2017 ISEAT
International Symposium on ElectroAcoustic Technologies

Digital X-Over and DSP Simulations

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Abstract: The traditional way of designing cross-overs for high quality loudspeaker systems aims for a ruler-flat SPL response on axis. There is however no guarantee that this will sound good in a normal listening room. Extensive research including both blind listening tests and comprehensive measurements has shown that the radiated sound power from the loudspeaker system must be well-controlled. Controlling the off-axis responses is an efficient way for obtaining a controlled sound power. Modern software is very helpful in this process by being able to simulate DSP Biquads and Digital filters applied to both on- and off-axis measurements of the loudspeaker. Fine optimizations are possible by allowing non-standard filter coefficients for compensating real loudspeaker anomalies. A fully Digital loudspeaker requires one amplifier (IC) for each driver. Using a simple passive crossover plus DSP is a powerful combination which can still control the sound power while saving costs. By including the driver impedance, it is further possible to calculate the Safe Operating Area (SOA) for the driver combination.

Keywords: Digital cross-over, DSP simulations

1 HIGH QUALITY SPEAKER DESIGN

The traditional way of designing cross-overs for high quality loudspeaker systems aims for a ruler-flat SPL response on axis. There is however no guarantee that this will sound good in a normal listening room. Floyd Toole from Harman has made a very large study [1] of High Quality loudspeakers including both blind listening tests and comprehensive measurements. The curves in Fig. 1 shows on-axis and two off-axis responses for the described test where the worst sounding speakers had fidelity ratings 6.0 – 6.5. The 3 curves are not well separated and not smooth.

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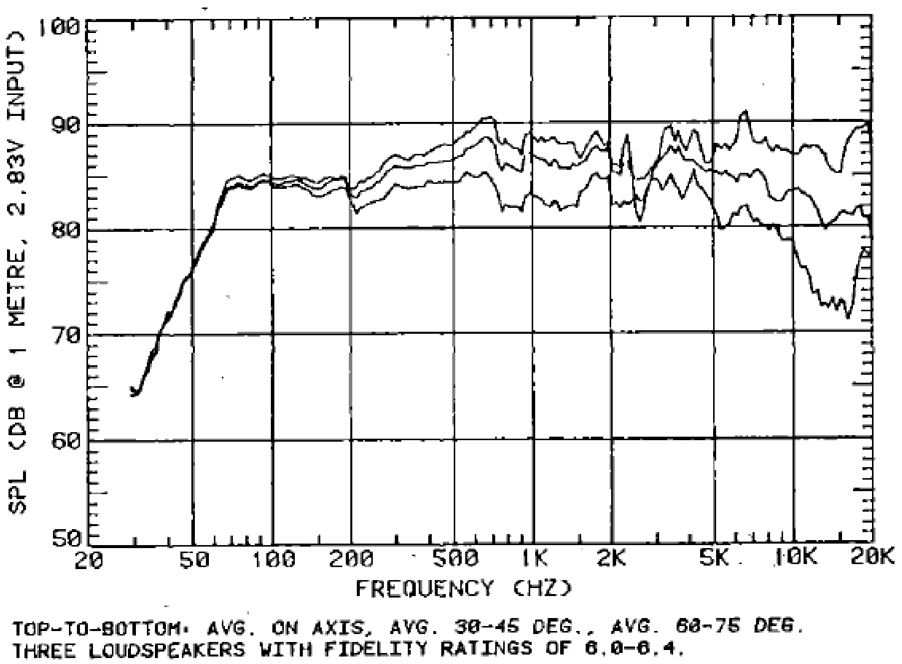
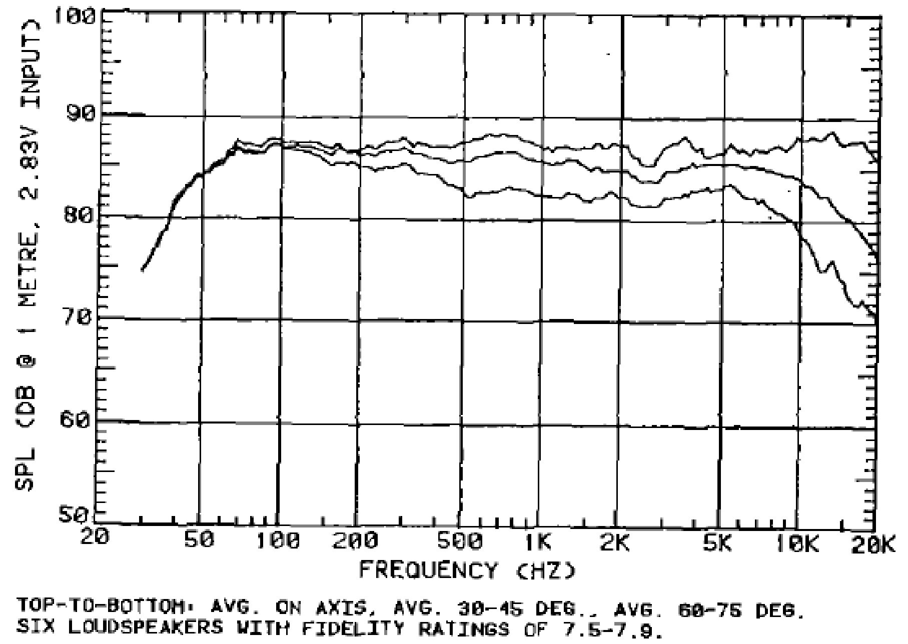


(a)

Fig.1 On-axis and two off-axis responses for the worst sounding speakers with fidelity ratings 6.0 – 6.5. (Floyd Toole)

In contrast Fig. 2 shows on-axis and two off-axis responses for the best sounding speakers in the test having the very highest fidelity ratings from 7.5 to 7.9. Here the on-axis response is very flat, and the off-axis responses are very smooth with a gently slope downwards from low to high frequencies.



