Grimm | AUDIO

SPEAKERS

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Grimm Audio's LS1 is an unusual loudspeaker. Its wide but shallow cabinet is the direct opposite of nearly all contemporary loudspeakers and the DSP filter is mainly IIR based in spite of the availability of cheap DSP. Why? This white paper argues that all choices derive from basic acoustics, signal theory and psychoacoustics.

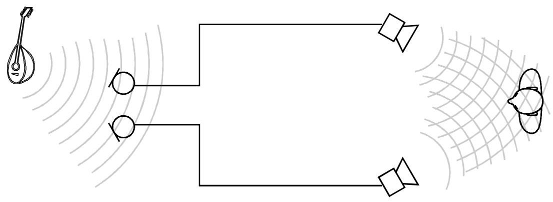
1 What on Earth Were We Thinking

1.1 The mission (1)

First things first. What is a loudspeaker for? Let's have a shot at a mission statement.

"A loudspeaker should aim for accurate reproduction in a practical listening room."

Really? Can we do that? Can we "reproduce" an original acoustic event? Clearly not, as this reminder of stereo theory shows:



Two microphones take two samples from a 3D sound field, and two loudspeakers generate a pair of interfering spherical fields. Clearly the two do not resemble each other. If the listener hears a mandolin somewhat to the right of the stereo image this is owed to the brain trying to interpret a very rare occurrence (two strongly correlated equidistant sources) as a more common one (one source somewhat to the right). Stereo reproduction has no meaning without psychoacoustics. More crucially, without it we cannot make reasonable statements about things like the best pickup pattern of microphones or the radiation patterns of loudspeakers. Notions that the perfect radiation pattern should be some geometrical ideal are highly suspicious.

1.2 An ill-posed problem

Things get worse. Practical rooms aren't anechoic. Few are even dry enough for early reflections and reverberation to be fully masked. Speaker-room interactions provide cues to the listener that all isn't right. For instance, during recording a highly directional instrument pointed straight at the microphones. That instrument will sound quite dry, while less directional ones in the same ensemble will sound wetter. Upon replay the listening room adds its own reflections and reverb, but only in function of the radiation pattern of the *speaker*. The listening room treats all instruments equally. This is very noticeable when listening to big-band music on omnidirectional speakers. The sound stage may be quite convincing but the brass section sounds ghostly, as if it's pointing in all directions. Well, it is. Not in the recording, but during replay. The percussion section in the meantime sounds fine. Making the loudspeaker extremely directive and strapping the listener into the hotseat is hardly a solution. It wouldn't work at low frequencies anyway. There's no unambiguously "right" radiation pattern for a speaker to have. There are only wrong ones and even more wrong ones.

Loudspeaker design is an "ill-posed problem" because some requirements are in conflict while others can't even be definitively stated. Working such problems into an acceptable compromise is done not by starting from what we want to get done, but from the things we want to avoid and prioritizing those.

1.3 The mission (2)

Let's first rewrite the mission statement:

"A loudspeaker should aim to create a credible illusion in a practical listening room"

We don't need to go into the finer detail of psychoacoustics to work out exactly how stereo works, because it's easy enough to see what will make the speakers give their presence away and kill the illusion.

Regardless of the room, speakers give themselves away by having a response significantly deviating from flat or by producing excessive amounts of non-linear distortion. A speaker that is good on these two counts already sounds great in an anechoic room. Placed in a real room a speaker betrays its presence by its off-axis radiation. Too wide dispersion causes the "omnidirectional brass section" effect while strongly frequency-dependent dispersion makes the direct/indirect balance frequency dependent as well.

1.4 Boiling down

The LS1 is designed as a transportable compact system. Dipole or cardioid cabinets are therefore out of the question. Accepting that the LS1 will be omnidirectional at low frequencies, we can not make high-frequency directivity too strong. What we can do is decide where to put the baffle-step frequency. Psychoacoustics tells us that below about 300Hz the ear will no longer clearly discern direct sound, first reflections or reverberant sound. Presumably the evolutionary background of this is that the vocal range starts here. Anyhow, we can live with a transition to omni below 300Hz. If we can get reasonable, constant directivity above that, we have a design spec!

