

Timing issues in desktop audio playback infrastructure

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About myself

- ▶ I am not working for any audio or open-source company
- ▶ I have submitted some PulseAudio patches
- ▶ I wrote dcaenc
- ▶ I added a high-quality resampler to Wine

Primary references

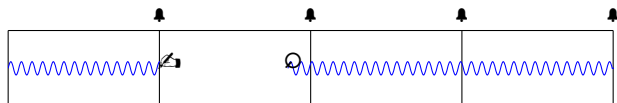
- ▶ <http://0pointer.de/blog/projects/pulse-glitch-free.html>
- ▶ <https://wiki.freedesktop.org/www/Software/PulseAudio/Backends/ALSA/Issues/>

ALSA architecture

- ▶ Raw hardware (hw:) devices
- ▶ Plugins
 - ▶ resampling, format conversion, channel remapping
 - ▶ volume attenuation, mixing
 - ▶ output to pulse, cras, ...
- ▶ Common API
- ▶ `.asoundrc` to glue pcm names with plugin chains

Traditional scheduling

- ▶ Buffer, divided into periods
- ▶ Sound card tells the kernel when a period elapses
- ▶ One period = one application wakeup



- Hardware Pointer
- 🔔 Wakeup Position
- ➡ Application Pointer

Latency Requirements

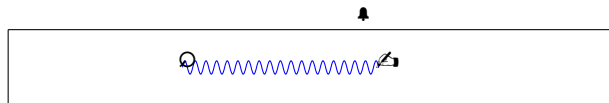
- ▶ Latency = buffer size
- ▶ Wakeup interval = period size
- ▶ Too much latency is bad for games and VoIP
- ▶ Low latency \Rightarrow more dropouts
- ▶ Too low wakeup interval eats battery

Conflict!

- ▶ Consider mixing with dmix
 - ▶ Period size is common
 - ▶ Period size is not reconfigurable at runtime
 - ▶ \Rightarrow Fixed low wakeup interval for the worst case

Timer-based Scheduling

- ▶ Soundcard interrupt period is not reconfigurable ☹️
- ▶ We can use a timer instead 😊



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Loop

- ▶ Query application & hardware pointer difference
- ▶ Write sound data
 - ▶ low latency \Rightarrow just some data
 - ▶ high latency \Rightarrow a LOT of data
- ▶ Schedule a timer that fires just before it plays out
- ▶ Sleep

Implementations

- ▶ PulseAudio
- ▶ CRAS

We've got

Dynamic latency 😊

We've got

Corner cases 😞

On stream start

- ▶ To process (resample, mix, encode): 2000 ms of sound
- ▶ Budget: 200 ms of real time (due to rtkit)
- ▶ Not easy:
 - ▶ On a weak CPU (ARM), or
 - ▶ With software DTS encoder, or
 - ▶ Under valgrind, or
 - ▶ ...
- ▶ Result: **Killed**

On stream start

- ▶ To process (resample, mix, encode): 50 ms of sound
 - ▶ `load-module module-udev-detect tsched_buffer_size=50000`
- ▶ Budget: 200 ms of real time (due to rtkit)
- ▶ Easy!

Wakeup timing

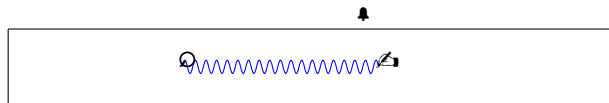
- ▶ PulseAudio goal: wake up as late as possible
- ▶ Adaptive watermark-based scheduling algorithm
 - ▶ Reacts to underruns, near-underruns or absence of them
 - ▶ Needs timestamp conversion

Wakeup timing issues

- ▶ Xonar DX eats first 5 ms of audio in no time
 - ▶ Already worked around in PulseAudio:
 - ▶ Cut sleep time in half until one buffer is played
- ▶ Imprecise hardware pointer reports
 - ▶ Adaptive watermark-based scheduling algorithm gets fooled
 - ▶ Worst case: double-buffered (batch) audio transfers
 - ▶ PulseAudio switches to period-based scheduling on batch cards

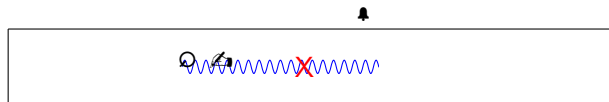
Reacting to unexpected events

- ▶ External events
 - ▶ New streams
 - ▶ Volume changes
- ▶ Need to react quickly
 - ▶ Even if a high-latency stream is playing
- ▶ Solution: rewinds!
 - ▶ ???