(d)

Fig. 2 Amplitude response measurements of loudspeakers with highest fidelity ratings 7.5 to 7.9. (Floyd Toole)

This paper will show how to obtain similar good results using real Digital loudspeaker DSP simulations.

2 SIMPLE 2-WAY DSP X-OVER EXAMPLE

Let us start with a simple 2-way loudspeaker cross-over demonstrating the basics of Digital crossover design using the software FINE DSP [2]. Thereafter a more advanced example will be shown where many off-axis responses are used at the same time. (Power, Voltage and Xmax in Drivers & passive components). Before starting both drivers were placed in the actual cabinet and connected so that the drivers could be measured individually. The microphone was placed in the listening position in a quite large listening room, see Fig. 3. Then the driver responses were measured with phase one by one using a FFT based system with a Time window for excluding the reflections [3]. The resulting responses were therefore true loudspeaker responses, as if they were recorded in an anechoic chamber. The measurements were done with automatic delay ensuring that phase responses were absolute. Initially the microphone was placed on the listening axis, (usually) in line with the tweeter as shown in Fig. 3. After measuring the on-axis response the speaker is rotated around the vertical speaker axis (through the tweeter) by keeping the same distance all the time, and without moving the microphone. Keeping this the off-axis responses are measured for every 15 deg. (0-15-30-45deg). This is repeated for both drivers, including driver impedances. Normally you would need to specify the nominal impedance for a passive loudspeaker system, but that is not needed for this active (Digital/DSP) system. However the impedance can be quite useful for further power calculations, see later in Chapter (Power, Voltage and Xmax in Drivers & passive components).

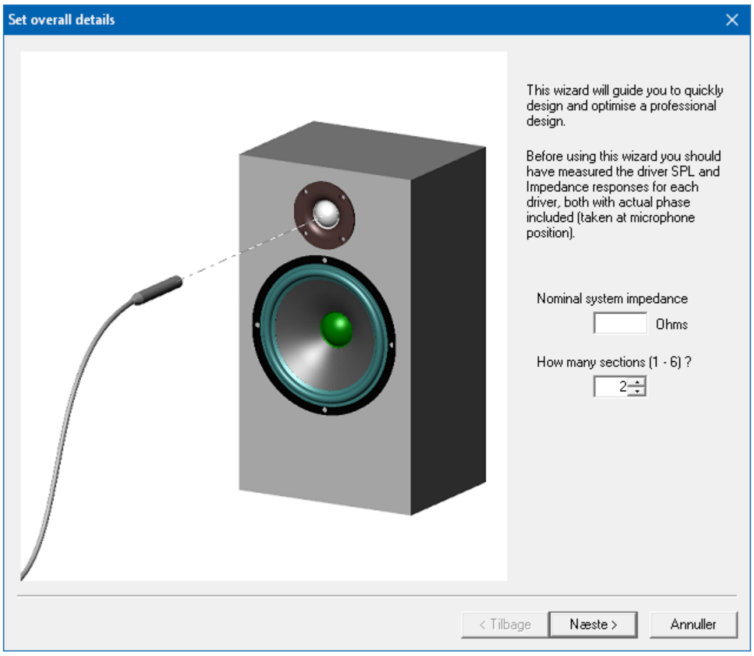


Fig.3 Measuring driver responses on the listening axis

All the on- and off-axis responses are now seen in Fig. 4 (Wizard). The imported measurements must have phase included, i.e. complete Transfer functions, which are going to be used in the full simulations, which includes the simulated transfer functions of the DSP elements.