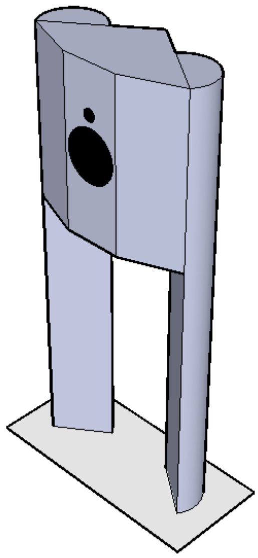
1.5 Not many options left

I don't think I've made any controversial assumptions so far. Yet, many commercial designs must be definitely wrong when held against this light. I once gave a talk about this where the slide show contained a photograph of such an obviously flawed design. I can't reproduce it here because it's actually on the market. I can describe it though. It looks perfectly normal by audiophile standards: a pair of 8" woofers in a 9" wide cabinet with a ribbon tweeter between them. Vertically, the wide spacing of the woofer results in strong lobing. The ribbon might have good horizontal dispersion (never omitted in data sheets), it is very directive vertically (never published). The narrow cabinet places the baffle step frequency in the middle of the vocal range, resulting in a dry top end and woolly low-mids.

Conversely, doing it right means: a single unit per "way", spaced closely together on a baffle of at least 20" wide. Now, wide baffles have a reputation of less than stellar imaging. This is because diffraction artefacts become audible as distinct reflections whereas on narrow baffles they are perceived only as colouration. The solution is simple enough: generously round off the edges.

1.6 Sketches of Speakers

Enough rambled. This is the first sketch.



The chief features of the LS1 are already in place: two half cylinders serve as legs and round off the baffle. Problem #1 is manufacture. Problem #2 is that the edge closest to the tweeter (the top) is not rounded. A second drawing with a cylinder stuck on the top earned the moniker "Iroquois speaker". The third attempt turned the enclosure into a flat rectangular box. Tweeter and woofer are reversed so now the bottom edge had to be rounded. The top could remain flat. Some diffraction from the top can be found, but the woofer is a greater source of diffractions on that side.

Next it was realised that the same extrusion used to make the legs could serve underneath as well.

Finally, paint texture and wood colour are selected. And of course, the famous laser-engraved logo.

1.7 Compromises, compromises and subs.

The LS1 is a compact system. This has its drawbacks. For most frequencies it's desirable to put the sound source more or less at ear height. The bottom end is the exception. Low frequencies emanating from a few feet above the floor will interfere mostly with their floor reflection, creating a dip in the low mids that is usually called "small speaker sound". A better place to put the low-frequency transducer would be on the floor. This is why the LS1 was designed to integrate well with a sub, and why a matching sub is on its way. The sub is preferably crossed high (100Hz or so), creating a true 3-way system of which the top part can be transported for location work.

1.8 Why the LS1 looks different from other 2-way speakers

This is why the LS1 looks the way it does. Why other 2-way speakers don't look like it is, frankly, a mystery.

2 Designing an active speaker

2.1 Optimal driver complement and cabinet design

The LS1 is aimed at the high-end segment, where passive systems still have a strong foot-hold. Several manufacturers respond by offering the same electro acoustic design with active and passive filters. This is not an optimal strategy. In passive systems, the cabinet is co-designed with the low-frequency driver to obtain a certain LF roll-off. The drivers are selected in function of their Thiele-Small parameters, relative efficiencies and response flatness. By this point the frequency response requirements have cast most of the design in stone. There isn't much freedom of choice left. In an active system on the other hand, response correction is cheap and efficiency differences are a non-issue. The designer can select drive units and cabinet size based on other factors: size, distortion, dispersion. A cabinet optimized for passive control does not benefit from active control while a cabinet optimized for active control is all but impossible to use with passive crossovers. The design procedure of active speakers does not run parallel with that of passive speakers. The LS1 is optimized for active control.

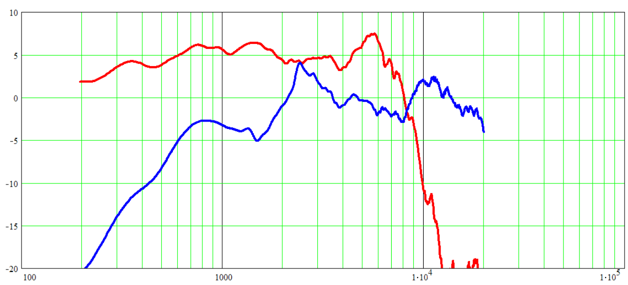
2.2 DSP control

The advantages of DSP based crossover filtering are generally listed as:

