

The PulseAudio Sound Server

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Who Am I?

Software Engineer at Red Hat, Inc.

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Introduction

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What is PulseAudio?

Usage

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Recipes

Outlook

Current State of Linux Audio

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OSS API emulation is kludgy: `$LD_PRELOAD!` (esddsp, artsdsp, aoss)

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If ALSA dmix wins OSS applications lose.

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- “Spatial” event sounds
- Adjust volume of one stream, if there’s a signal on another one
- ...

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- but in different configuration (no dmix!)

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Well defined, accepted APIs for pro audio, such as JACK for interconnection or LADSPA for plugins

Competition

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Microsoft ships a new userspace sound system with Windows Vista.

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We should stop abstracting abstracted abstraction layers

Missing Pieces

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Play a little catch-up with Apple, Microsoft

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LGPL licensed (practically downgraded to GPL on the server side)

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- Mixing multiple streams together
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- Do volume adjustments
- Apply other filters, echo cancellation
- Redirect or copy to another audio device or application, possibly over the network
- Reroute channels (e.g. Stereo to Surround)
- ...

Modular Design

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Toys: LIRC, multimedia keyboard (Linux evdev)

Desktop integration: X11 bell, X11 credentials

Modular Design II

Integration: JACK, EsoundD

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LADSPA filters

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Not just “Yet another audio API”!

Not a try to push yes another Esound on the people!

Features

Supports up to 32 channels.

Wide range of sample formats (PCM, μ Law, aLaw)

Realtime scheduling

Network transparent sample cacheing

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Lots of room for improvement

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PulseAudio replaces the `libasound` plugin `dmix`, `plughw`, others

PulseAudio vs. Esound

There is practically no reason left to use Esound instead of PulseAudio

PulseAudio vs. JACK

There is no “vs.”!

Different objectives - JACK: inter-application communication for pro audio

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Future integration with PulseAudio?

PulseAudio vs. aRts

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PulseAudio vs. aRts

There is no “vs.”!
aRts is officially dead.

Usage

Required Dependencies

`liboil`: Optimized inner loops

`libsndfile`: Loading sound files

Optional Dependencies

ALSA: Hardware access

GLIB: Support for integration into the GLib event loop - no hard dependency

Avahi: for Zeroconf support

JACK: for JACK integration

X11: Hook into bell event, store authentication credentials

liboil: Asynchronous name resolution

TCPWRAP: access control

LIRC: remote control

libsamplerate: High quality resampling

Compatibility

Plugin for `libasound`: most ALSA applications can access PulseAudio like a local sound card

`$LD_PRELOAD` based OSS compatibility

Implementation of the Esound protocol

Integration with JACK

Plugin for `libao`

Plugin for GStreamer (With nice tricks!)

Compatibility II

Plugin for XMMS, Audacious

Driver for MPlayer, Xine

Driver for MPD

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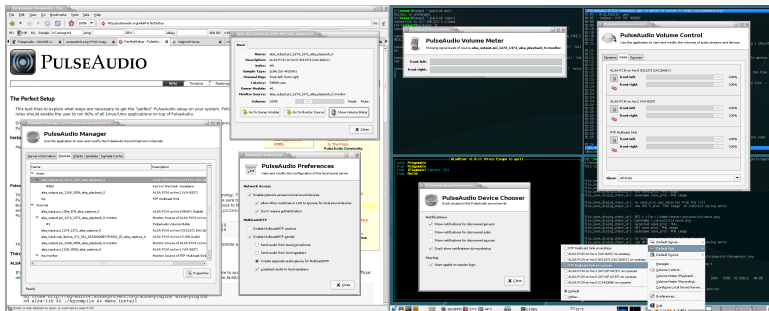
Driver for MPD

> 90 % of all Linux audio applications should run on it.

