When stopping
 - End Of Stream

We don't want either!

- Newsegment
 - A source being started in a running pipeline will emit a newsegment with start 0. (Baseclass functionality, not configurable (yet...))
 - However, the buffer timestamps needs to be starting from running_time, not 0, due to lipsync with other media. (the horrors of ts_offset...)
 - The newsegment will be appended in the sink, effectively delaying all buffers with running_time.
 - Our problem: After starting self-view, application would lock up for the length of the conversation up to that point!
- End Of Stream
 - Would effectively stop all elements upstream. (rtp send...)
 - Muting video, we unlink the src (free up the resources)
 - Unmuting would not work...



Our solution

- Attaching a probe, removing all Newsegments and EOS!
- We have yet to miss them...

```
static void
attach_event_dropper_on_tee (GstElement * camera_tee)
{
    GstPad * pad_for_event_dropper = gst_element_get_pad (camera_tee, "sink");
    gst_pad_add_event_probe (pad_for_event_dropper, (GCallback)event_dropper, NULL);
    gst_object_unref (pad_for_event_dropper);
}

static gboolean
event_dropper (GstPad * pad, GstMiniObject * mini_obj, gpointer user_data)
{
    GstEvent * event = GST_EVENT (mini_obj);
    int event_type = GST_EVENT_TYPE (event);

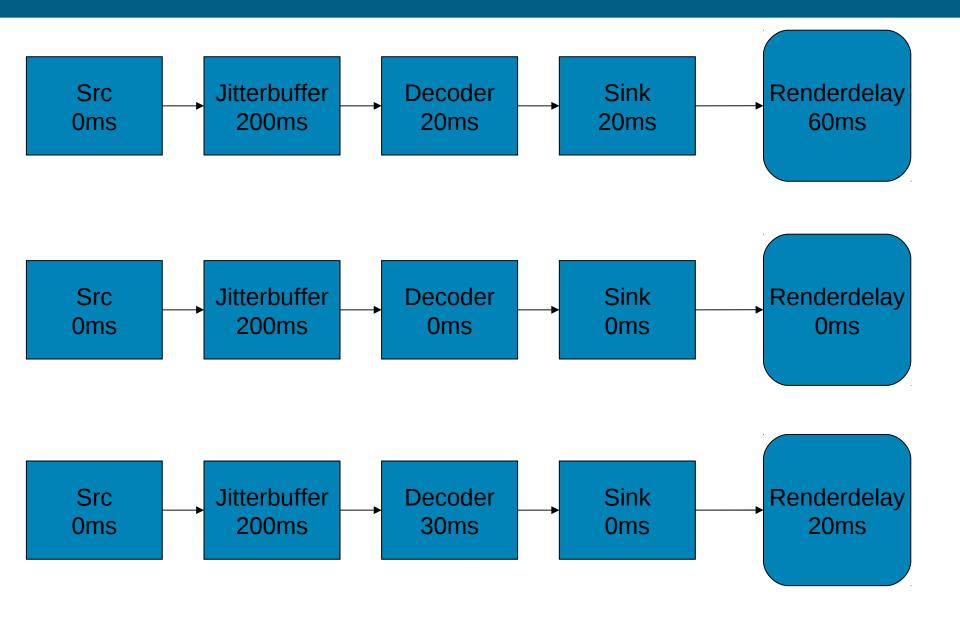
    switch (event_type)
{
        case GST_EVENT_EOS:
        case GST_EVENT_NEWSEGMENT:
        // drop
        return FALSE;
}
return TRUE;
}
```

