

A room-corrected Ambisonic listening rig made with free software

A fun usecase for Linux as a versatile and reliable tool for electro-acoustic hobbyists and professionals.

With a brief glimpse into Ambisonics in case you always wondered what it might be about.

(But the methods described here also work for stereo or any other surround system.)

And most of the discussed software runs on Mac OS X, too.

About me

Jörn Nettingsmeier, aged 33,
freelance audio engineer and IT admin,
classically trained musician,
qualified event technician,
been hanging out with the Linux audio crowd
since 1998.

About your next 20 mins:

My assumptions are:

- You are interested in **Linux** and/or in setting up an Ambisonic surround rig at home.
- You have between 4 and 24 speakers and sound I/O hardware available.
 - Your speakers and room are **not perfect**.
 - You are willing and able to afford software licensing costs of 0.00€ **and**
 - You are interested in **doing stuff yourself**.

Why Ambisonics?

Free software is only fun with **free codecs**, **free algorithms** and **free exchange of information**.

All major Ambisonic patents have now expired, and there is a huge academic and hobbyist **community** of enthusiasts, plus a rich supply of freely available **code** and **documentation**.

And: *it sounds good and scales well.*

Why Linux?

Thanks to the **JACK** sound server, Linux has very flexible **inter-application routing**.

You can **mix and match** different specialised applications.

This makes Linux/JACK an ideal platform for experimental *and* production use with non-standard requirements and workflows.

And: it's free as in „speech“ *and* „beer“.

Step 1a: arrange your speakers



sweet spot
(listening seat)



For best spatial reproduction, choose a regular polygon or polyhedron.

(Yes, you can do 3D!)

Try to get the angles right. Distance errors can be corrected with delay.

Step 1b: measure angle and distance



sweet spot
(listening seat)



For each speaker, measure its actual angle (front is 0° , turn is counter-clockwise) and distance from sweet spot.



**laser range finder
laser angle gauge**

Step 2: Get speaker IRs



instrumentation
microphone in
sweet spot



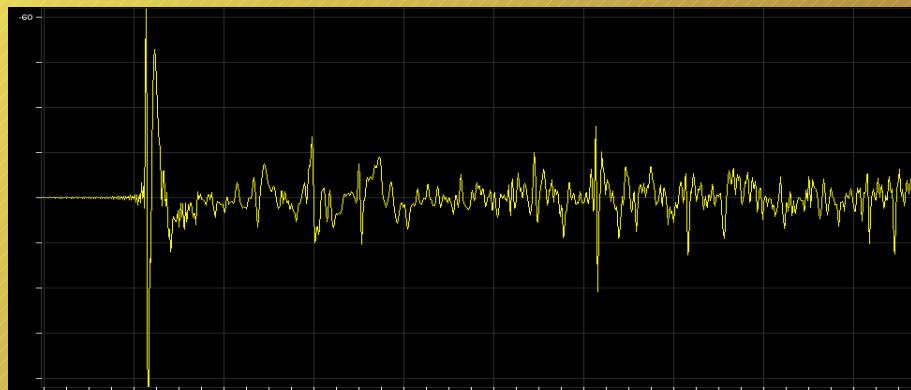
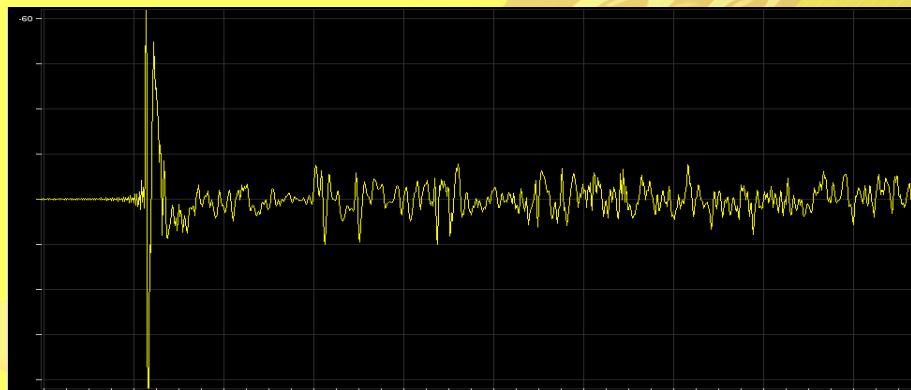
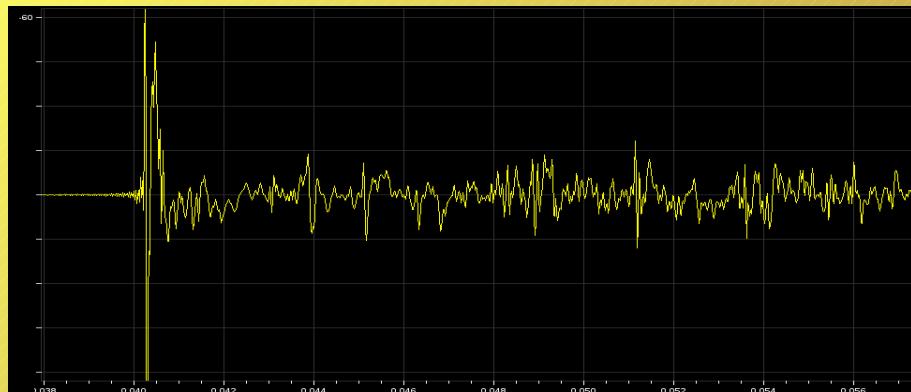
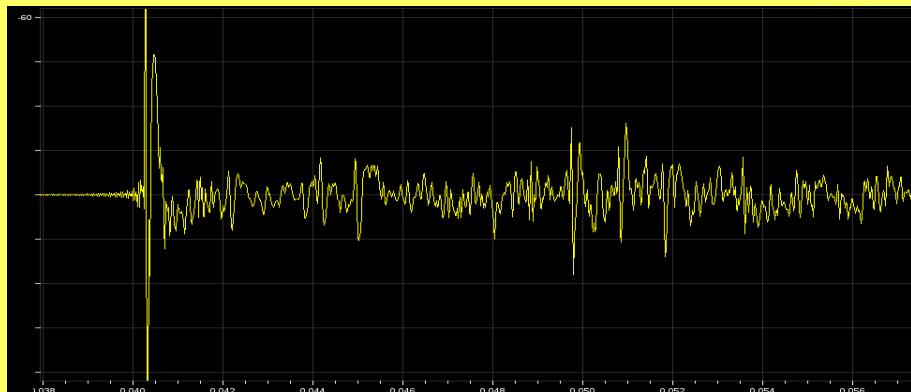
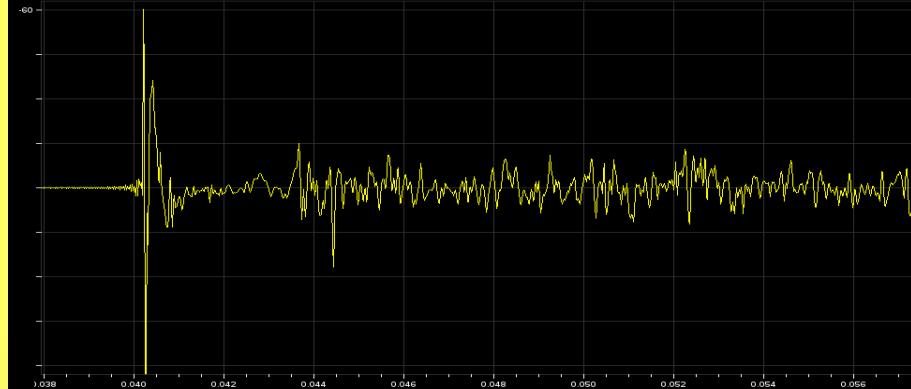
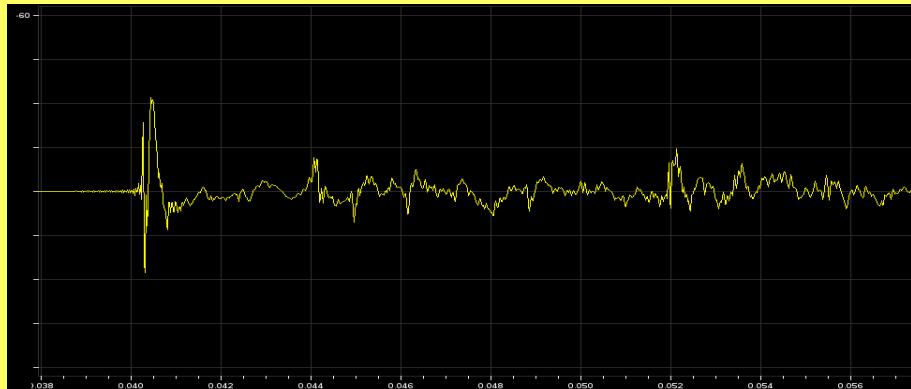
For each speaker,
measure its impulse
response at the listening
position (which will
include room effects).

For best S/N ratio, use
the swept-sine technique
(Farina 2000).



Aliki
a flat omni mike

Speaker IRs after deconvolution:



Step 3: „invert“ your IRs

To get a correction filter, answer this simple question:

„Which FIR kernel will yield a perfect pulse when convolved with the actual speaker/room IR?“

Ok, jokes aside, computer to the rescue
(and of course getting a usable, stable filter is a lot harder in practice).