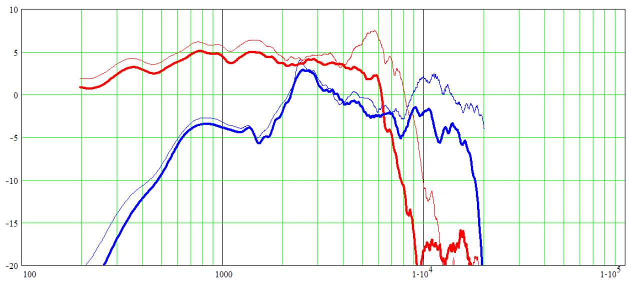
- Perfect impulse response
- Available from any speaker
- Automated design: measure, process, done!
- Ultra-sharp filters

Instead of arguing against these claims, let me just see where they lead. Here's a random DIY project for DSP to chew on:

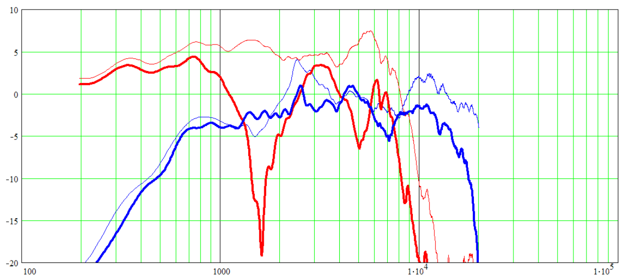
Classic Mid-Tweeter-Mid arrangement with decently performing drive units. The on-axis measurements reveal acceptable woofer performance and a decidedly ragged tweeter response.



The tweeter as measured on a large baffle looks quite fine, so the problem is not in the tweeter itself. Off-axis measurements may show what the causes are.

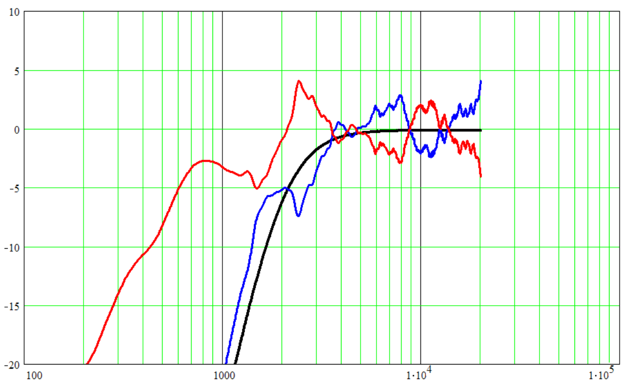


30 degrees horizontally, the woofer response is largely unchanged, except the cone breakup which is very directive. The tweeter response rolls off too, but notice the double peak around 10 kHz. In the off-axis measurement it is still visible, but it has shifted to a lower frequency. This indicates diffractions in the horizontal plane i.e. on the cabinet edges left and right of the tweeter.



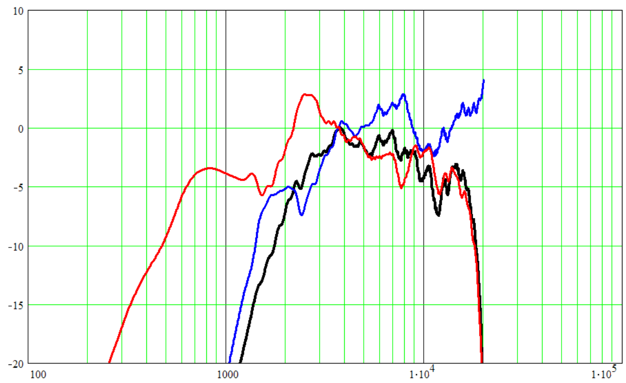
30 degrees vertically we find trouble. The large spike at 2.5 kHz has dropped and the dip at 8 kHz is now at 7 kHz. This points toward diffractions in the vertical plane. Such as there are? Well, the woofers, for one. Normal, non-MTM speakers also have noticeable diffraction against the woofers, but on an MTM it is twice as bad. The woofer response tends towards the useless, with a notch around the prospective crossover frequency.

But, we soldier on and let the DSP work its magic.