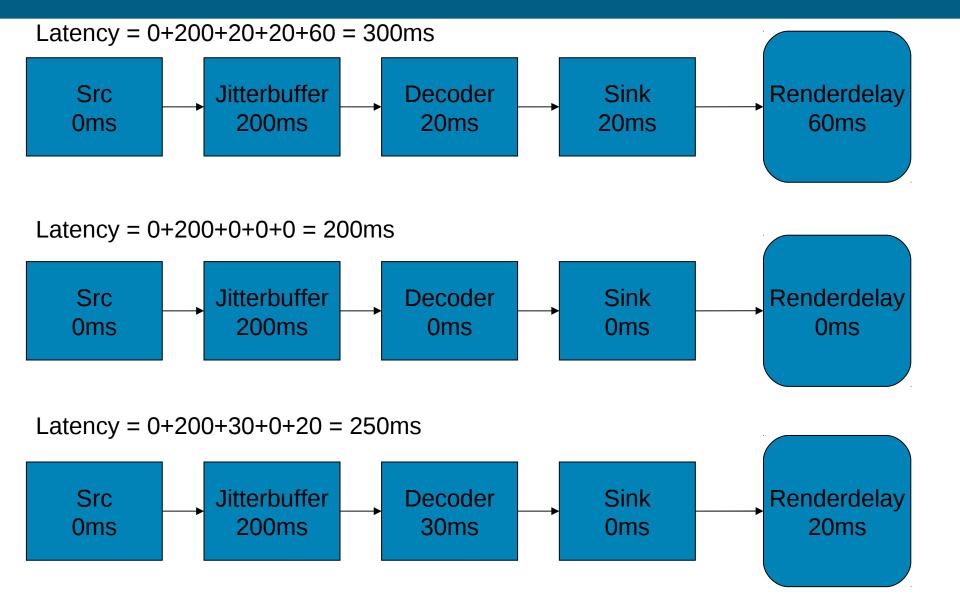
Renderdelay

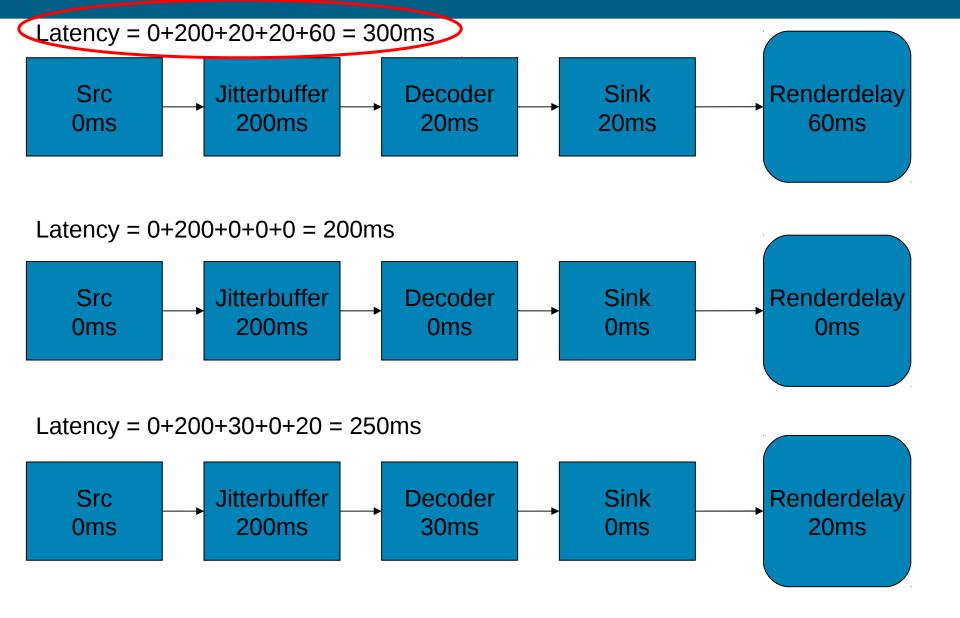
- To be able to compensate for known delays after the buffer has left the sink.
- In our case, ex. DirectSound buffer headroom and processing delay.
- Description:
 - Set on the sink. (basesink)
 - Adds the configured renderdelay as extra reported latency.
 - Then subtracts its own renderdelay after latency has been added.