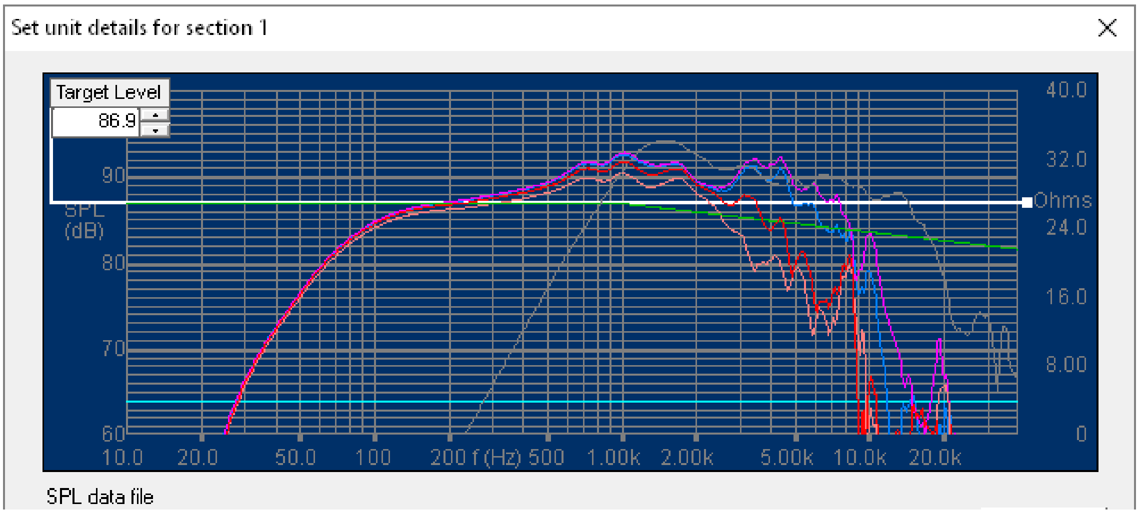


Fig.4 Imported Woofer responses 0-15-30-45 deg. off-axis

In this first example only the on-axis response was imported, however all the responses are seen in Fig. 4. The target level is indicated by a horizontal white line. Note that nearly all curves are sloping up around 6 dB (~200-1000Hz). That is caused by the relative small baffle of the actual speaker (Fig. 3). At low frequencies where the wavelength is much longer than the front baffle this loudspeaker is effectively radiating into the full sphere (4pi). At higher frequencies where the wavelength is comparable to the dimensions of the front baffle the speaker is only radiating into half space (2pi), causing the response to rise approximately 6dB. Above ~1 kHz the off-axis responses become different from the front radiation because the wavelength is becoming smaller than the effective diameter of the woofer piston. This is also clearly illustrated in the Woofer Contour Plot Fig. 5, where the red color is indicating the higher SPL up to ~2 kHz. Above this frequency the SPL is concentrating more on axis.

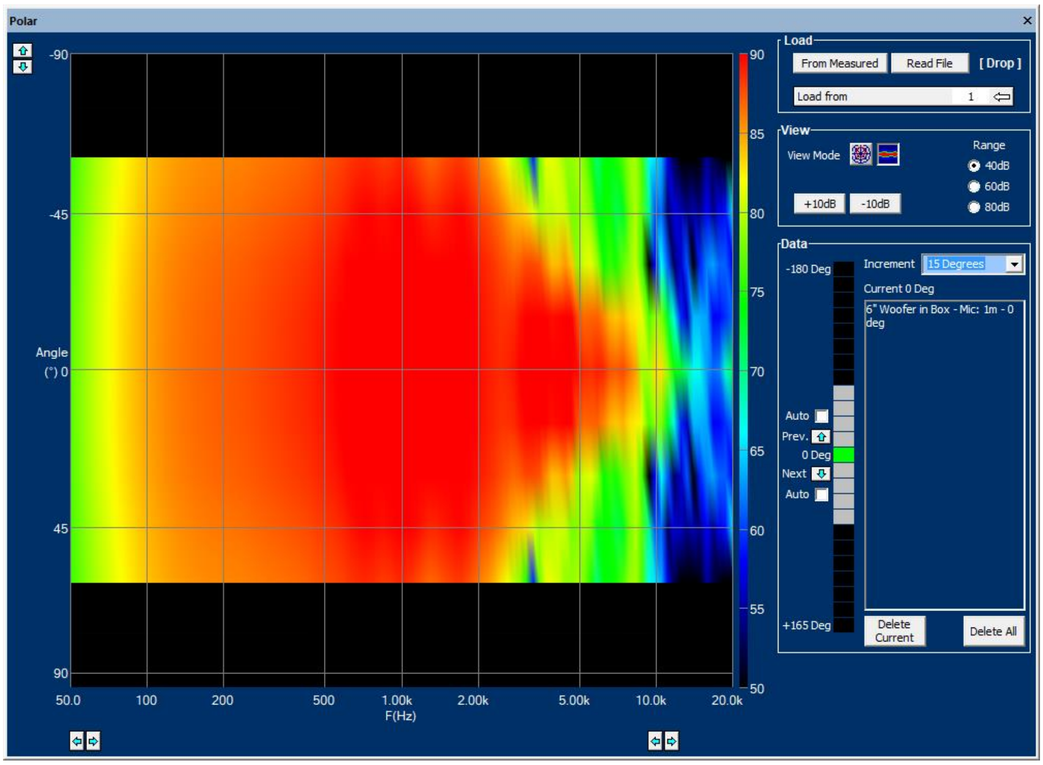


Fig.5 Contour Plot of the Woofer in box

In contrast the Tweeter Contour Plot Fig. 6 has high SPL from ~1kHz and then reduced dispersion above ~8kHz due to the dome size.