Step 3: „invert“ your IRs

drc is a room correction package with excellent documentation:

<http://drc-fir.sourceforge.net>

Use version 2.7.0
(3.0.0 has some unresolved numerical instabilities
and chokes on some IRs, like mine...)

Step 3: „invert“ your IRs

Begin with a mild correction setting (such as the ERB example configuration) for a large sweet spot.

Define your desired frequency response as a set of spline points in a text file.

If your speakers are port-loaded: don't push them below the port frequency!

0	-40.0
35	-40.0
40	0.00
50	0.00
100	0.00
150	0.00
19900	-3.95
20000	-4.00
20500	-4.00
22050	-40.0

Step 3: „invert“ your IRs

Oh, I forgot: For those new to the UNIX way of doing things, drc is a **command line tool**.



Step 3: „invert“ your IRs

```
$ for i in speaker_ir-*.pcm ; do drc --BCInFile $i -PSOutFile  
filter-$i erb-48.0.drc ; done
```

```
DRC 2.7.0: Digital Room Correction  
Copyright (C) 2002-2008 Denis Sbragion
```

```
Input configuration file: erb-48.0.drc  
Parsing configuration file...  
<...>
```

```
$ for i in filter-speaker_ir-*; do sox -f -4 -r48000 -c1 -t raw $i  
$i.wav; done
```

Invoke drc with the ERB configuration file (included), specify the input IR and the output file name for the filter kernels.

Step 3: „invert“ your IRs

```
$ for i in speaker_ir-*pcm ; do drc --BCInFile $i -PSOutFile  
filter-$i erb-48.0.drc ; done
```

```
DRC 2.7.0: Digital Room Correction  
Copyright (C) 2002-2008 Denis Sbragion  
  
Input configuration file: erb-48.0.drc  
Parsing configuration file...  
<...>
```

```
$ for i in filter-speaker_ir-*; do sox -f -4 -r48000 -c1 -t raw $i  
$i.wav; done
```

Process all speaker
IRs at once using a
shell loop.
(Nice, isn't it?)

Step 3: „invert“ your IRs

```
$ for i in speaker_ir-* .pcm ; do drc --BCInFile $i -PSOutFile  
filter-$i erb-48.0.drc ; done
```

```
DRC 2.7.0: Digital Room Correction  
Copyright (C) 2002-2008 Denis Sbragion
```

```
Input configuration file: erb-48.0.drc  
Parsing configuration file...
```

```
<...>
```

```
$ for i in filter-speaker_ir-* ; do sox -f -4 -r48000 -c1 -t raw $i  
$i.wav; done
```

Convert the raw filters to .wav files with **sox** (again in a loop to save typing).



Step 3: „invert“ your IRs

```
$ for i in speaker_ir-*.pcm ; do drc --BCInFile $i -PSOutFile  
filter-$i erb-48.0.drc ; done
```

DRC 2.7.0: Digital Room Correction
Copyright (C) 2002-2008 Denis Sbragion

Input configuration file: erb-48.0.drc
Parsing configuration file...
<...>

```
$ for i in filter-speaker_ir-*; do sox -f -4 -r48000 -c1 -t raw $i  
$i.wav; done
```

**Scared?
Well, you can
always...**



...rent a nerd!

Write to linux-audio-user@lists.linuxaudio.org!

