



# Time and Synchronization for dummies

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# Goals

- Feel comfortable with timing and synchronization in GStreamer
- From real-life examples ...
- .. to what GStreamer can do ...
- .. and understand why



# Time

- Ordering (past, present, future) of events
- Measure of duration of events and intervals between them
- “a certain number of repetitions of one or another standard cyclical event ... constitutes one standard unit of time such as the second”



# Chrono-meter (clocks)

- Sundials, water clocks (clepsydra)
- Hourglass (Magellan) and candles
- Mechanical clock
- Quartz and atomic (caesium) clocks



# Time

- Clocks have different rates and precision
- Clocks measure the passing of time (duration, intervals)
- The absolute time is not useful
  - *Meeting at 232895437843294 !*
- You need a reference
  - 01/10/2013 00:00 UTC is 332895437840000
  - Meeting in 10mins from “now”



# GstClock

- API and base implementation
- Monotonic rate
- Get current clock ***absolute time***
  - `gst_clock_get_time()`
- Schedule event for absolute time
  - `gst_clock_id_wait()/wait_async()`
- Works without a pipeline





# GstClock

- Different implementations
- Doesn't matter for rest of talk
- Assume it's the monotonic POSIX clock





Clock  
Absolute  
Time





# Buffer timestamps

- videotestsrc ! timeoverlay ! xvimagesink sync=False
- Timestamps on buffers produced by videotestsrc
- 0, 1/30s, 2/30s, 3/30s, ....



# Buffer timestamps

- Argh, everything goes too fast !
- Without synchronization, it's like unix shell piping (+/-)
- But the buffers had timestamps !
- Synchronization, it's useful (c) (tm)
- How are we going to synchronize against a clock ?



# Running time

- We want buffers to be synchronized N seconds after we start playing
- Moment the pipeline switches to playing (***base time***)
  - + N seconds, AKA : ***Clock running time***



0

Running Time

Base Time

Clock Absolute Time



# Segment

- How do we figure out the running time for buffers ?
  - Take the absolute value ? What if the first buffer PTS is not 0 ?
- We need a reference (from which to calculate the running time of each buffer)
- Enter GstSegment !



# Segment

- Helps define the various time relationships in a stream
- ***start, stop***: first and last valid buffer timestamp
- For any buffer:
  - $\text{PTS} - \text{Segment.start} \Rightarrow$  buffer running time
  - (not final formula)





**Segment.start**

**Buffer.PTS**

**Segment.stop**

Buffer  
Timestamps

0

Running  
Time

**Base Time**

Clock  
Absolute  
Time



# Base Time

- What if I pause ?
- Running time is the amount of time “spent” in PLAYING
- Need to update ***base\_time***
- PLAYING=>PAUSED : remember running\_time
- PAUSED=>PLAYING :  $base\_time = current\_absolute\_time - running\_time.$



# Segment rate

- What if I play faster/slower ?
- I want buffers to be synchronized faster/slower
- Segment **rate** property
- running\_time gets adjusted accordingly
- $(B.PTS - S.start) / ABS(S.rate)$



Segment.rate > 1.0

*Segment.start*

*Buffer.PTS*

*Segment.stop*

Buffer  
Timestamps

0

Running  
Time

*Base Time*

Clock  
Absolute  
Time



# Segment rate

- What if I play backwards (in reverse) ?
- `Segment.rate < 0.0`
- Buffer have decreasing timestamps
- Running time is calculated using `Segment.stop`
  - $(S.stop - B.PTS) / ABS(S.rate)$



Segment.rate < 0.0

**Segment.start**

**Buffer.PTS**

**Segment.stop**

Buffer  
Timestamps

0

Running  
Time

**Base Time**

Clock  
Absolute  
Time





# Stream time

- “User-facing time”
  - Position reporting
  - Seek values
- Quite confusing since it's quite often the same as buffer time.
- When isn't it the same ?
  - RTP use-cases
  - DVB use-cases
  - Some formats



# Stream Time

- You connect to a live presentation via RTP which started 30mins before
- RTP timestamps (i.e. Buffer timestamps) can be anything
- You want to be shown how much in the presentation you are
- => Stream Time
- **Segment.time** (reference for stream time)



**Segment.time**

**Segment.start**

**Buffer.PTS**

**Segment.stop**

Stream Times

Buffer Timestamps

Running Time

**Base Time**

Clock Absolute Time



# So far....

- Absolute Clock Time
- Running Time
- Buffer Time
- Stream Time
- Need base\_time and segment



# Live sources/pipelines

- Time and Clocks are not just used for synchronizing buffers/events,
- Also used for knowing when an event happened.
- Live sources (webcam, microphone, ...)



# Live sources/pipelines

- A “live” event is an event that happens “now”
- If you try to capture too early/late you will miss it
- “now” is the current running time of the clock.





Audio Capture Delay

Video Capture Delay

0

Running Time

Real World Event

Clock Absolute Time



# Live sources/pipelines

- The same event captured over different sources should have the same timestamp
- But we have different capture time (1 audio segment duration vs 1 webcam frame duration)
- So we just subtract that value from the current running time ?



Audio Capture Delay

Video Capture Delay

0

Buffer Time

Clock Absolute Time

Real World Event



# Live sources/pipelines

- Subtracting capture delay from running-time helps ...
- But would result in all buffers always arriving late (if you wanted to play them back in the same pipeline)
- Enter latency !