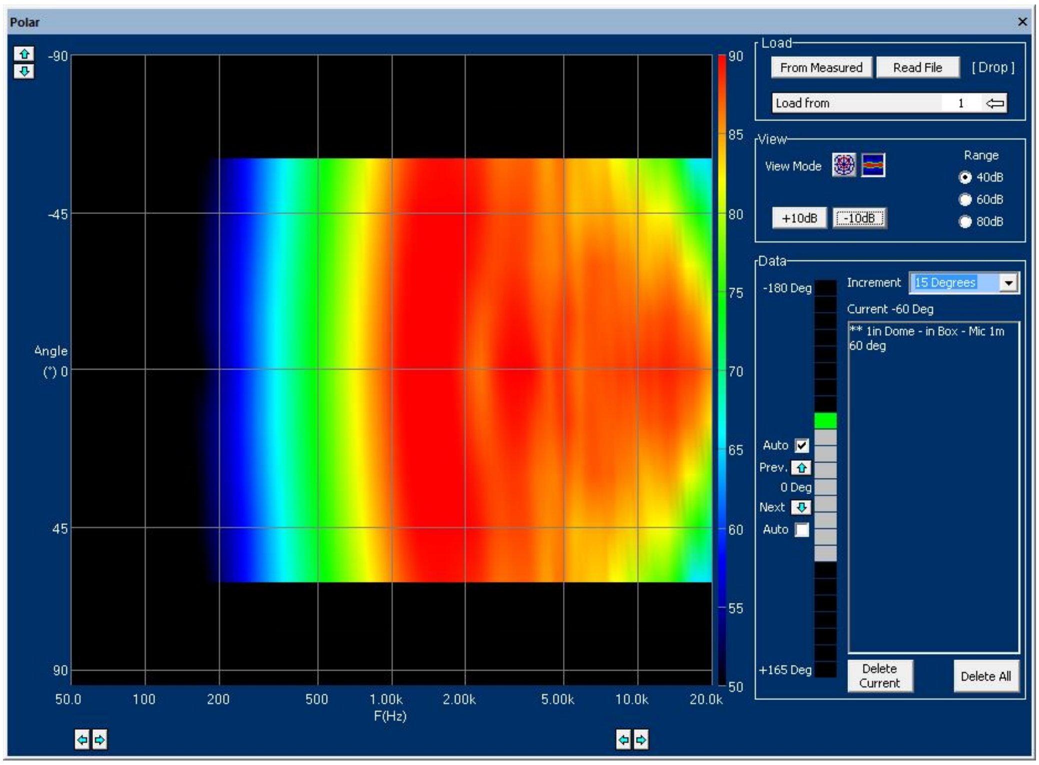


Fig.6 Contour Plot of the Tweeter in box

The purpose of the crossover is therefore combining these two quite different radiation patterns or contours, and at the same time protecting the tweeter from low frequency overload.

Using only the on-axis responses an optimized 4th order Butterworth DSP crossover at 1600 Hz was designed. The common IIR filters can produce very high cut-off slopes (Brickwall), by combining many Biquads, but that is not necessary when designing high quality loudspeaker systems. In addition the additional delay may cause problems. The rising woofer response is corrected using a low Q Shelf Element at 514 Hz, see Fig. 8. The tweeter needed more correction in the form of 4 Biquad Peak/Dip Elements. Only one at 3133 Hz is shown in Fig. 9. The optimized Q is 1.253, and the level -7.321 dB necessary to equalize a large peak in the response.In addition to the Peak/Dip elements in the tweeter branch, it was also necessary to insert a global Biquad Peak/Dip Element at 1645Hz, effecting the total (combined) response, having a Q of 2.541 and level -4.086 dB (Not shown).

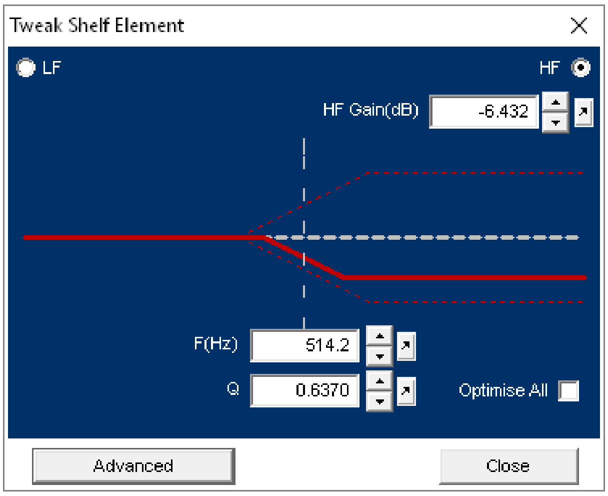


Fig.7 Woofer Shelf Element at 514 Hz

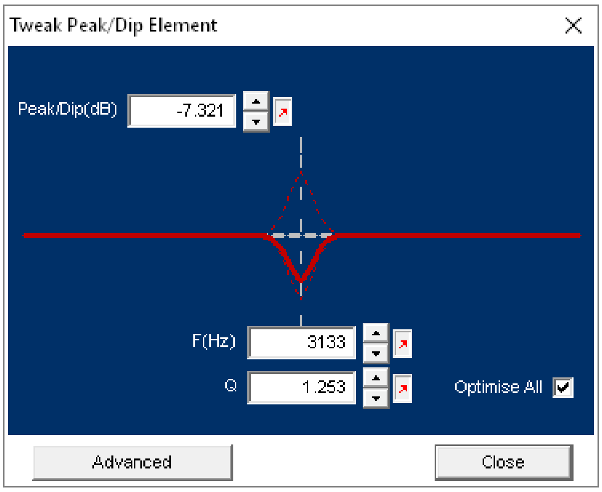


Fig.8 Biquad Peak/Dip Element at 3133 Hz

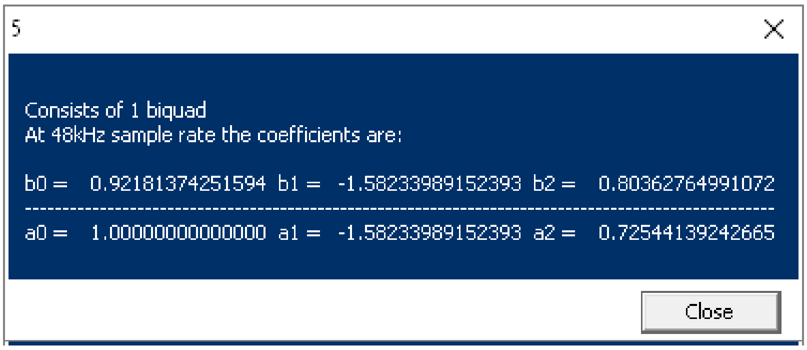


Fig.9 Biquad Coefficients for the 3133Hz Peak/Dip filter

The response of the final optimized crossover using only on-axis responses is shown as Fig. 10. The resulting crossover produced a very flat on-axis response, see the white curve in Fig. 10. However the off-axis responses are also shown, but not (yet) used in the optimization. They are far from smooth, but actually higher than the on-axis curve around 2000 Hz and not smoothly sloping down as for the high quality speakers shown in Fig. 2.