Step 4: Signal flow & software setup

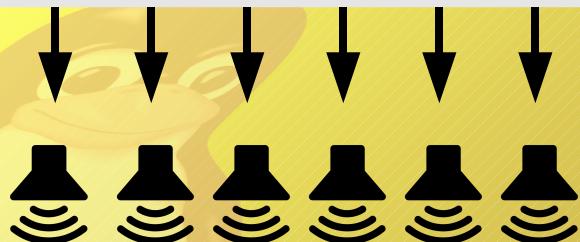
B-format audio



Ambisonic decoder

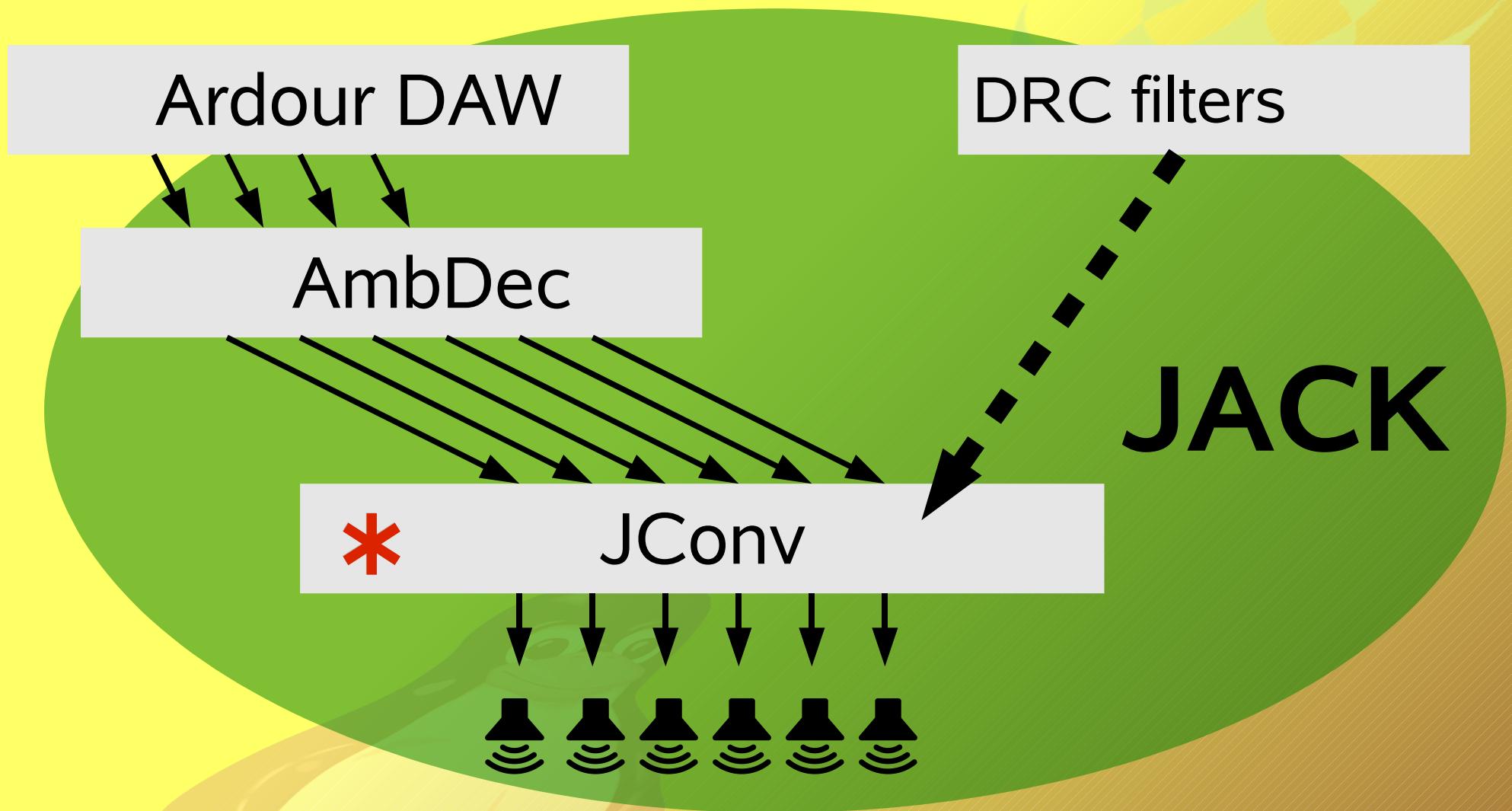
DRC filters

* Realtime convolver



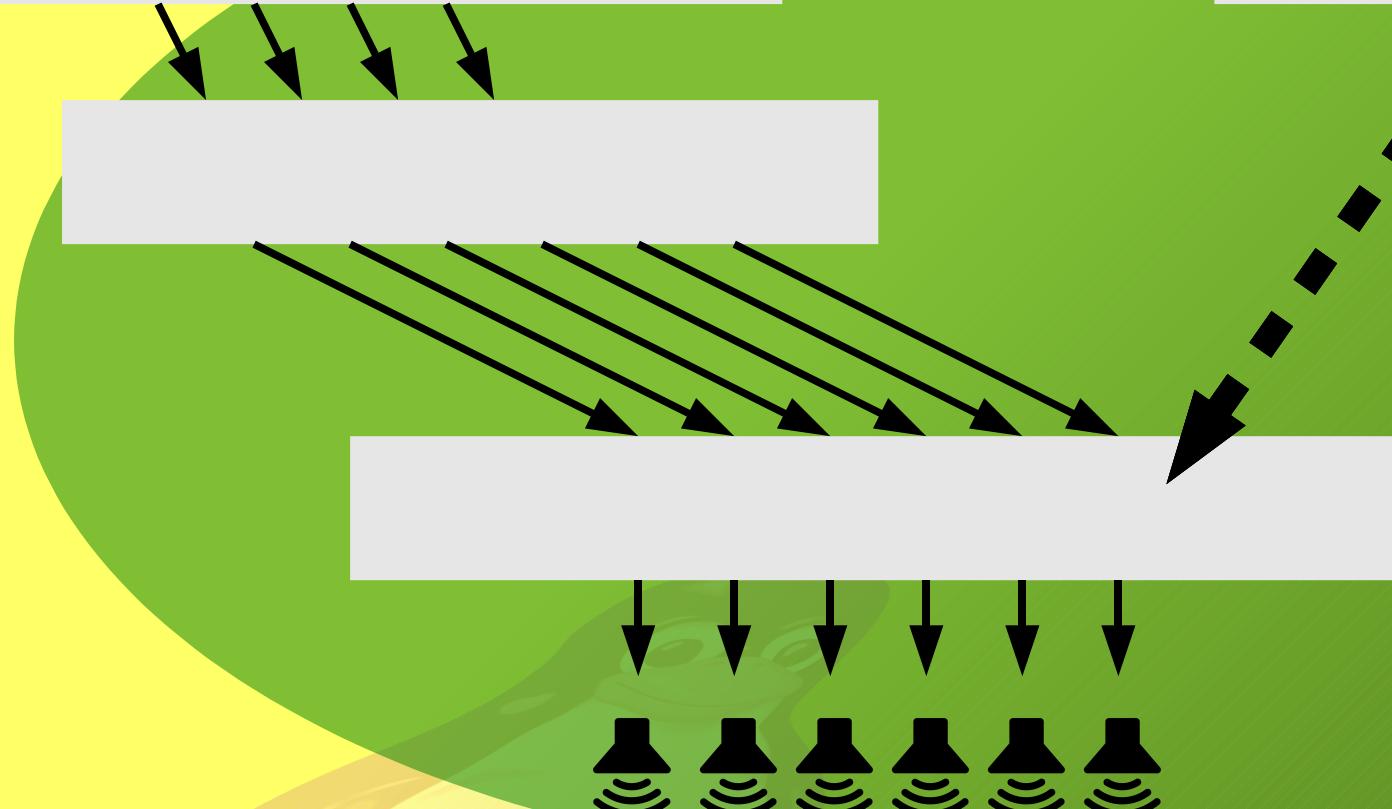
JACK
Ardour
AmbDec
JConv

Step 4: Signal flow & software setup



Step 4: Signal flow & software setup

Ardour DAW



Ardour is a complete DAW and also a very versatile mixer,

ardour 2.6.1
(built from revision 4119)

Session Transport Edit Region Track View JACK Window Options Help

48 kHz / 5.3 ms Buffers p:89% c:99% DSP: 23.1%

1.00 % sprung 30 Internal Punch In Auto Play Auto Input SOLO AUDITION

Time master Punch Out Auto Return Click

Slide Edit 00:00:00:00 No Grid Beats Mouse < > 00:00:05:00

Timecode 00:00:00:00 00:05:00:00 00:10:00:00

Meter 4/4

Tempo 120.00

Loop/Punch Ranges loop

CD Markers

Location Markers start

B-Format m s p a g

AJH_eight-Britten_1st-Britten_2nd-Britten_4th-Britten_3rd-Dvorak-Thr-AJH_Brahm-AJH_Stravi

Regions Tracks/Busses Snapshots

JKmeter jkmeter-0.4.0

Strips master Mon Mon DRC B-Format

Mute Solo Mute Solo Mute Solo Mute Solo Mute Solo

-0.0 -33.7 -0.2 -34.7 -0.0 -11.9 0.0 -11.9

-4 -0 -3 -10 -20 -30 -40 -50

ambi-drc - Mixer - Ardour <@kleineronkel>

JKmeter jkmeter-0.4.0

Strips master Mon DRC B-Format

Mute Solo Mute Solo Mute Solo Mute Solo Mute Solo

-0.0 -33.7 -0.2 -34.7 -0.0 -11.9 0.0 -11.9

-4 -0 -3 -10 -20 -30 -40 -50

AMB Rotate

JKmeter jkmeter-0.4.0

Strips master Mon DRC B-Format

Mute Solo Mute Solo Mute Solo Mute Solo Mute Solo

-0.0 -33.7 -0.2 -34.7 -0.0 -11.9 0.0 -11.9

-4 -0 -3 -10 -20 -30 -40 -50

JKmeter jkmeter-0.4.0

Strips master Mon DRC B-Format

Mute Solo Mute Solo Mute Solo Mute Solo Mute Solo

-0.0 -33.7 -0.2 -34.7 -0.0 -11.9 0.0 -11.9

-4 -0 -3 -10 -20 -30 -40 -50



with an arbitrary number of channels in tracks&buses!

*ambi-drc - Ardour <@kleineronkel>

Session Transport Edit Region Track View JACK Window Options Help

48 kHz / 5.3 ms Buffers p:89% c:99% DSP: 23.1%

1.00 % sprung 30 Internal Punch In Auto Play Auto Input SOLO AUDITION

Time master Punch Out Auto Return Click

Slide Edit 00:00:00:00 No Grid Beats Mouse < > 00:00:05:00

Timecode 00:00:00:00 00:05:00:00 00:10:00:00

Meter 4/4

Tempo 120.00

Loop/Punch Ranges loop

CD Markers

Location Markers start

B-Format m s p a g

AJH_eight-Britten_1st-Britten_2nd-Britten_4th-Britten_3rd-Dvorak-Thr-AJH_Brahm-AJH_Stravi

Regions Tracks/Busses Snapshots

Left

CC BY SA

The Ardour session window displays a timeline from 00:00:00:00 to 00:05:00:00. Multiple tracks are visible, with a red arrow pointing to the "AJH_Stravinsky-Pulcinella-Suite" track. The transport controls include play, stop, and record buttons, along with transport options like "Auto Input" and "SOLO". The bottom right corner features Creative Commons and Attribution-ShareAlike license icons.

ardour 2.6.1
(built from revision 4119)

JKMETER jkmeter-0.4.0

ambi-drc - Mixer - Ardour <@kleineronkel>

For example,
a 4 channel
B-format
surround
track...

Session Transport Edit Region Track View JACK Window Options Help

48 kHz / 5.3 ms Buffers p:89% c:99% DSP: 23.1%

1.00 % sprung 30 Internal Punch In Auto Play Auto Input SOLO NDF Time master Punch Out Auto Return Click AUDITION

Slide Edit 00:00:00:00 No Grid Beats Mouse < > 00:00:05:00

Timecode 00:00:00:00 00:05:00:00 00:10:00:00

Meter 4/4
Tempo 120.00
Loop/Punch Ranges loop
CD Markers
Location Markers start

B-Format m s p a g

AJH_eight-Britten_1st

AJH_Brahm

AJH_Stravi

Regions Tracks/Busses Snapshots

Left

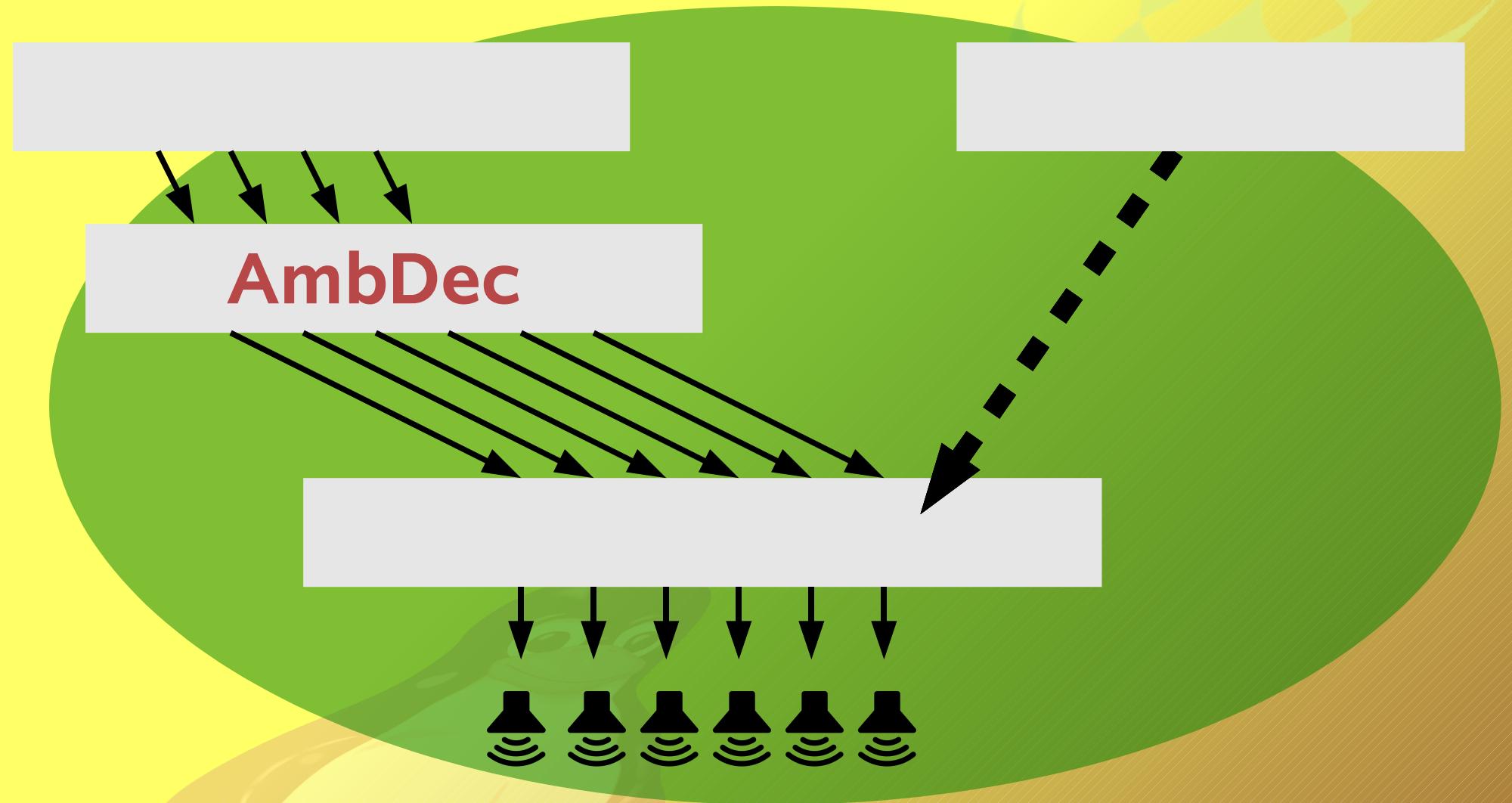
ardour 2.6.1
(built from revision 4119)