<http://pulseaudio.org/wiki/PerfectSetup>

Introduction What is PulseAudio? Usage Internals Recipes Outlook

Screenshot



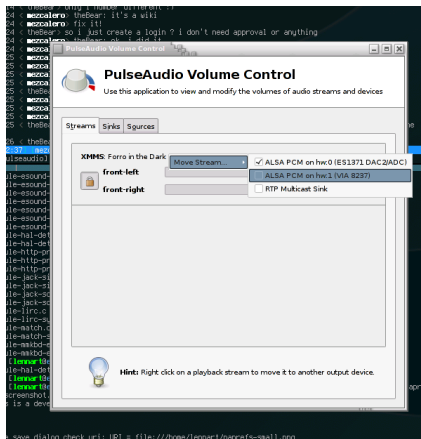
GStreamer Plugin

- Wraps playback, capturing, mixer, device enumeration
- Extracts song metadata from pipeline, uses it to name the stream in the PulseAudio server - independent from the application.
- Not yet part with upstream GStreamer, but will be

XMMS Plugin

- Wraps playback, mixer
- Uses song name to name stream in PulseAudio

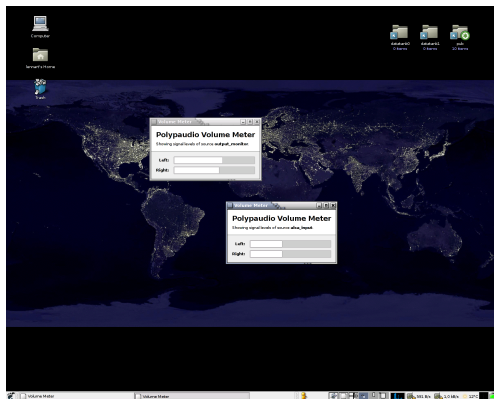
GUI: Volume Control



GUI: Volume Control II

- Shows streams, sinks, sources
- Allows volume changing for each channel separately
- Shows song name for each stream (that's why sliders are horizontal!)
- Allows to move stream from one sink to another without interrupting playback

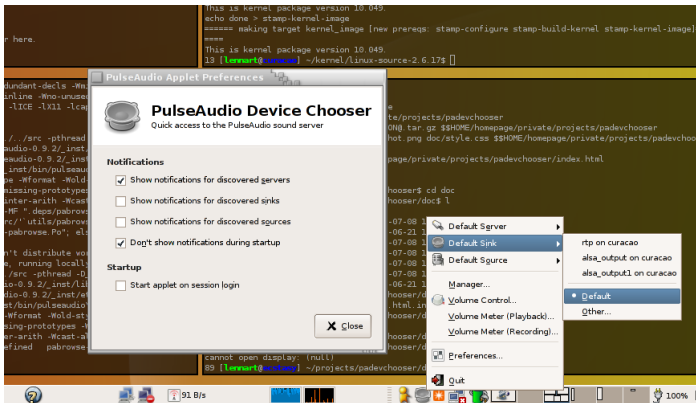
GUI: Volume Meter



GUI: Manager II

- Shows internals of a PulseAudio server
- Not for the regular user

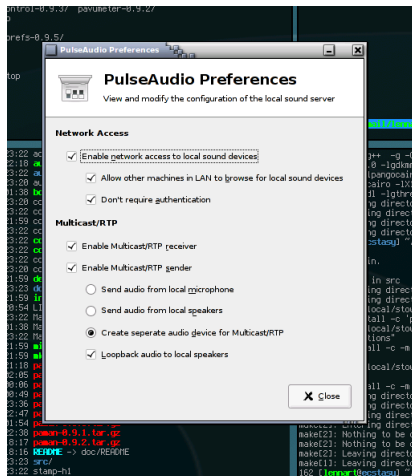
GUI: Device Chooser



GUI: Device Chooser II

- Sits in the notification area
- Allows easy changing of the input/output device/server
- Avahi support
- Easy access to the utility programs

GUI: Preferences



GUI: Preferences II

- Easy access to selected configuration options
- Communicates over GConf with PulseAudio
- Instant apply
- Requires `module-gconf` loaded into the server
- Current options: Remote access, Authentication, Zeroconf, RTP multicast receive, send

Internals

Nomenclatura

Sink - A clocked output device PA writes data to (e.g. ALSA sound card)

Source - A clocked input device PA reads data from (e.g. ALSA sound card)

Monitor Source - Implicitly attached to every sink for monitoring what is currently being played

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Sink Input - An output stream whose data PA sends on to a sink; not clocked! (e.g. a client application such as Rhythmbox)

Source Output - An input stream where PA sends to data from a source; not clocked! (e.g. a client application such as Ekiga)

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PulseAudio does not know the notion of “pipelines”!