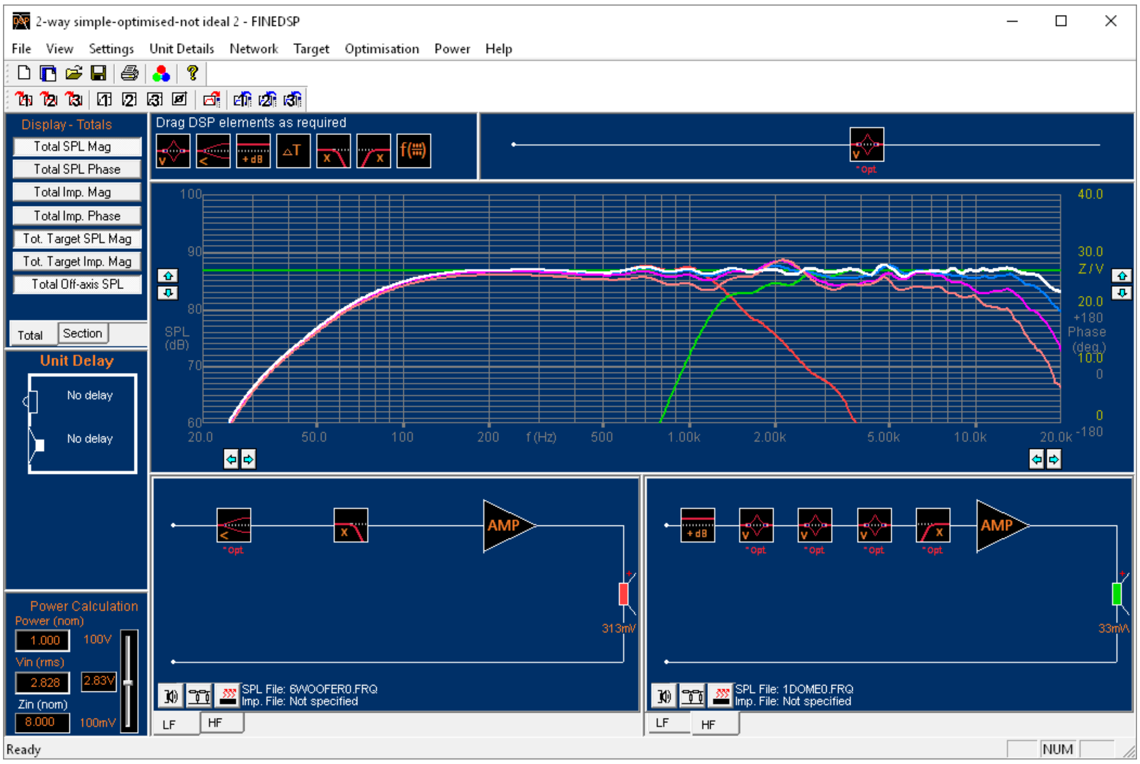


Fig.10 Two way optimized cross over response using only on-axis SPL. Off-axis responses are shown but not used for optimization.

3 ADVANCED 2-WAY EXAMPLE WITH OFF-AXIS RESPONSE

The method used in the previous example is clearly not the ideal approach. The next picture Fig. 11 is an advanced example using the same two drivers however now including on axis as well as 15-30-45 off-axis responses measured in the actual cabinet.Realising that the on-axis response is only valid for one point on the listening axis, and that a major part of the perceived sound is reflected from the room boundaries it is an advantage first controlling the off-axis behavior. Therefore the 30deg response was chosen for optimization.This time several DSP elements has been applied, both in each section and also globally Fig. 10. Like in the first simple example here is also used DSP elements as 4th order Butterworth (BU4) x-over, a shelf in the woofer section and tweeter level (dB). A couple of Tweak Peak/Dip DSP elements were inserted at 1510 Hz and 17.4 kHz in the tweeter section.