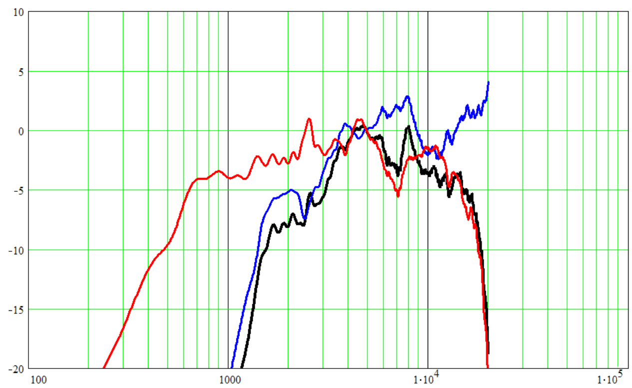


In red, the on-axis tweeter response. In blue, the filter calculated to correct the tweeter and leave a 4th order overall high-pass response. In black, the result. Quite amazing, considering that some of the ripples were caused by reflections. Can the DSP prevent those?

Remeasuring at 30 degrees will tell.



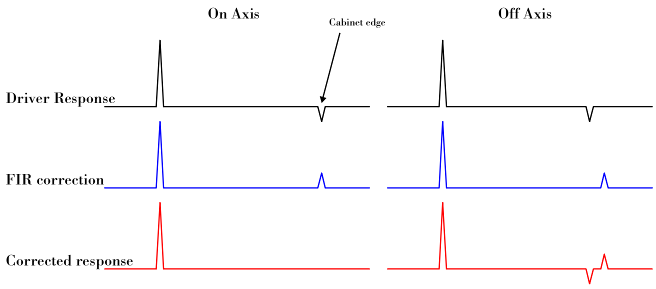
I take that as a “no”. The filter fills up holes by adding peaks and vice versa. Off-axis the 10 kHz error, after shifting down, gets compounded by the DSP filter, not corrected. The point is made again by the vertical off-axis measurement.



The error at 7 kHz has, if anything, gotten worse.

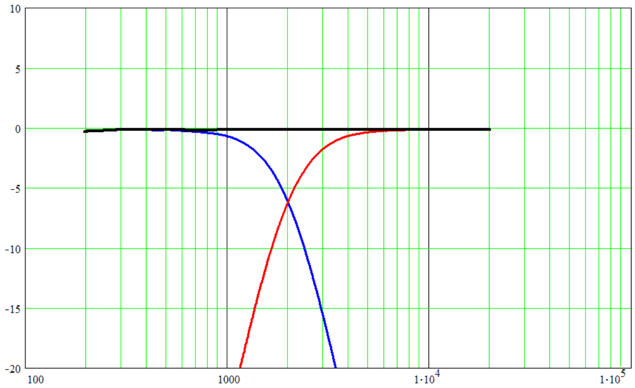
The problem is easiest to understand in the time domain. Shown in black is the impulse response of a driver and some time later an echo caused by diffraction where the baffle ends. In red the signal the correction filter sends the driver. It effectively makes the driver put out a mirror-image of the echo, and at the measurement position the two will cancel.

Away from the original measurement position, the diffraction no longer arrives together with the correction and both show up, one after the other.

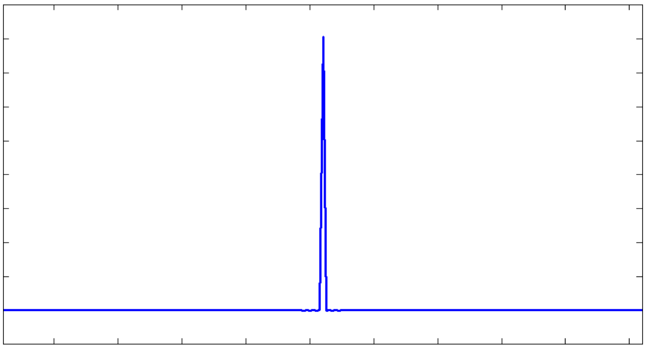


I probably shouldn’t need to take the story to its gory conclusion, but I must give in to human’s natural fascination watching a train collision in progress. It starts with the bumpers just a hair’s breadth apart and nothing has happened yet.

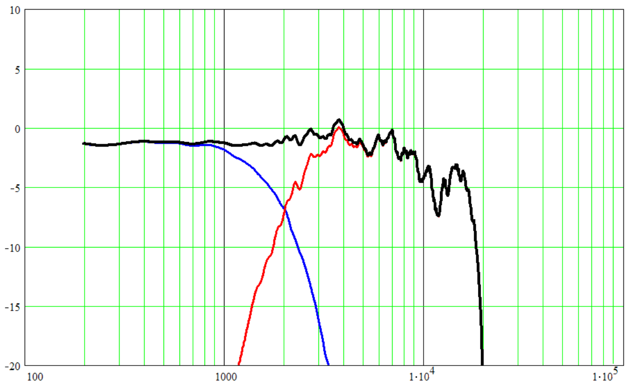
The on-axis sum looks brilliant.