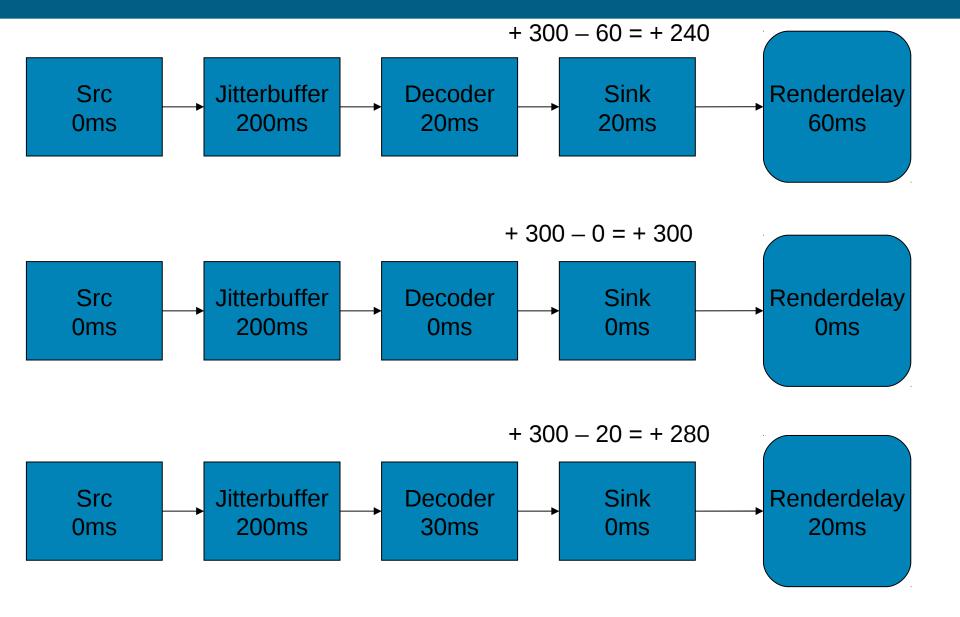
?!?!?

- What is the point then?
 - No point if there is only one stream.
 - For 2 or more streams, you could have something like this:









Locked State

- Deciding when elements change state
- Crucial to dynamic pipelinebuilding due to a shortcoming of state-changes (is it fixed?)
- We use locked-state for all elements that are dynamically added or removed from the main pipeline.

Both elements in PLAYING

GstBin:

State: PLAYING

Pending: -

Target: -

GstElement:

State: PLAYING

Pending: -

Target: -

- PLAYING
- PAUSED
- READY
- NULL

Setting the child to NULL

GstBin:

State: PLAYING

Pending: -

Target: -

GstElement:

State: PLAYING Pending: PAUSED

Target: NULL

- PLAYING
- PAUSED
- READY
- NULL

Parent is set to NULL

GstBin:

State: PLAYING

Pending: PAUSED

Target: NULL

GstElement:

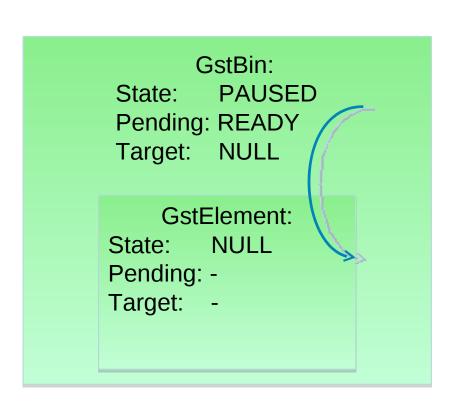
State: NULL

Pending: -

Target: -

- PLAYING
- PAUSED
- READY
- NULL

next is PAUSED, which is set on the children



- PLAYING
- PAUSED
- READY
- NULL

Effectively bringing the element back up from NULL

GstBin:

State: READY

Pending: NULL Target: NULL

GstElement:

State: NULL
Pending READY
Target: PAUSED

- PLAYING
- PAUSED
- READY
- NULL

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Bug List: (1 of 2) First Last Prev Next Show last search results

Bug 594248 - Use locked-state on internal rtp-bin to avoid shutdown-state-race (edit)

Collapse All Comments - Expand All Comments

Håvard Graff (hgr) [reporter] 2009-09-05 18:24:53 UTC Description [reply] [-]

Created an attachment (id=142546) [details] [review] patch

When rtpbin is going from PLAYING to NULL, it will set state on its internal bins (session and demux). If those internal bins are already in NULL as a result of freeing the request-pads, they will be brought back up again.