JKMETER jkmeter-0.4.0

ambi-drc - Mixer - Ardour <@kleineronkel>

...or
6 channel
speaker feed
monitor
buses.

Step 4: Signal flow & software setup



AmbDec

LF Decoder					HF Decoder					
Speakers			G(order)		0.3535	0.3535	G(order)		0.3535	0.2500
ID	Dist	Azim	Elev	W	X	Y	W	X	Y	
1	1.39	0.0	0.0	1.0000	1.4142	0.0000	1.0000	1.4142	0.0000	
2	1.38	60.0	0.0	1.0000	0.7071	-1.2247	1.0000	0.7071	-1.2247	
3	1.38	120.0	0.0	1.0000	-0.7071	-1.2247	1.0000	-0.7071	-1.2247	
4	1.74	180.0	0.0	1.0000	-1.4142	0.0000	1.0000	-1.4142	0.0000	
5	1.40	-120.0	0.0	1.0000	-0.7071	1.2247	1.0000	-0.7071	1.2247	
6	1.42	60.0	0.0	1.0000	0.7071	1.2247	1.0000	0.7071	1.2247	

an Ambisonic
decoder for
2D and 3D

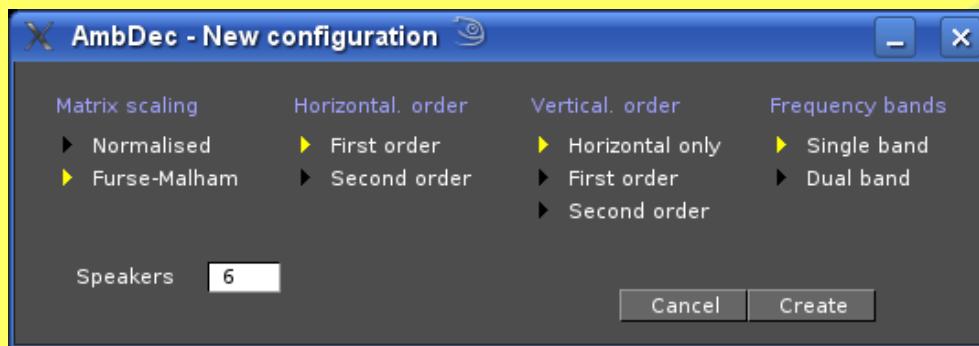
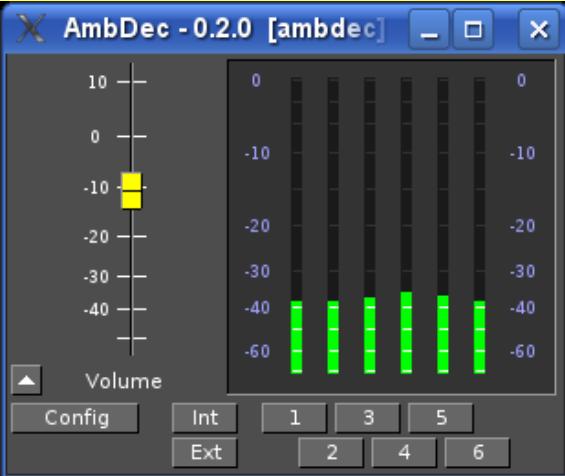
up to 24
speakers

psycho-
acoustically
correct
dual-band
operation

near-field
compen-
sation

ves.net>





AmbDec - Configuration

Load Save New Cancel Apply .ambdec/hexagon-2009-09-11.ambdec

Description hexagon, max rV at LF, max rE at HF

Matrix scaling Input scaling Speaker distance Near-field comp. Crossover freq.

- ▶ Normalised
- ▶ Furse-Malham
- ▶ Normalised
- ▶ Furse-Malham
- ▶ Delay comp.
- ▶ Gain comp.
- ▶ None
- ▶ On inputs
- ▶ On outputs

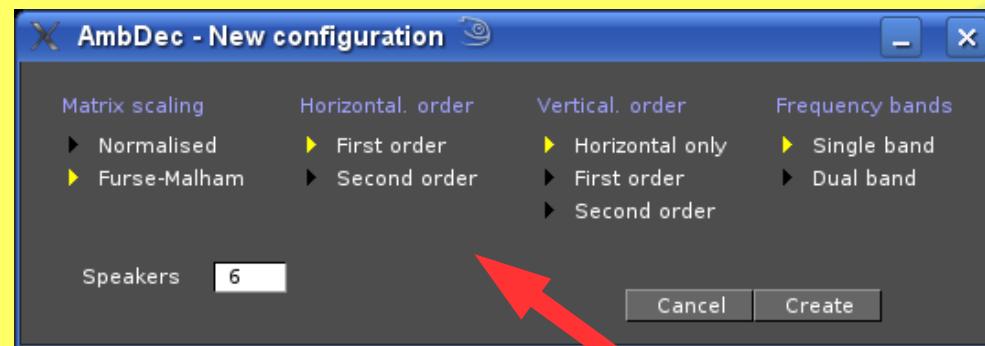
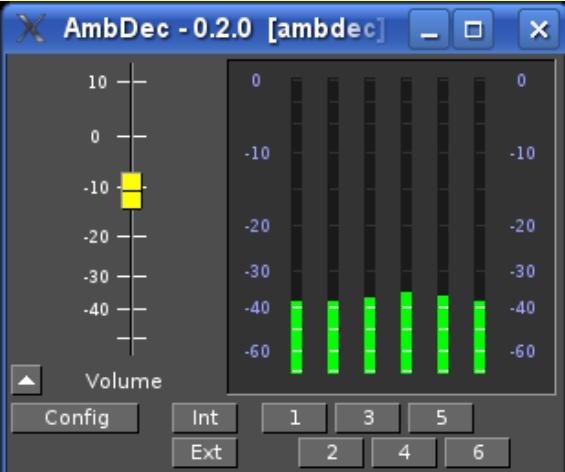
500

LF Decoder					HF Decoder				
Speakers			G(order)	0.3535	0.3535	G(order)	0.3535	0.2500	
ID	Dist	Azim	Elev	W	X	Y	W	X	Y
1	1.39	0.0	0.0	1.0000	1.4142	0.0000	1.0000	1.4142	0.0000
2	1.38	60.0	0.0	1.0000	0.7071	-1.2247	1.0000	0.7071	-1.2247
3	1.38	120.0	0.0	1.0000	-0.7071	-1.2247	1.0000	-0.7071	-1.2247
4	1.74	180.0	0.0	1.0000	-1.4142	0.0000	1.0000	-1.4142	0.0000
5	1.40	-120.0	0.0	1.0000	-0.7071	1.2247	1.0000	-0.7071	1.2247
6	1.42	60.0	0.0	1.0000	0.7071	1.2247	1.0000	0.7071	1.2247

Speakers Decoder

AmbDec

Create a configuration for your speaker rig:



AmbDec - Configuration

Load Save New Cancel Apply .ambdec/hexagon-2009-09-11.ambdec

Description: hexagon, max rV at LF, max rE at HF

Matrix scaling: Normalised, Furse-Malham

Input scaling: Normalised, Furse-Malham

Speaker distance: Delay comp., Gain comp.