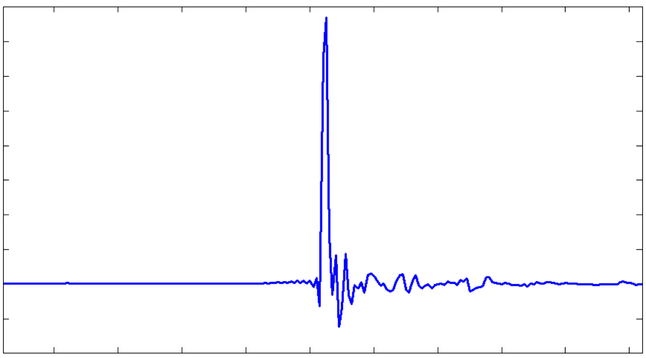
So does the impulse response, at which point the trains make first contact.



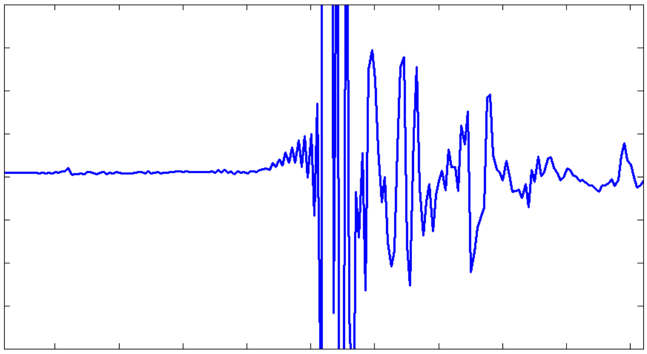
As the bumpers crumble, the front window cracks. Witness the horizontal off-axis sum.



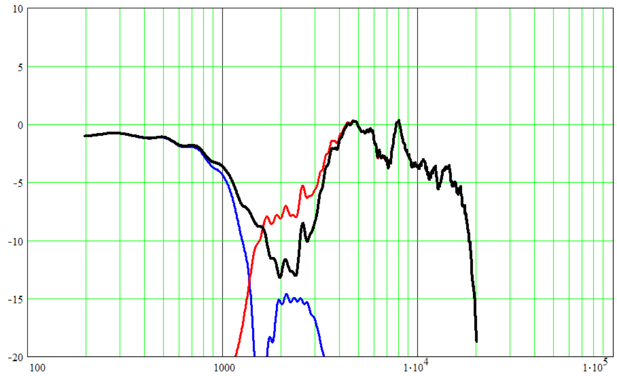
The corrections and diffractions have separated, resulting in a fairly jumbled post-echo.



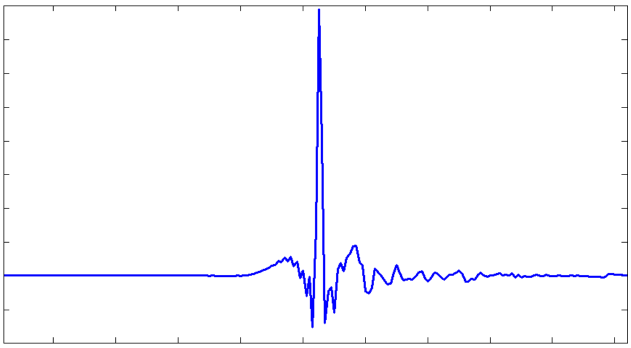
But, zooming in vertically, also a pre-echo (at the first horizontal division). Pre-masking may save the day, but this is the first indication that trying to get exactly linear phase has serious perils.



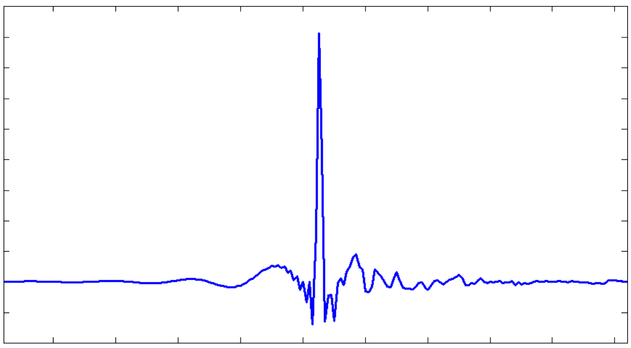
We measure the sum 30 degrees vertically and watch the engines explode in a shower of twisted metal while the undercarriage derails.