# Latency

- Ensure buffers/events will be able to be synchronized downstream (i.e. Not dropped)
- As quickly as possible
- Not too early and not too late (grmbl !!!)
- How do we figure that

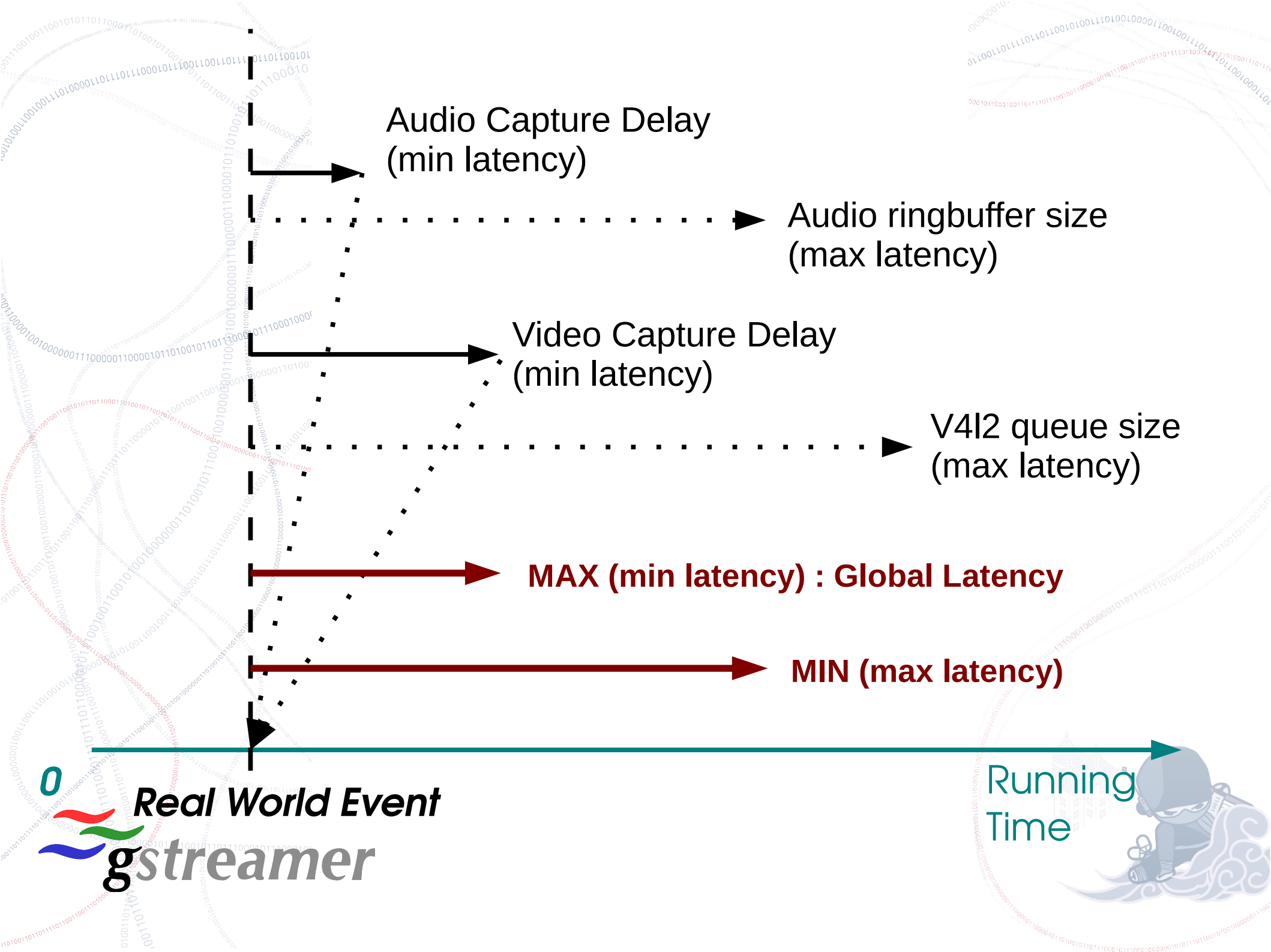


# Latency

- Let every element in the pipeline report what
  - is the minimum latency it is introducing (for producing/processing data)
  - is the maximum latency it can support (before dropping/blocking)
- GST\_QUERY\_LATENCY
- Pipeline emits and distributes ideal latency







Audio Capture Delay  
(min latency)

Audio ringbuffer size  
(max latency)

Video Capture Delay  
(min latency)

V4L2 queue size  
(max latency)

**MAX (min latency) : Global Latency**

**MIN (max latency)**



# Latency

- Rendering time becomes:
- Latency + running\_time



Audio Capture Delay  
(min latency)

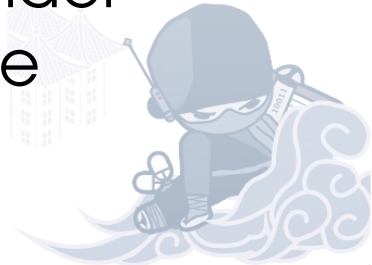
Video Capture Delay  
(min latency)

Running  
Time

Pipeline Latency

Render  
Time

Real World Event



# Latency

- Other elements can introduce latency
  - Decoders (frame reordering)
  - Transformation elements
  - .....
- Or increase max-latency
  - Queue !



# No more time !

- Different clocks
- Slaving clocks and distributed synchronization
- Advanced techniques
- Go see Jan's talk
- You have the basics !



# Time for questions ?

- Or time for lunch ?