This can easily be solved by setting locked state on the bins prior to setting them to NULL.

Wim Taymans [GStreamer developer] 2009-09-08 10:43:48 UTC Comment 1 [reply] [-]

commit e08e610db0d5e9f7e4d5d5c42b026b2853b1321d
Author: Havard Graff < havard.graff@tandberg.com>
Date: Mon Aug 31 18:46:51 2009 +0200

rtpbin: use locked state on internal bins

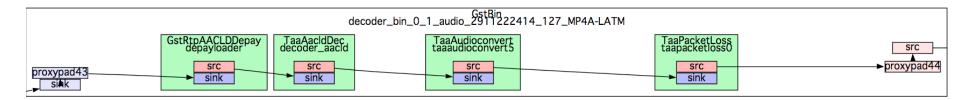
Set the locked state on internal elements to make sure that they don't change

back to another state when shutting down.

Fixes #594248

Additional Comments:

The lost-event

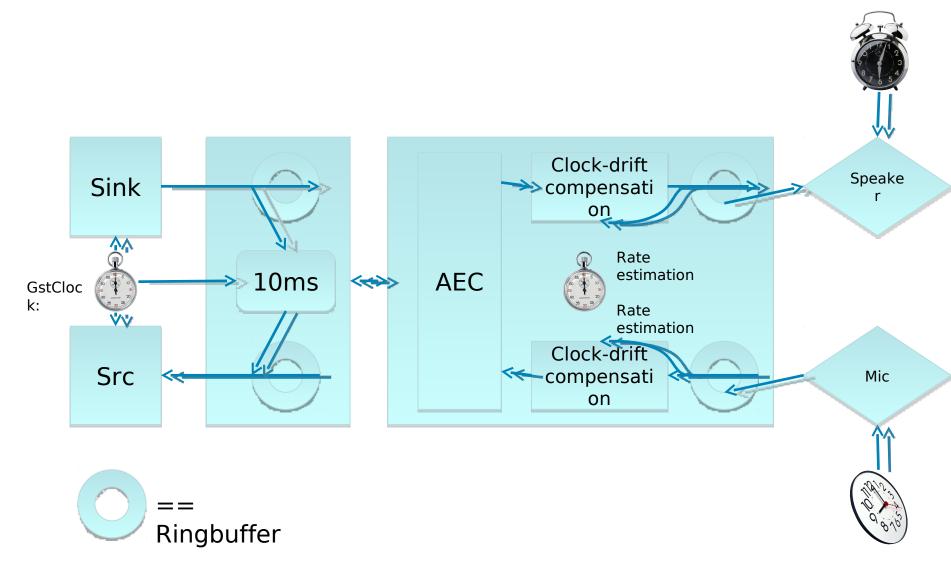


- Emitted from the GstRtpJitterBuffer
- Will wait for a packet until its last possible time to be played out expires
- We use the lost-event for:
 - Audio packetloss-concealment
 - Video decoder error-reporting
 - PLI / FUR
- FEC sits upstream from the jitterbuffer, reconstructing some lost packets

GstTestClock

- You have to manually drive the clock, as well as releasing the waits
- Make deterministic synchronization tests
- What really goes on inside a live pipeline?
- Done some work here, a lot more to come
 - Lipsync, latency etc...
- Not yet upstream, coming soon!

The Audio System



Acoustic Echo Canceller

- Gives -35dB
- Works with all known sound devices, on Linux, OSX and Windows
- Double-talk! (not suppression)
- Several TANDBERG patents involved
- Same code that runs on the endpoints

One more thing...



THANK YOU! (questions?)



See you on #gstreamer ©