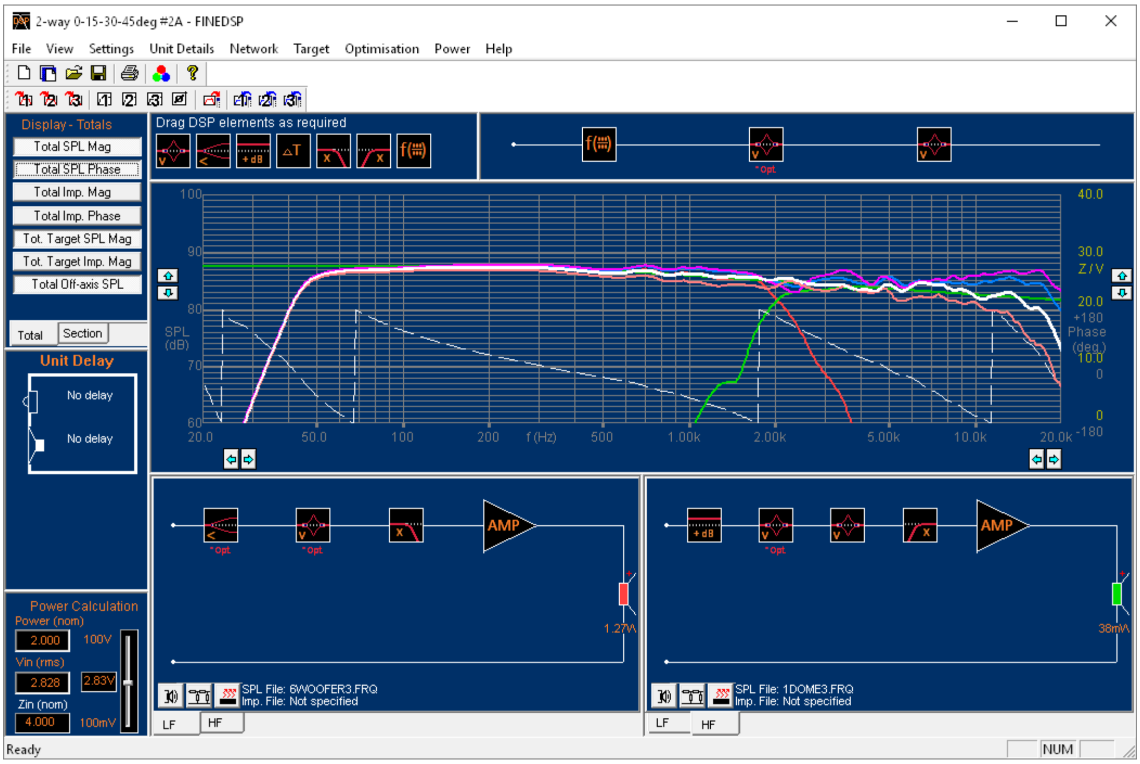


Fig.11 Advanced 2-way example with 15-30-45 degree off-axis responses. The 30deg curve (White) was optimized. The resulting Phase Response is shown as a dashed white line.

Looking at the 4 curves in Fig. 11 we notice that 3 of the responses are close together, whereas the on-axis (Magenta) shows a bump. This was intentional as controlling the off-axis responses is more important than the on-axis. By moving the X-over frequency up/down this may be further improved. The Total Acoustic Phase is included as the white dashed curve. Considering that good phase curves are parallel this phase response is good. Actually the phase curve may be further improved by introducing a Digital Delay (deltaT button), but that is not considered here.The System Target for the 30deg off-axis curve is shown as the green dashed curve, which is gently sloping down. This is specified in Fig. 12. The target sensitivity 85.8 dB (1 khz) is set to match the woofer level 200-300Hz, which is the piston range before cone break-up. The HF Slope is -1.0 dB/Oct from 300Hz, corresponding to the change in radiation seen in the Contour Plots, Fig. 5 and 6.