As the first train leaves the tracks completely and starts rolling down a slope, we look at the impulse response again.



Notice the bump in front of the impulse. This could or could not be a problem. It will be a problem if ultra-steep filters are used instead. Very sharp linear-phase filters cause pre-ringing, and out of the measurement axis the ringing of both drivers will no longer cancel:



The audibility of pre-ringing of >20 kHz filters in digital audio has always been a matter of some debate. Audibility of pre-ringing at 2 kHz is not.

The scene grinds to a halt and as the din rings out, a lone wheel detaches itself from the wreckage and bounces away into the distance.

2.3 DSP loudspeaker crossovers done right

From the above we've learned that

- Heavy-handed correction exacerbates acoustical problems
- Sharp, linear-phase filters cause pre-ringing
- Targeting an exact linear-phase sum can cause pre-echos.

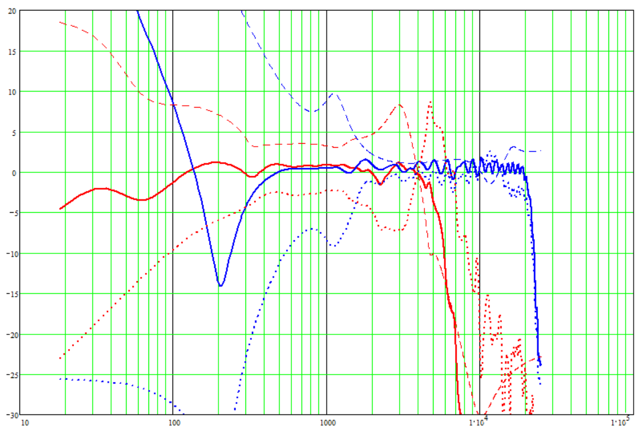
In short, brute-force correction sounds grainy and smudgy. When you hear cymbals go "splash" instead of "crash", it's naïve DSP at work. So:

- Do not shave off the hair, a nasty stubble will grow back.
- Do not correct beyond the very beginning of the impulse response.
- The gentler you correct, the wider the angle over which the correction still improves things.
- Target a minimum phase sum.

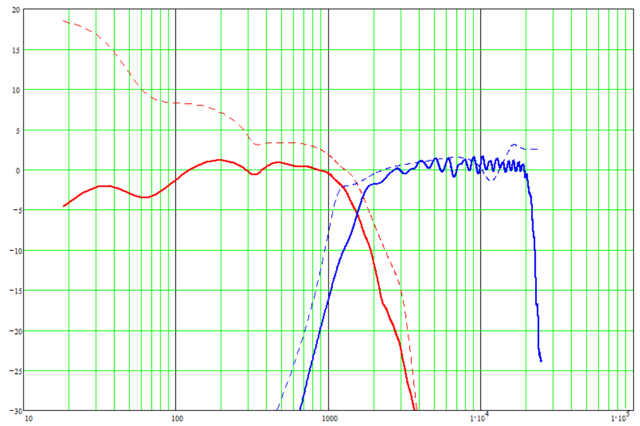
For the time being I would strongly recommend designing the correction manually. This rules out FIR as the main workhorse. For each bump or dip one corrects, one should know exactly where it comes from, and make sure that it isn't better corrected for acoustically. Unfortunately, designing DSP filters does not relieve one from having to know one's acoustics.

2.4 A practical procedure

First, both drivers are EQ'ed flat (again, within reason) over a frequency window well beyond their final operating range.



Next, LR4 filters are applied.



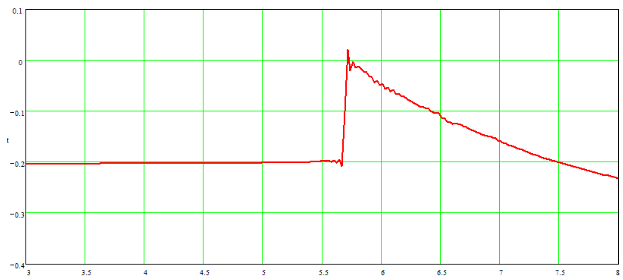
I must confess to an unconditional preference for Linkwitz-Riley crossover filters. I find it amazing that for an art like crossover design that has been practiced for so many years, it has taken so long before someone finally took time to sit down and work out the correct way of doing it. There is no excuse for anything less than full phase coherence between drivers so that they cross over at -6dB. It just doesn't get any better. Ever. 4th order roll-off will produce approximately 2nd order electrical slopes, which is enough for power handling.