Nomenclatura II

Module - A shared library code blob which can be loaded into the daemon at any time which can register any number of sinks, sources, inputs or outputs.

Client - A local or networked client which can allocate any number of inputs or outputs.

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Sinks and sources can be identified by a short string such as `dsp1` or `dsp1.monitor`. Name is generated automatically from the underlying device name - or may be configured manually.

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Zero-Copy memory management

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Lock-free (almost), real-time capable core

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Powerful playback model

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Automatic underrun handling

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Embeddable core

Supports multiple sinks/sources in a single daemon

Asynchronous Client API

Fully configurable during runtime

Zero-Copy Memory Management

We never copy memory around if not really necessary

Instead of queueing samples, we queue pointers to reference counted memory data blocks

Location of those memory data blocks is flexible: dynamic memory, SHM from other process, DMA buffer of sound card, mapped file

Advantages: low memory usage, low-latency

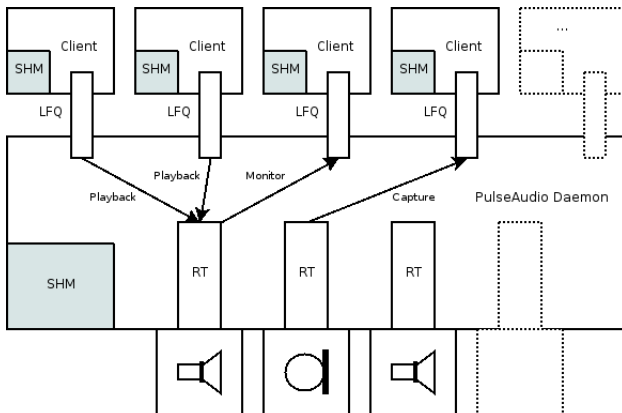
Shared-Memory data transfer

Audio data between local clients and server is transferred via shared memory

The server and each client allocates one shared memory segment and allocates its audio buffers from it. Indexes into this segment are passed between clients and between client and server.

RW access only to local segment, RO access to other process' segments

Internals



LFQ = Lock-Free Queue; SHM = Shared Memory Segment; RT = Realtime Thread

Playback Model

Playback buffer is a linked list of pointers to audio data blocks and their indexes

Monotonically advancing read index

Freely seekable write index: `SEEK_RELATIVE`, `SEEK_ABSOLUTE`, `SEEK_RELATIVE_ON_READ`, `SEEK_RELATIVE_END`

If data is written “left” of the current read pointer, it is immediately dropped

If due to seeking the buffer contains “holes” silence is automatically inserted.

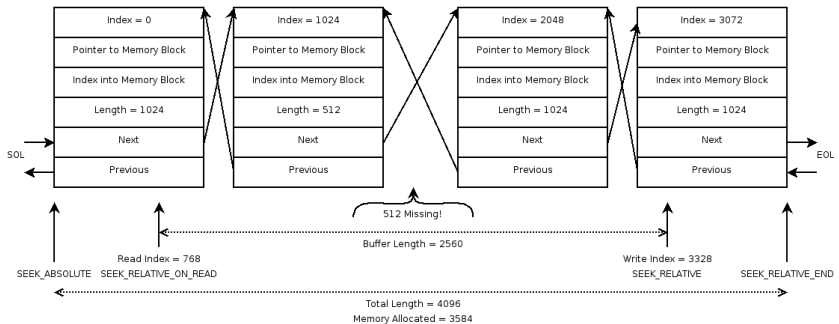
Playback Model II

Multiple streams can be synced together, read indexes will never deviate if enabled.

If buffer fill level becomes 0, an UNDERRUN event is sent to the client. If buffer fill level becomes smaller than a specified lower watermark a REQUEST event is sent to the client. If a specified maximum buffer length is reached an OVERRUN event is sent to the client.

Advantages: large buffers with quick reaction possible - useful especially over the network; easy to write RTP receivers; low memory usage

Playback Model III



Handling Underruns

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- Playback never stops, read index advances monotonically, and drops enough samples so that the time the underrun lasted is compensated.