Near-field comp.: None, On inputs, On outputs

Crossover freq.: 500

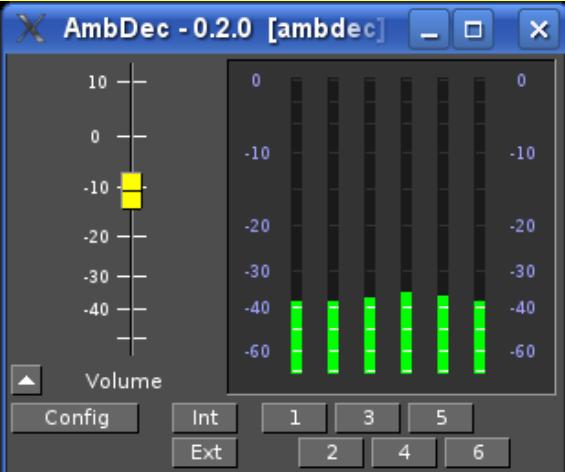
LF Decoder					HF Decoder				
Speakers			G(order)	0.3535	0.3535	G(order)	0.3535	0.2500	
ID	Dist	Azim	Elev	W	X	Y	W	X	Y
1	1.39	0.0	0.0	1.0000	1.4142	0.0000	1.0000	1.4142	0.0000
2	1.38	60.0	0.0	1.0000	0.7071	-1.2247	1.0000	0.7071	-1.2247
3	1.38	120.0	0.0	1.0000	-0.7071	-1.2247	1.0000	-0.7071	-1.2247
4	1.74	180.0	0.0	1.0000	-1.4142	0.0000	1.0000	-1.4142	0.0000
5	1.40	-120.0	0.0	1.0000	-0.7071	1.2247	1.0000	-0.7071	1.2247
6	1.42	60.0	0.0	1.0000	0.7071	1.2247	1.0000	0.7071	1.2247

Speakers Decoder

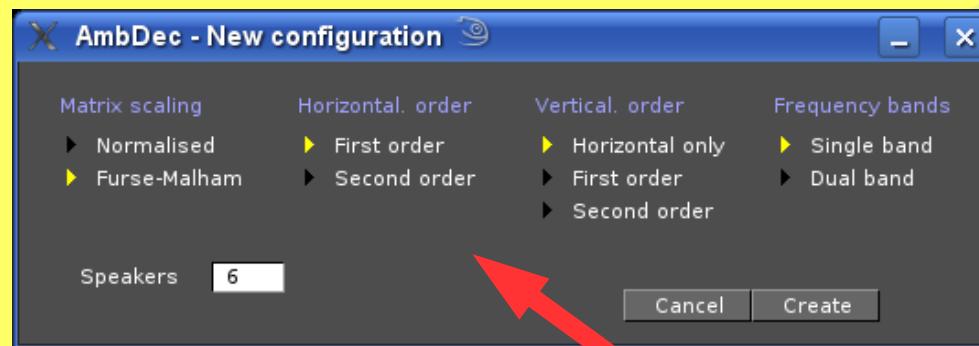
AmbDec

Create a configuration for your speaker rig:

Basic settings (order, number of speakers)



AmbDec



Create a configuration for your speaker rig:

AmbDec - Configuration

Load Save New Cancel Apply .ambdec/hexagon-2009-09-11.ambdec

Description: hexagon, max rV at LF, max rE at HF

Matrix scaling: Normalised, Furse-Malham

Input scaling: Normalised, Furse-Malham

Speaker distance: Delay comp., Gain comp.

Near-field comp.: None, On inputs, On outputs

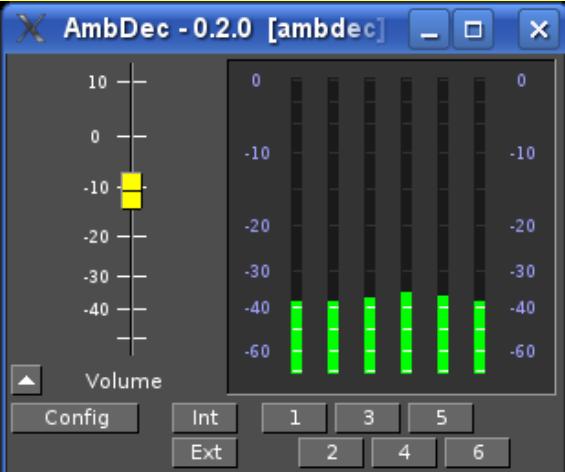
Crossover freq.: 500

LF Decoder					HF Decoder				
Speakers			G(order)	0.3535	0.3535	G(order)	0.3535	0.2500	
ID	Dist	Azim	Elev	W	X	Y	W	X	Y
1	1.39	0.0	0.0	1.0000	1.4142	0.0000	1.0000	1.4142	0.0000
2	1.38	60.0	0.0	1.0000	0.7071	-1.2247	1.0000	0.7071	-1.2247
3	1.38	120.0	0.0	1.0000	-0.7071	-1.2247	1.0000	-0.7071	-1.2247
4	1.74	180.0	0.0	1.0000	-1.4142	0.0000	1.0000	-1.4142	0.0000
5	1.40	-120.0	0.0	1.0000	-0.7071	1.2247	1.0000	-0.7071	1.2247
6	1.42	60.0	0.0	1.0000	0.7071	1.2247	1.0000	0.7071	1.2247

Speakers Decoder

Basic settings (order, number of speakers)

Decoding Matrix (fill in your speaker positions).



AmbDec

The figure shows the 'AmbDec - Configuration' dialog. It includes tabs for 'Load', 'Save', 'New', 'Cancel', and 'Apply'. The 'New' tab is selected, showing a file path: '.ambdec/hexagon-2009-09-11.ambdec'. The 'Description' field contains 'hexagon, max rV at LF, max rE at HF'. Under 'Matrix scaling', 'Normalised' is selected. Under 'Input scaling', 'Normalised' is selected. Under 'Speaker distance', 'Delay comp.' is selected. Under 'Near-field comp.', 'None' is selected. Under 'Crossover freq.', '500' is entered. The 'LF Decoder' and 'HF Decoder' sections show tables of matrix coefficients. A large red circle highlights the 'Matrix coefficients from literature' section, which contains a table of speaker positions and their corresponding matrix coefficients.

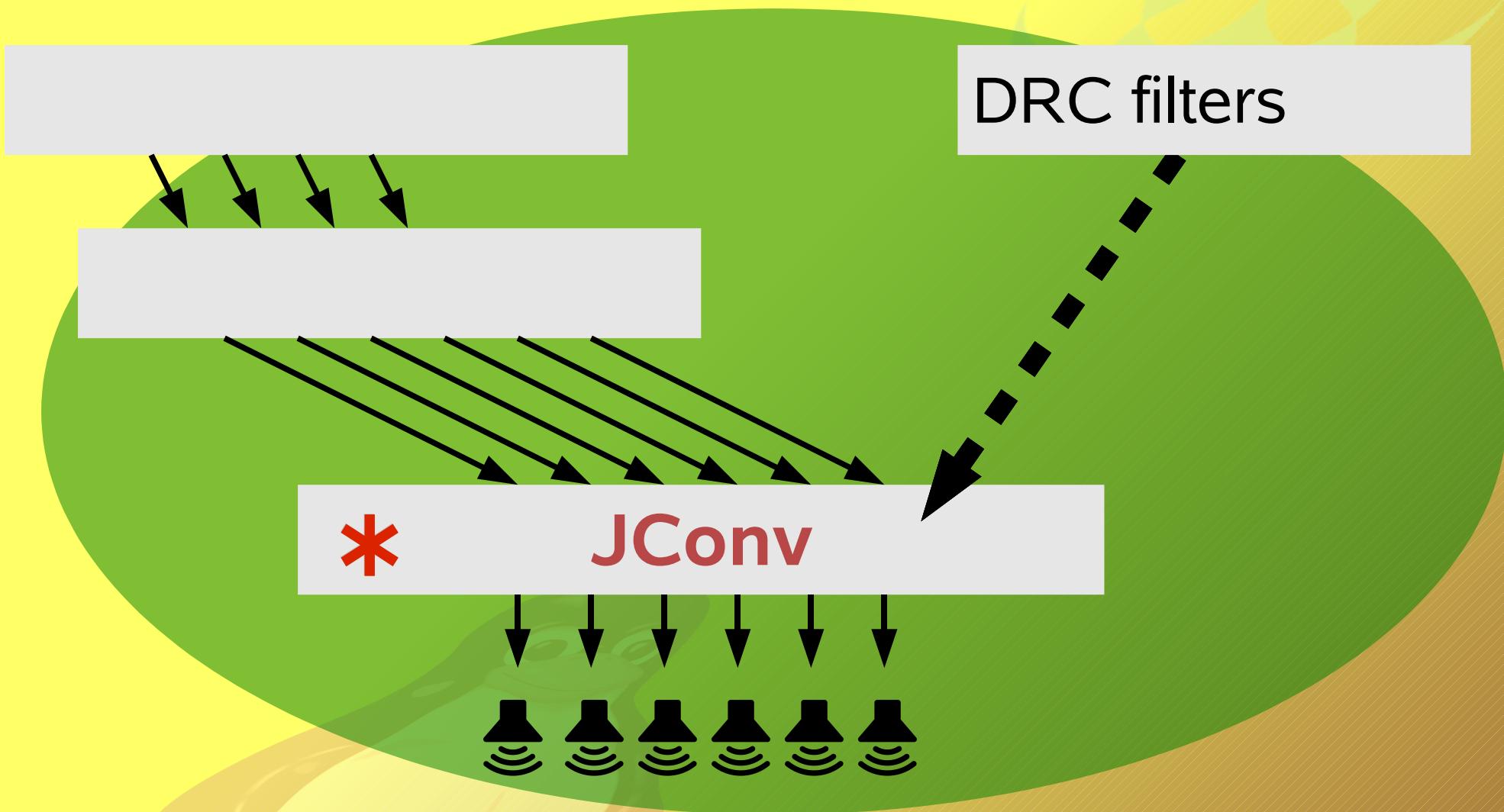
ID	Dist	Azim	Elv	W	X	Y	Z	G(order)	X	Y	Z	G(order)
1	1.39	0.0	0.0	1.0000	1.4142	0.0000	0.0000	0.3535	1.0000	1.4142	0.0000	0.3535
2	1.38	60.0	0.0	1.0000	0.7071	-1.2247	1.0000	0.7071	0.7071	-1.2247	0.0000	0.5535
3	1.38	120.0	0.0	1.0000	-0.7071	-1.2247	1.0000	-0.7071	-0.7071	-1.2247	0.0000	0.2500
4	1.74	180.0	0.0	1.0000	-1.4142	0.0000	1.0000	-1.4142	-1.4142	0.0000	0.0000	
5	1.40	-120.0	0.0	1.0000	-0.7071	1.2247	1.0000	-0.7071	0.7071	1.2247	0.0000	
6	1.42	60.0	0.0	1.0000	0.7071	1.2247	1.0000	0.7071	-0.7071	1.2247	0.0000	

Create a configuration for your speaker rig:

Basic settings (order, number of speakers)

Decoding Matrix (fill in your speaker positions).