Next, delay the tweeter to align it with the woofer.

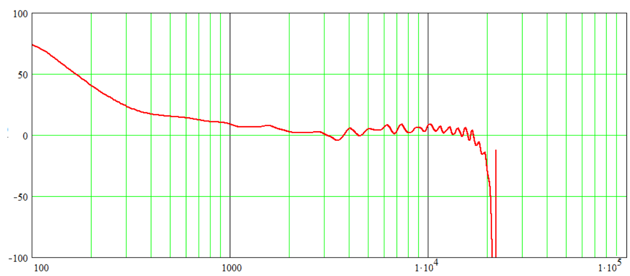
A sure-fire way of telling whether the first step has worked is when the delay corresponds exactly to the horizontal distance between the voice coils. After all, the first step removed group delay caused by the natural roll-off.

Finally, the icing on the cake. The sum of an ideal LR4 system is a second order all-pass with a Q of 0.7. In order to avoid the problems associated with correcting phase exactly, build an inverse all-pass filter based on the theoretical ideal. This filter will be non-causal so there's a good reason for using FIR.

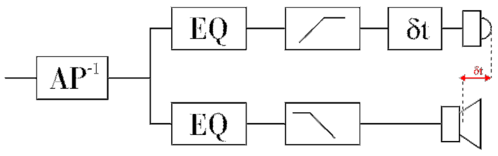
The step response of this system is precisely what we wanted: no pre-echo, nearly minimum phase.



Deviation from linear phase is gratifyingly "minimal".



And this is the block diagram of the DSP firmware:



3 Driver choice

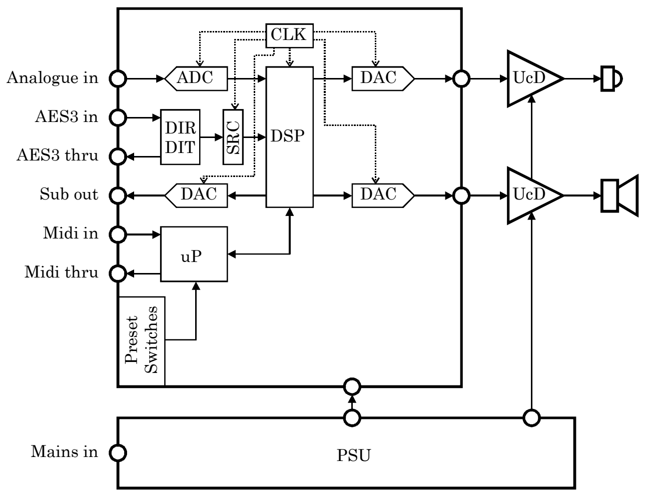
A fair bit of work went into selecting the best drivers. I'll spare you the story, suffice to say that the woofer is a 22cm Seas Excel and the tweeter a Seas 27mm DXT tweeter. They were selected on grounds of unit-to-unit consistence and distortion. The DXT tweeter has excellent directivity control.

4 Subwoofer integration

The DSP corrects the natural woofer/box response to a high-pass function of which the cut-off frequency is user selectable, as is Q. For standalone operation the cut-off frequency is set quite low (around 40Hz depending on material and SPL habits) and the Q to 0.7. For operation with a sub, Q is set to 0.5, resulting in an LR2 alignment. A subwoofer output is available on two pins of the Remote In connector. This signal is filtered to become the corresponding LR2 low-pass, so the internal filter in the subwoofer should be defeated. The subwoofer output can be time-aligned and level aligned using the remote control.

5 Electronics

The active circuitry is built on an insert sitting in the left leg (or the right leg should one wish to have the connectors facing forward). This isolates the PCB's from magnetic interference from the voice coils and mechanical vibrations in the cabinet. The electronics consists of four modules: a DSP card (including analogue I/O and connectors), a power supply module and two class D power amplifiers.



5.1 Conversion

The AD/DA converters were specifically selected for clean operation at moderate levels, more so than absolute SNR. No discernible distortion components are found above the noise floor for levels of -20dBFS or lower, which is the typical operating regime in a loudspeaker.