Emulating OSS

Difficult, because kernel interfaces cannot easily be emulated from userspace

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DMA practically impossible.

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Quake2 - What about DMA?

Emulating OSS II

Possible solution: FUSD - emulating character devices from userspace.

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DMA still a problem?

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Worth it? Does anyone still use it?

Probably more applications around that run exclusively on aRts than run exclusively on Esound

Authentication

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Remote: Cookie, IP ACL, X11 root window

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Remote: Cookie, IP ACL, X11 root window

No encryption - no challenge response

Recipes

CLI - Command Line Language

PulseAudio may be configured during startup and runtime with a simple imperative command line language. (Not Turing complete).

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Interesting Commands:

```
load-module module-oss device="/dev/dsp"  
sink_name=output source_name=input channels=1
```

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Interesting Commands:

```
load-module module-oss device="/dev/dsp"
```

```
sink_name=output source_name=input channels=1
```

```
load-sample x11-bell /usr/share/sounds/notify.wav
```

Recipe: RTP Multicast “Radio” Receiver

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```
load-module module-rtp-recv
```

Recipe: RTP Multicast “Radio” Sender

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```
load-module module-null-sink sink_name=rtp
```

Recipe: RTP Multicast “Radio” Sender

```
load-module module-null-sink sink_name=rtp  
load-module module-rtp-send source=rtp.monitor
```

Recipe: RTP Multicast “Radio” Sender

```
load-module module-null-sink sink_name=rtp  
load-module module-rtp-send source=rtp.monitor  
set-default-sink rtp
```

Recipe: Output audio on two soundcards simultaneously

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```
load-module module-oss device="/dev/dsp"  
sink_name=output0
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sink_name=output0  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1
```

Recipe: Output audio on two soundcards simultaneously

```
load-module module-oss device="/dev/dsp"  
sink_name=output0  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1  
  
load-module module-combine sink_name=combined  
master=output0 slaves=output1
```

Recipe: Output audio on two soundcards simultaneously

```
load-module module-oss device="/dev/dsp"  
sink_name=output0  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1  
  
load-module module-combine sink_name=combined  
master=output0 slaves=output1  
  
set-sink-default combined
```

Recipe: Combine two Stereo devices into a virtual 4-channel Surround device

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```
load-module module-oss device="/dev/dsp"  
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```

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```
load-module module-oss device="/dev/dsp"  
sink_name=output0 channel_map=left,right channels=2  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1 channel_map=rear-left,rear-right  
channels=2
```

Recipe: Combine two Stereo devices into a virtual 4-channel Surround device

```
load-module module-oss device="/dev/dsp"  
sink_name=output0 channel_map=left,right channels=2  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1 channel_map=rear-left,rear-right  
channels=2  
  
load-module module-combine sink_name=combined  
master=output0 slaves=output1  
channel_map=left,right,rear-left,rear-right  
channels=4
```

Recipe: Combine two Stereo devices into a virtual 4-channel Surround device

```
load-module module-oss device="/dev/dsp"  
sink_name=output0 channel_map=left,right channels=2  
  
load-module module-oss device="/dev/dsp1"  
sink_name=output1 channel_map=rear-left,rear-right  
channels=2  
  
load-module module-combine sink_name=combined  
master=output0 slaves=output1  
channel_map=left,right,rear-left,rear-right  
channels=4  
  
set-sink-default combined
```

Outlook

Current Work

Best possible OSS compatibility

Current Work

Best possible OSS compatibility

More earcandy

Current Work

Best possible OSS compatibility

More earcandy

“glitch-free-ness”, timer based scheduling

Current Work

Best possible OSS compatibility

More earcandy

“glitch-free-ness”, timer based scheduling

libsydney

Future Work

Better integration with JACK

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RTP Timing synchronisation

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Better GUI tools

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Better GUI tools

<http://pulseaudio.org/browser/trunk/todo>

That's all, folks.

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Any questions?

PulseAudio

<http://pulseaudio.org/>

[https://tango.0pointer.de/mailman/listinfo/
pulseaudio-discuss](https://tango.0pointer.de/mailman/listinfo/pulseaudio-discuss)

#pulseaudio on irc.freenode.org



PulseAudio