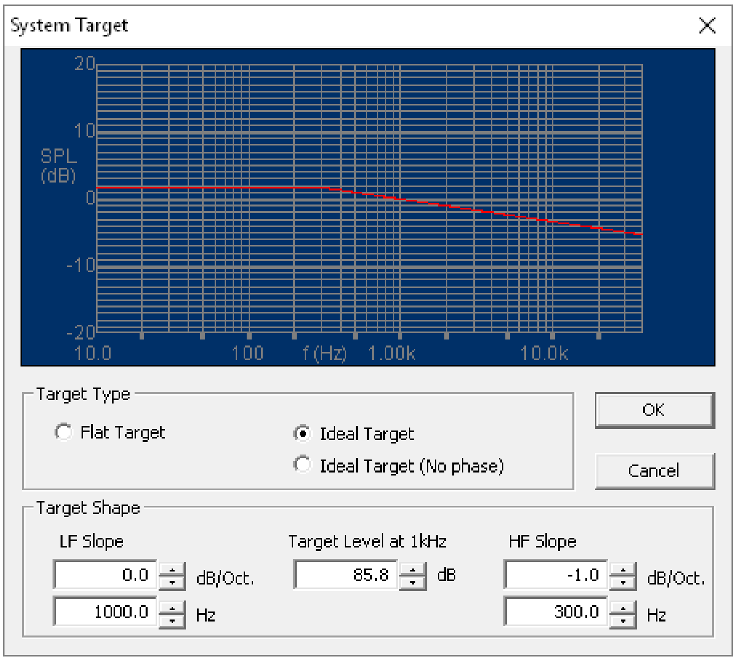


Fig.12 System Target curve set for the 30deg off-axis curve

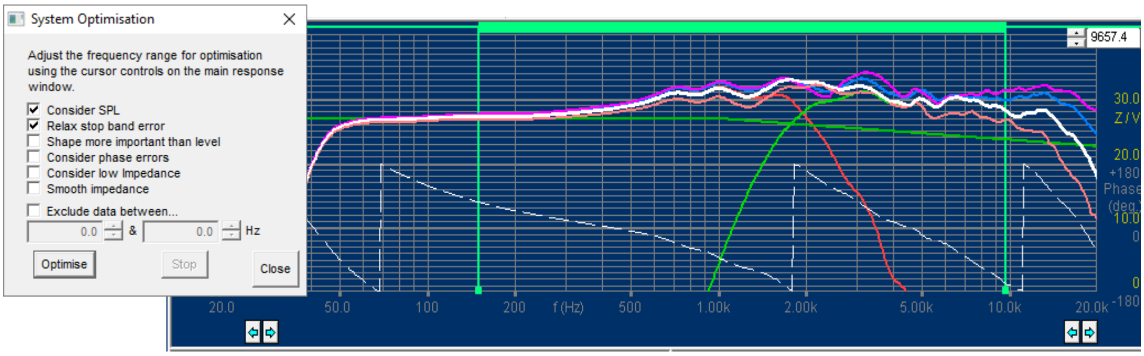


Fig.13 System Optimization

The total system may be optimized automatically. When System Optimization is selected a range is shown with green bars. This range must be selected with consideration. The imported responses were measured at a certain distance and represents the direct radiated sound from the loudspeaker, since they are recorded with the FFT technique which leaves out the reflections from the room.The time window for such measurement in normal (large) rooms seldom allows a time range more than 10mS, corresponding to 100 Hz. However in this example the low frequency range was measured using the powerful nearfield method [x], and then spliced together with the farfield curve(s). This way even a bass reflex system can be measured well down below 20 Hz.

Therefore the optimization range is set from 150 Hz, and up to appr. 10 kHz. Compensating a limited high frequency tweeter response too much is not adviseable as that may sound bad.In this example the crossover frequency was not set for optimization a.o due to power rating of the tweeter (see also later). Then the selected DSP elements can be optimized, Fig. 13.

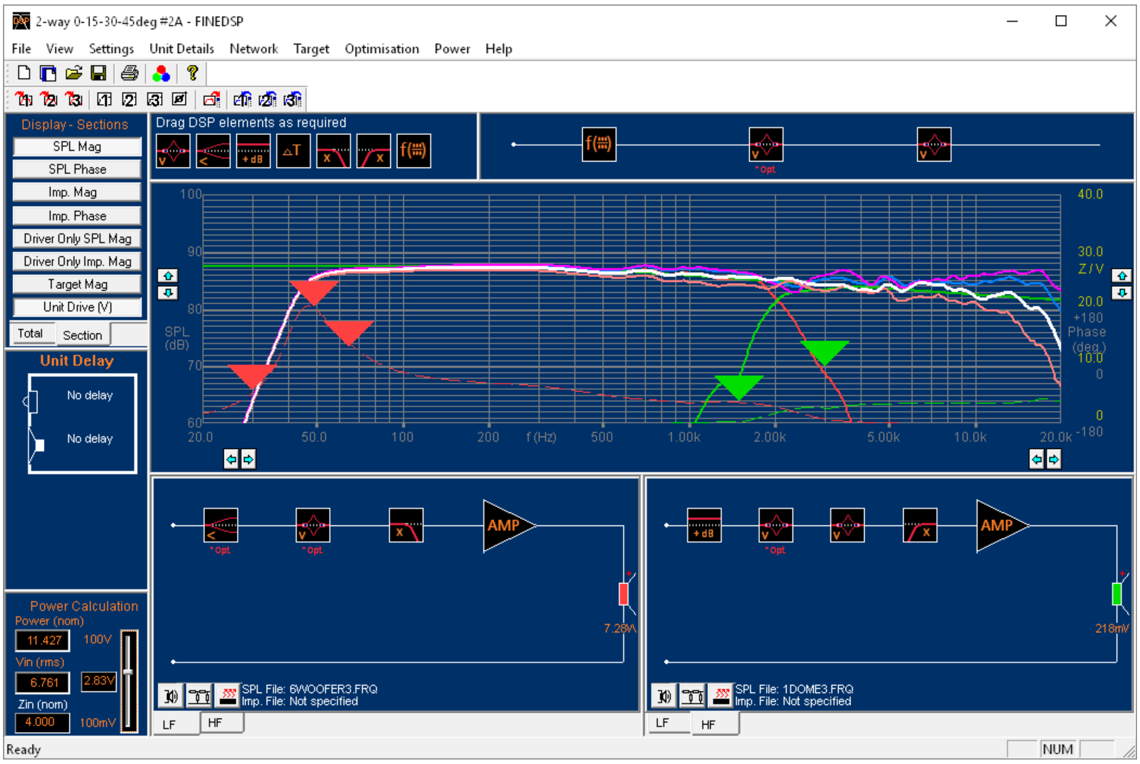


Fig.14 Optimized 0-15-30-45 deg. 2-way System with Power Limits

In this case the 30deg (White) response was optimized for the dashed (green) target, Fig. 13 and 14. The final on- and off-axis curves were all exported into the measurement system. Based on this the total radiated sound power in the current angle was calculated. This is shown in Fig. 15. Since further data was available the sound power was calculated for the total angle of 120deg (+/-60deg). The response is very smooth, gently sloping down towards the high frequencies. This is quite close to the recommended response for a high quality 2-way loudspeaker system, and demonstrates that controlling the off-axis responses is an efficient method for obtaining a controlled sound power. Using this information the contour plot was calculated for the total angle of 120deg (+/-60deg), Fig 16. Comparing with the driver contour plots this is remarkably even, showing very good dispersion. Actually a little problem appears at 2 kHz, where both the sound power response and the contour plots show a little dip. Ideally that may be corrected in the crossover. But compensating for errors only appearing above 45deg may not be adviseable, as the main listening window is most critical.