5.2 Clock

The clock circuit is the same as that used in the CC1 except that the sampling rate is set to 93.75kHz instead of one of the more traditional audio rates. This is specifically done to improve the performance of the SRC chip. An uncommon clock frequency reduces the odds that mix products between the incoming clock and the internal clock fall inside the PLL loop bandwidth of the SRC.

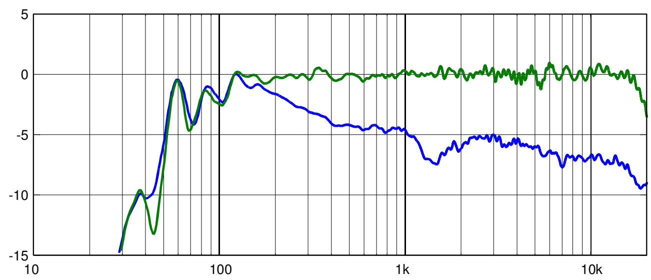
5.3 Power Amplification

The power amplifiers are UcD180 modules, slightly modified. Their frequency-independent distortion and extremely low output impedance minimizes their sonic footprint, a trait that has earned these amplifiers a following in mastering studios.

6 Results

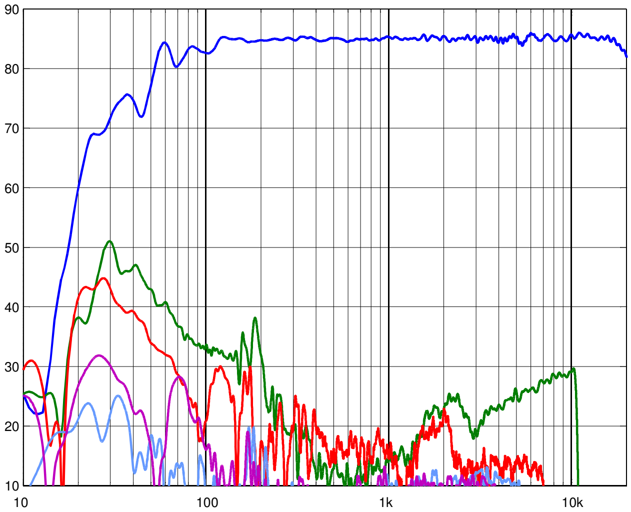
Measurements were made in the NOB anechoic room, Hilversum. The room is anechoic down to 150Hz so response data below that should be taken with a grain of salt.

6.1 Frequency and power response



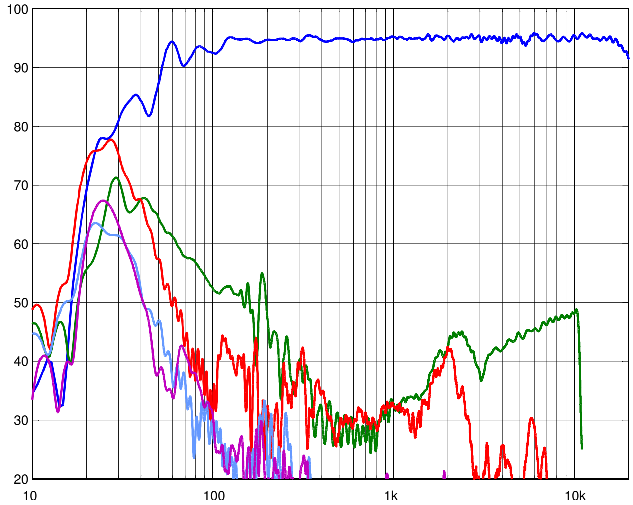
The power response was obtained by power-averaging responses over a horizontal and vertical orbit, weighted according to the solid angle each measurement represents. Clearly the off-axis response is well-controlled. The crossover dip is very mild and no further oddities are visible. This verifies that the steps taken in order to have reasonable directivity with no significant departures in function of frequency, were successful.

6.2 *Response and distortion at 85dB SPL*



Response and distortion measurements are made using an exponential sweep. Harmonic plots are represented on an x axis corresponding to the fundamental, not the frequency of the harmonic itself. Green=2nd, red=3rd, purple=4th, light blue=5th.

6.3 *Response and distortion at 95dB SPL*



At loud listening levels, the woofer is strained below 70Hz. For bass-heavy material replayed at high SPLs, a sub is definitely recommended. Otherwise clean performance.

7 Conclusions

The unusual route followed during the design phase of the LS1 is fully justified on technical grounds and by the performance of the final product.

Bruno Putzeys & Eelco Grimm