Step 4: Signal flow & software setup



JConv is a real-time convolution engine (and another command-line tool). :-D

It runs in the background and takes a simple configuration file:

```
/cd /home/nettings/drc-filters

#           in   out   partition    maxsize
# -----
/convolver/new  6     6          1024      256000

#
#           in   out   gain   delay   offset   length   chan   file
# -----
/impulse/read   1     1     1       0        0        0        1   filter-speaker_ir-1.pcm.wav
/impulse/read   2     2     1       0        0        0        1   filter-speaker_ir-2.pcm.wav
/impulse/read   3     3     1       0        0        0        1   filter-speaker_ir-3.pcm.wav
/impulse/read   4     4     1       0        0        0        1   filter-speaker_ir-4.pcm.wav
/impulse/read   5     5     1       0        0        0        1   filter-speaker_ir-5.pcm.wav
/impulse/read   6     6     1       0        0        0        1   filter-speaker_ir-6.pcm.wav
```

JConv is a real-time convolution engine (and another command-line tool). :-D

It runs in the background and takes a simple configuration file:

```
/cd /home/nettings/drc-filters

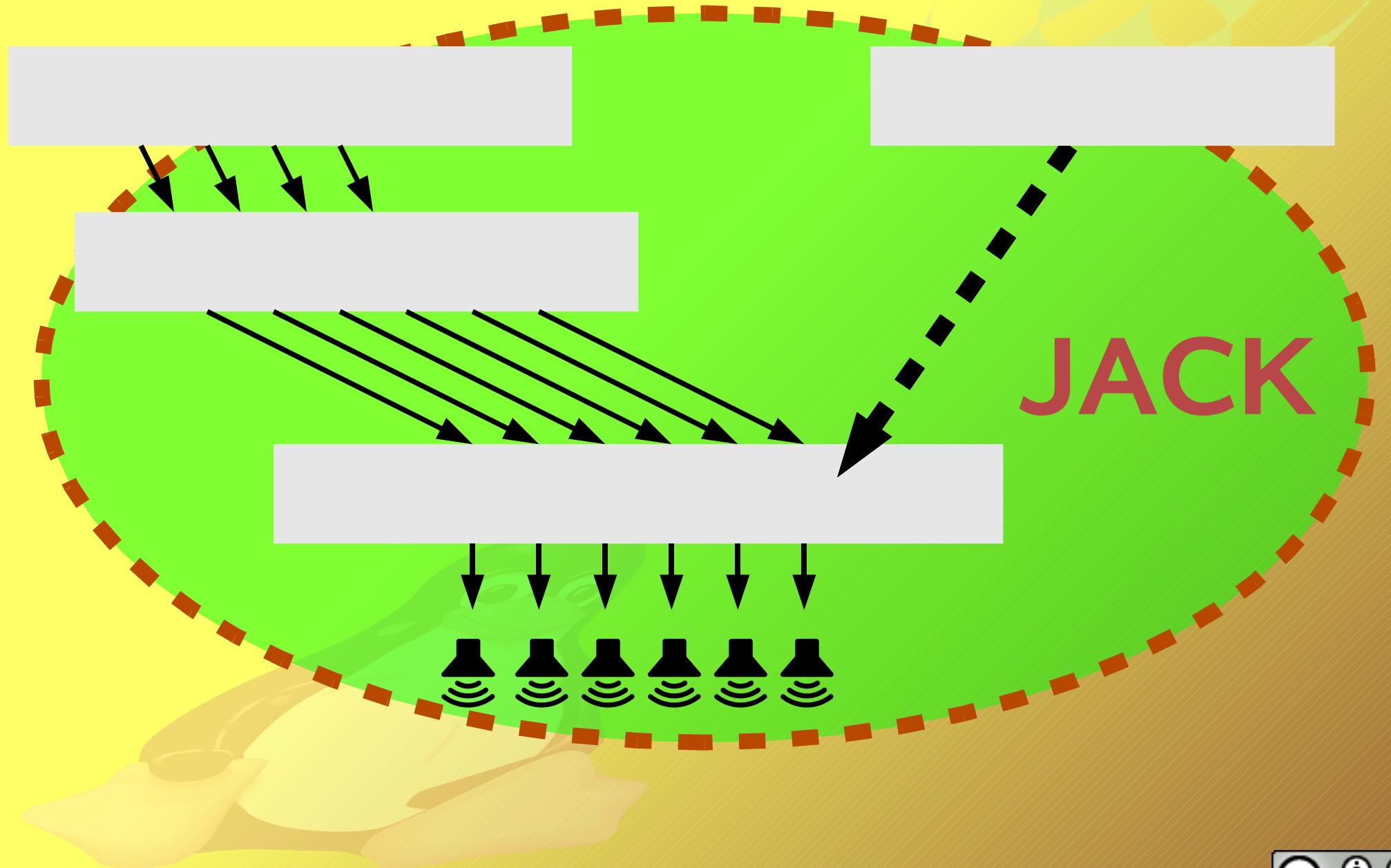
#          in   out   partition      maxsize
# -----
/convolver/new    6     6           1024      256000

#
#          in   out   gain   delay   offset   length   chan   file
# -----
/impulse/read    1     1     1       0       0       0       1   filter-speaker_ir-1.pcm.wav
/impulse/read    2     2     1       0       0       0       1   filter-speaker_ir-2.pcm.wav
/impulse/read    3     3     1       0       0       0       1   filter-speaker_ir-3.pcm.wav
/impulse/read    4     4     1       0       0       0       1   filter-speaker_ir-4.pcm.wav
/impulse/read    5     5     1       0       0       0       1   filter-speaker_ir-5.pcm.wav
/impulse/read    6     6     1       0       0       0       1   filter-speaker_ir-6.pcm.wav
```

The filter kernels we created earlier:



Step 4: Signal flow & software setup

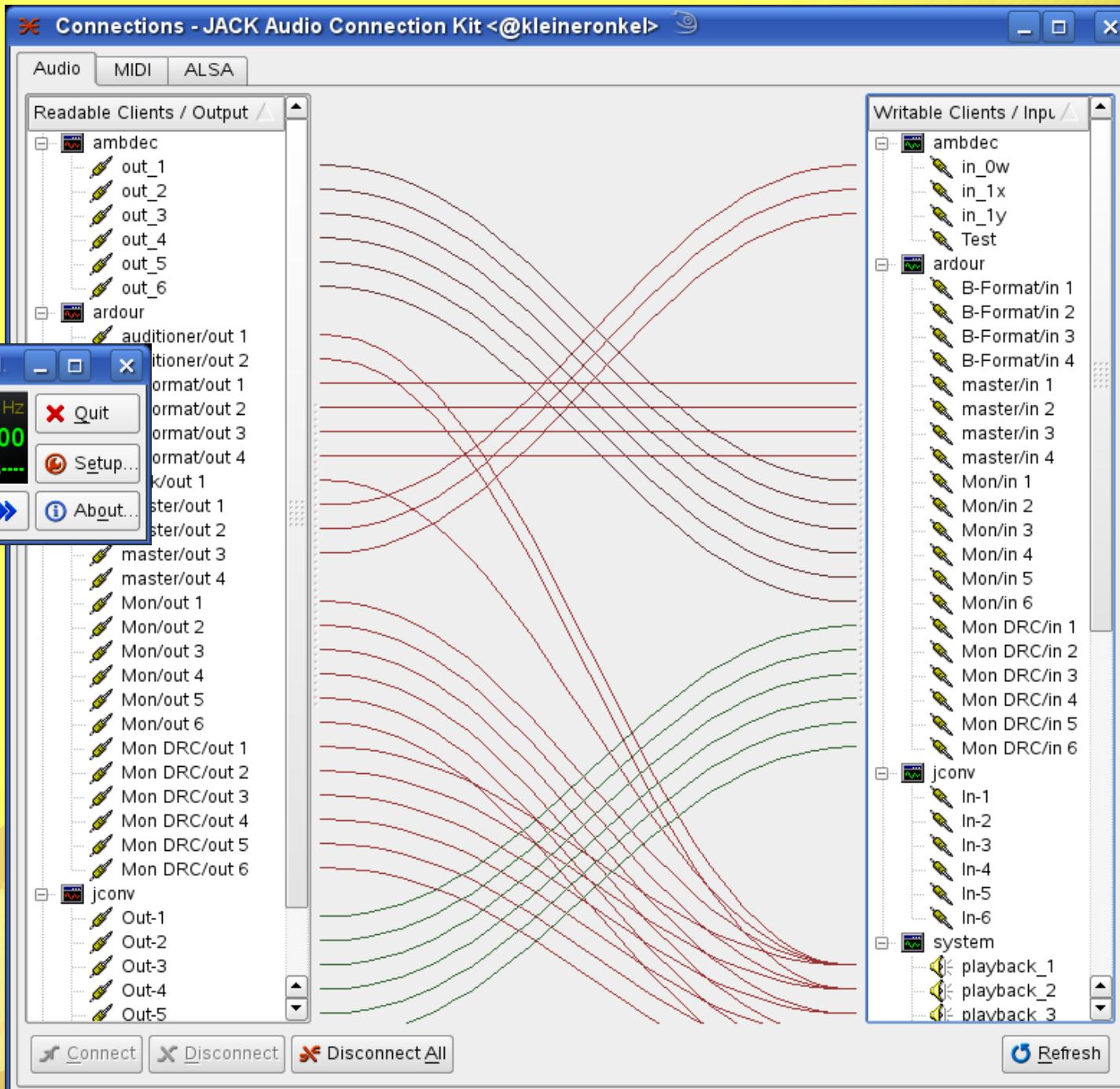


Putting it all together with JACK and qjackctl:



Imagine this with 24 speakers!

Thankfully, virtual cables are cheap :-D



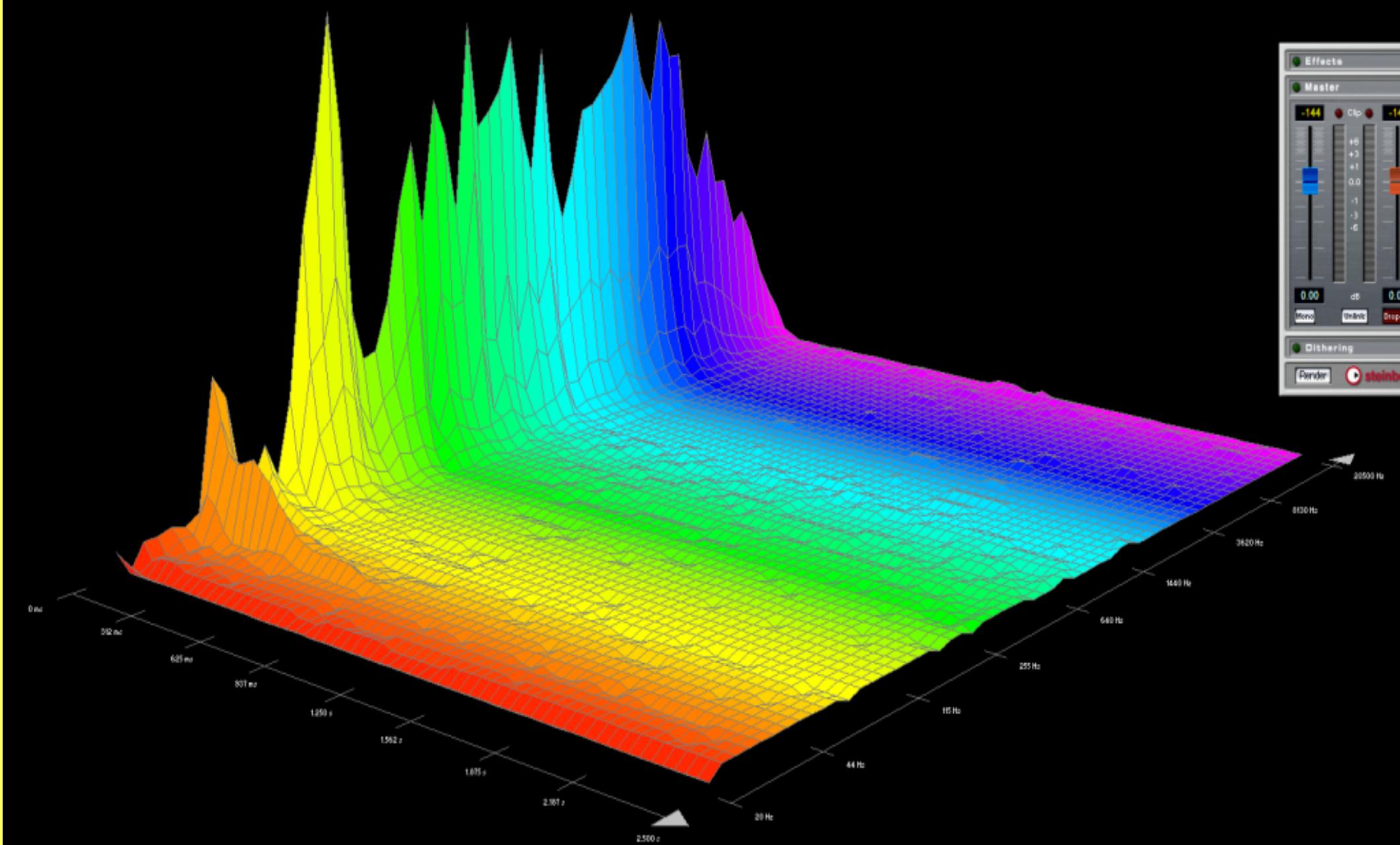
So what does it sound like?

To evaluate the effects of the application of DRC,
here are a few „before & after“ comparisons...:

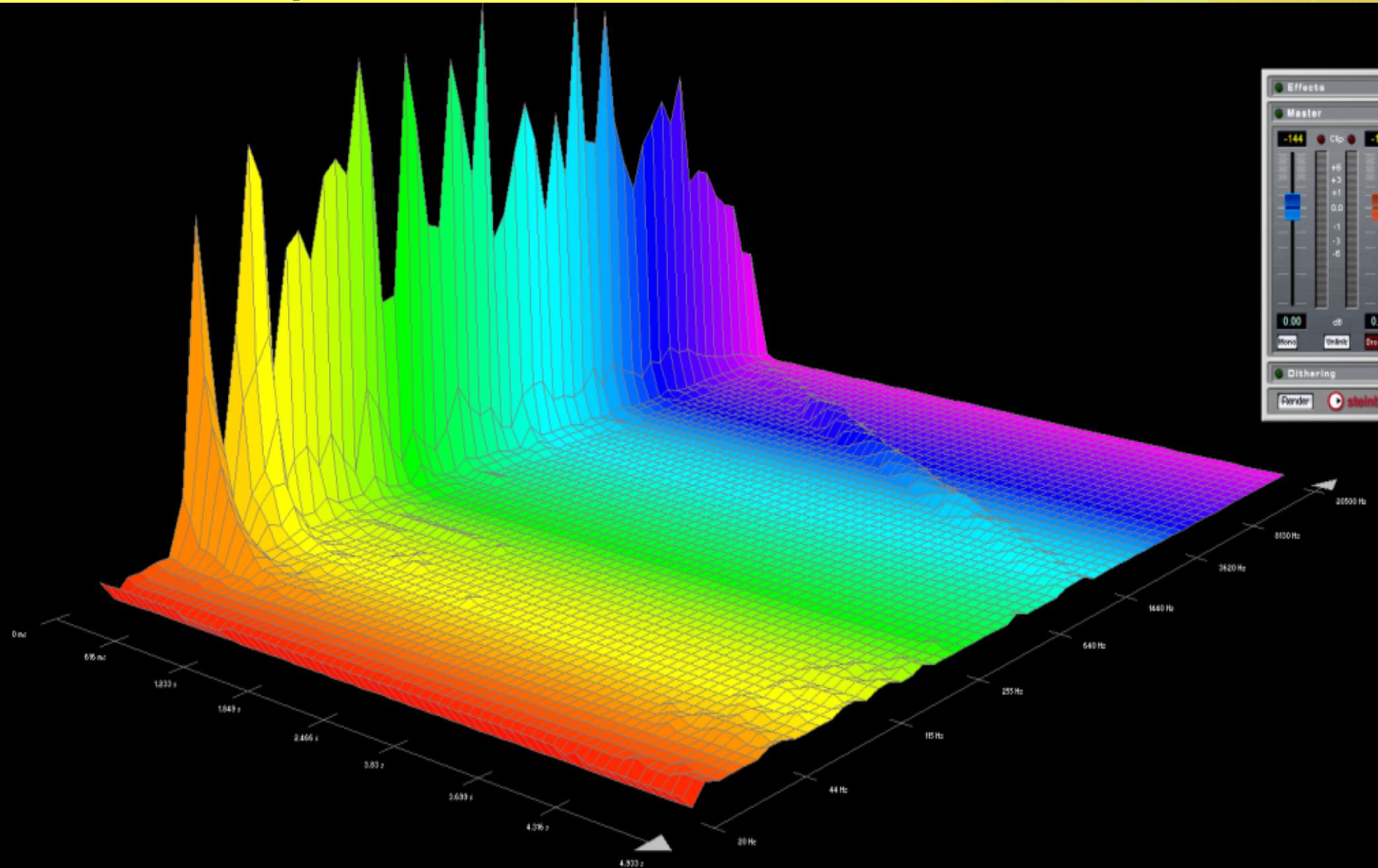


**JAPA,
WaveLab (er, well...)**

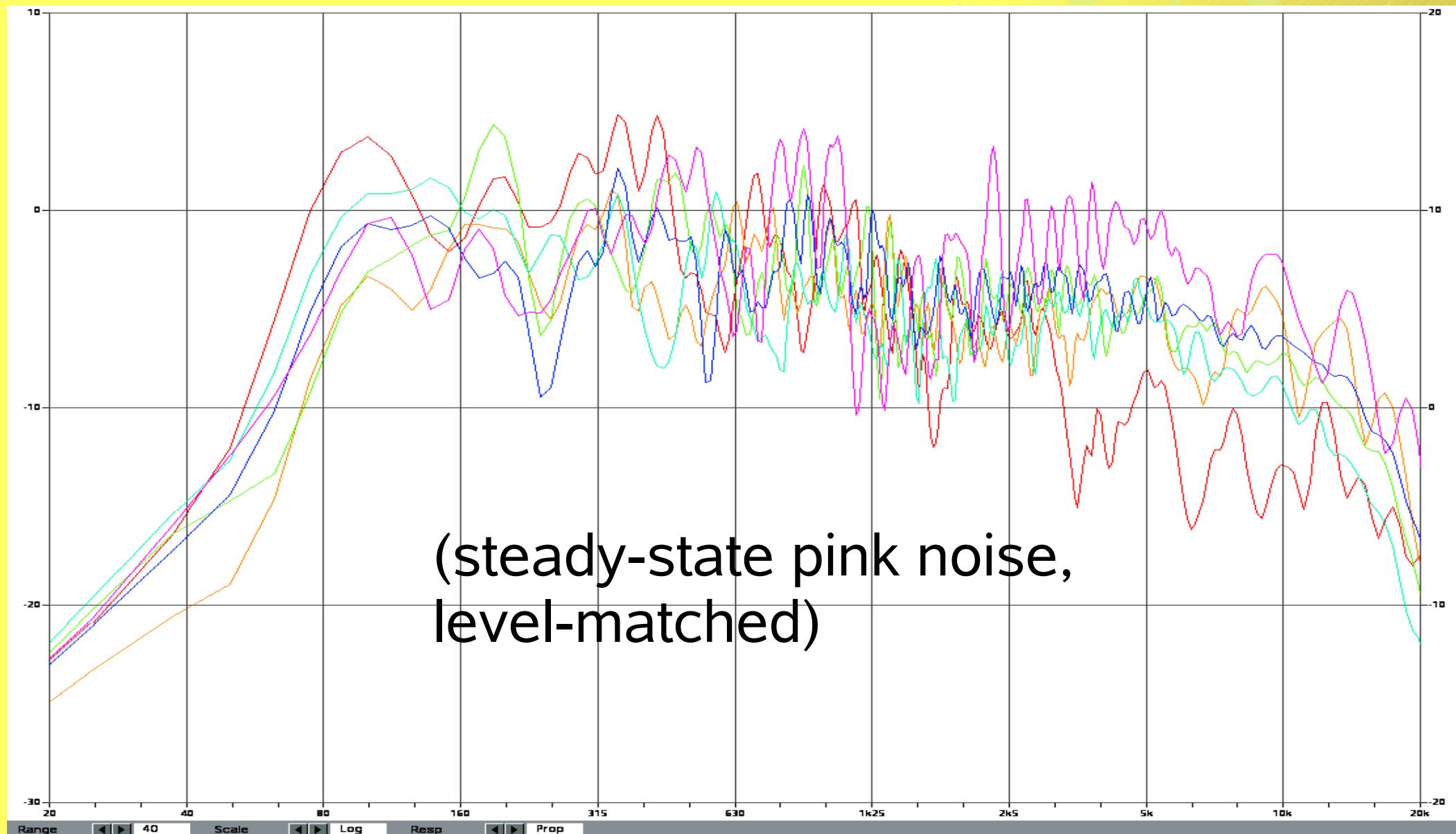
Speaker No. 6 before DRC:



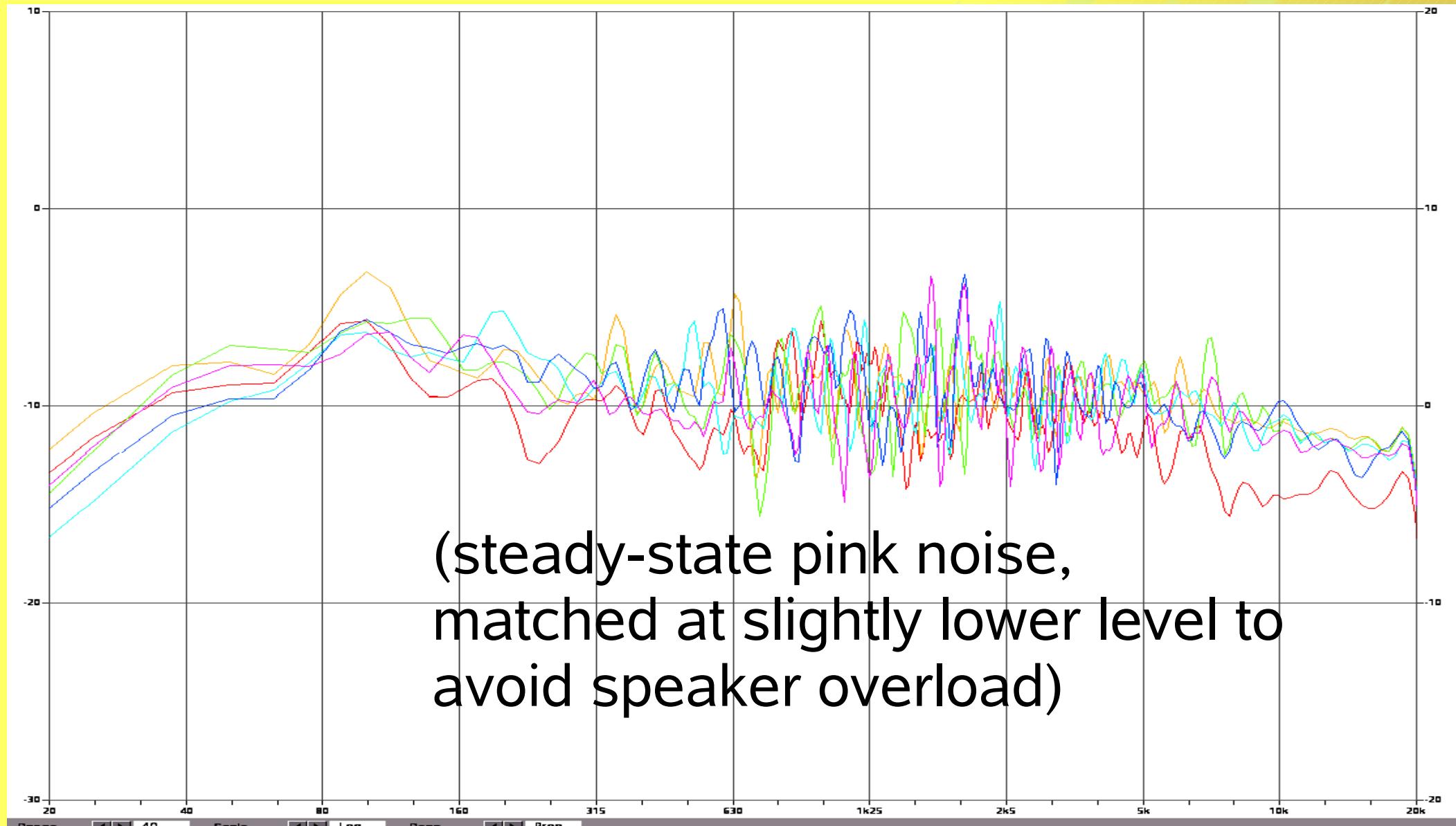
Speaker No. 6 after DRC:



Frequency response before DRC:



Frequency response after DRC:



Evaluation of the DRC:

Listening tests show **more pleasant timbre**, especially with stringed instruments, more „**sparkle**“ in the treble band, a „**tighter**“ bass and improved precision of **localisation**.

(The speakers used for this test are low-end active studio monitors with a cost of ca. 200€ per channel.)

Thanks to:

the VdT and especially Sebastian Gabler for the invitation,

Fons Adriaensen (author of Aliki, AmbDec and JConv),

Denis Sbragion (author of drc),

the JACK development team,

the Ardour development team,

Eric Benjamin, Richard Lee and other patrons of the sursound mailing list,

and of course

<http://www.have-a-great-time-in-south-australia.com> for the shark images.

Thanks for your attention!

Your Questions?



References:

Aliki, AmbDec, JConv, JAPA:

<http://kokkinizita.net>

drc: <http://drc-fir.sourceforge.net>

JACK: <http://jackaudio.org>

Ardour: <http://ardour.org>

qjackctl: <http://qjackctl.sourceforge.net>

sox: <http://sox.sourceforge.net>

Farina 2000: *Angelo Farina, Simultaneous measurement of impulse response and distortion with a swept-sine technique, AES Preprint, <http://pcfarina.eng.unipr.it/Public/Papers/134-AES00.PDF>, 2000*

*The corresponding paper to this presentation can be downloaded from
http://spunk.dnsalias.org/public_stuff/linux_audio/tmt08/*