Reacting to unexpected events

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Rewinds in ALSA

- ▶ `snd_pcm_rewind()`
 - ▶ Please let me overwrite the last N samples!
- ▶ `snd_pcm_rewindable()`
 - ▶ How much can be rewound now?
- ▶ `snd_pcm_forward()`, `snd_pcm_forwardable()`
 - ▶ Undo a rewind
- ▶ PulseAudio assumes that full rewinds work

Rewinding hw devices

- ▶ Rewinding is easy!
 - ▶ Just move the application pointer
- ▶ Telling how much to rewind is not easy 😞
 - ▶ Problem: imprecise pointer position
 - ▶ Problem: interference with DMA controller
 - ▶ Workaround: static 256-byte or 1.33 ms “safeguard” in PulseAudio

Testing rewinds

- ▶ Use a buffer with four periods
- ▶ In a loop, after filling the buffer with silence:
 - ▶ Rewind one period
 - ▶ Write one period of silence
 - ▶ Write one period of square waves
- ▶ Correct output: silence
 - ▶ hw devices pass the test

Rewinding plugins

- ▶ Callbacks in `snd_pcm_fast_ops_t`
- ▶ Default implementations in `src/pcm/pcm_generic.c` and `src/pcm/pcm_plugin.c`
 - ▶ Forward the request to slave
 - ▶ Move application pointer

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- ▶ **Also one needs to restore state**

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 - ▶ Forward the request to slave
 - ▶ Move application pointer
- ▶ Also one needs to restore state
 - ▶ No state, no problem

Rewind support status

Good: hw, alaw, asym, copy, empty, hooks, linear, lfloat, mmap_emul, mulaw, multi, route, softvol (if nobody changes volume)

Dmix bug

- ▶ Look at this old bug:

```
if (dmix->state == SND_PCM_STATE_RUNNING ||
    dmix->state == SND_PCM_STATE_DRAINING)
    return snd_pcm_dmix_hwsync(pcm);
```

- ▶ Net result: return 0; and do not rewind
- ▶ Introduced in 2008 (patch adds 459 lines)
- ▶ Noticed and fixed in 2014
- ▶ Still there are other bugs (yet undiagnosed) ☹

iec958 plugin

- ▶ Needed on old cards for adding preambles and various auxiliary bits
- ▶ Preamble sequence:
ZYXYXYXYXYXY...ZYXYXYXYXYXY... (period = 384)
- ▶ State: position in that sequence

adpcm plugin

- ▶ Software adpcm codec
- ▶ State: `snd_pcm_adpcm_state_t`
 - ▶ Needs to be stored for past samples
 - ▶ Is now stored past the last sample only
 - ▶ Problem with testing the change

Rewind support status

Good: hw, alaw, asym, copy, empty, hooks, linear, lfloat, mmap_emul, mulaw, multi, route, softvol (if nobody changes volume), iec958 (1.0.28)

Bad but fixable: dmix, dshare, file, adpcm

Interfacing with the world

- ▶ ioplug
 - ▶ pulse, bluetooth (old), cras, a52
- ▶ extplug
 - ▶ upmix, vdownmix
 - ▶ dca, alsaequal
- ▶ ladspa

ioplug

- ▶ `struct snd_pcm_ioplug_callback`
- ▶ has `.transfer` callback
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 - ▶ They wouldn't be implementable anyway!
 - ▶ Think about unsending Bluetooth packets ☹
 - ▶ External libraries are not rewindable

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 - ▶ They wouldn't be implementable anyway!
 - ▶ Think about unsending Bluetooth packets ☹
 - ▶ External libraries are not rewindable
 - ▶ They aren't needed if `.transfer` does nothing irreversible
 - ▶ `jack` plugin has no `.transfer` callback and is rewindable ☺

Rewind support status

Good: hw, alaw, asym, copy, empty, hooks, linear, lfloat, mmap_emul, mulaw, multi, route, softvol (if nobody changes volume), iec958 (1.0.28), ioplug (without .transfer)

Bad but fixable: dmix, dshare, file, adpcm

Unfixable: ioplug (with .transfer), extplug, ladspa

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Bad but fixable: dmix, dshare, file, adpcm, rate (in principle)

Unfixable: ioplug (with .transfer), extplug, ladspa, rate (library-based or with current set of ops)

Relevant results

- ▶ a52 (ioplug)
 - ▶ already worked around (hackishly)
 - ▶ `max_rewind = 0`
- ▶ dca (extplug)
 - ▶ patch rejected
 - ▶ ALSA changes are wanted

ALSA changes

- ▶ `snd_pcm_hw_params_can_rewind()`
- ▶ Added, but then removed in favour of `snd_pcm_rewindable()`
 - ▶ Works only if the buffer size is already set
 - ▶ Returns 0 for an empty buffer
 - ▶ Verdict: unusable for PulseAudio purposes

Internal processing in PulseAudio

- ▶ Resampling
 - ▶ https://bugs.freedesktop.org/show_bug.cgi?id=50113
- ▶ Virtual sinks (echo cancellation, virtual surround)
 - ▶ Same problem with state
- ▶ Software crossover for LFE channel extraction
 - ▶ Took four attempts
 - ▶ Provoked a “how to test” question from devs
 - ▶ Works now 😊

pulse ALSA plugin issues

- ▶ Does not tell PulseAudio about rewinds
- ▶ Blindly agrees to “impossible” buffer metrics

Conclusions

- ▶ Timer-based scheduling works in simple cases
- ▶ In other cases, PulseAudio needs/has workarounds
- ▶ CRAS doesn't have any of the discussed workarounds
 - ▶ Self-inflicted problems?