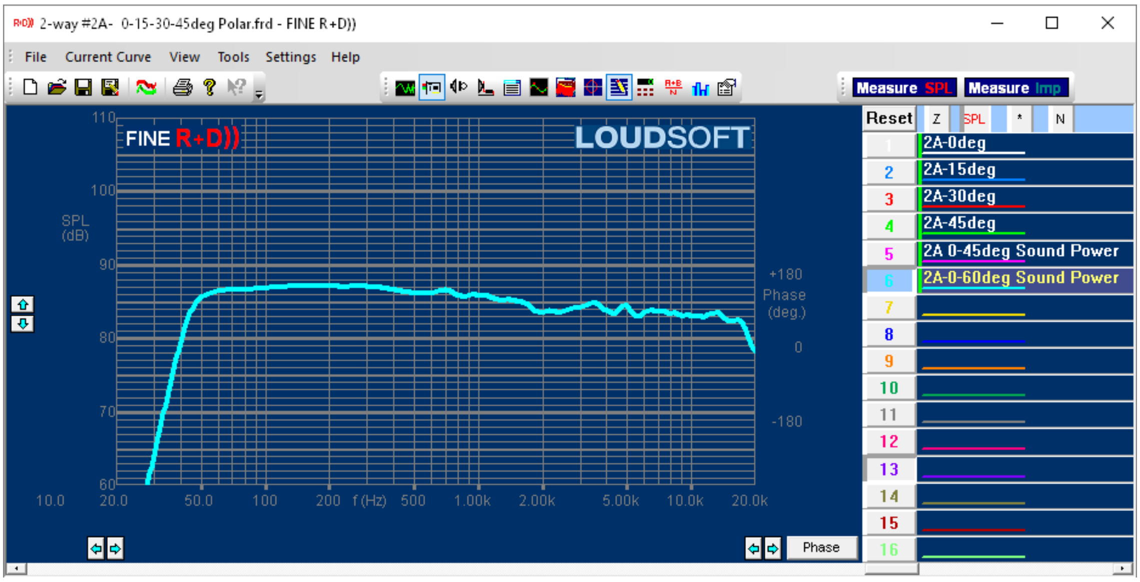


Fig.15 Radiated Sound Power of the final 2-way Loudspeaker calculated over 120deg (+/-60deg)

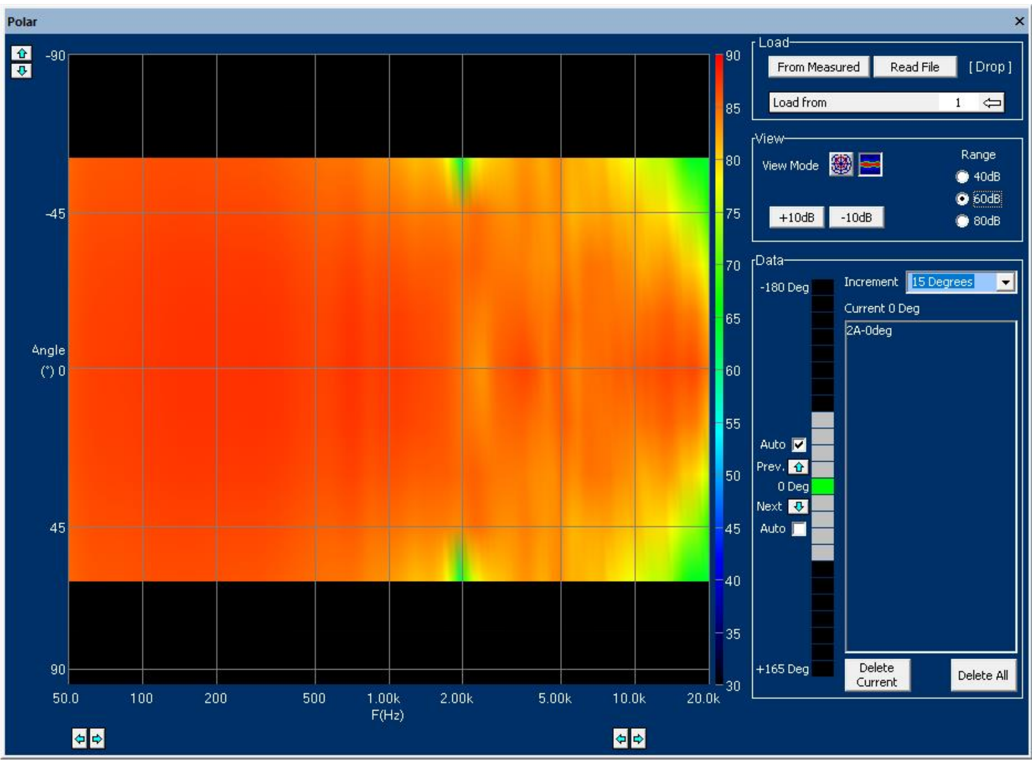


Fig.16 Contour Plot of the final 2-way Loudspeaker calculated for the total angle of 120deg (+/-60deg)

4 POWER, VOLTAGE AND X$_{MAX}$ IN DRIVERS AND PASSIVE COMPONENTS

The actual power in the woofer (4.98W) (Fig. 14) and tweeter (270mW) is now calculated accurately because the measured impedance was used. Both calculated acoustical phase, electrical phase and impedance was calculated and displayed.When considering the maximum power input into the system we need to know the Voice Coil Displacement or Excursion. The maximum is Xmax.Unfortunately the voltage to obtain Xmax for each driver depends very much on the frequency, see Fig. 17.

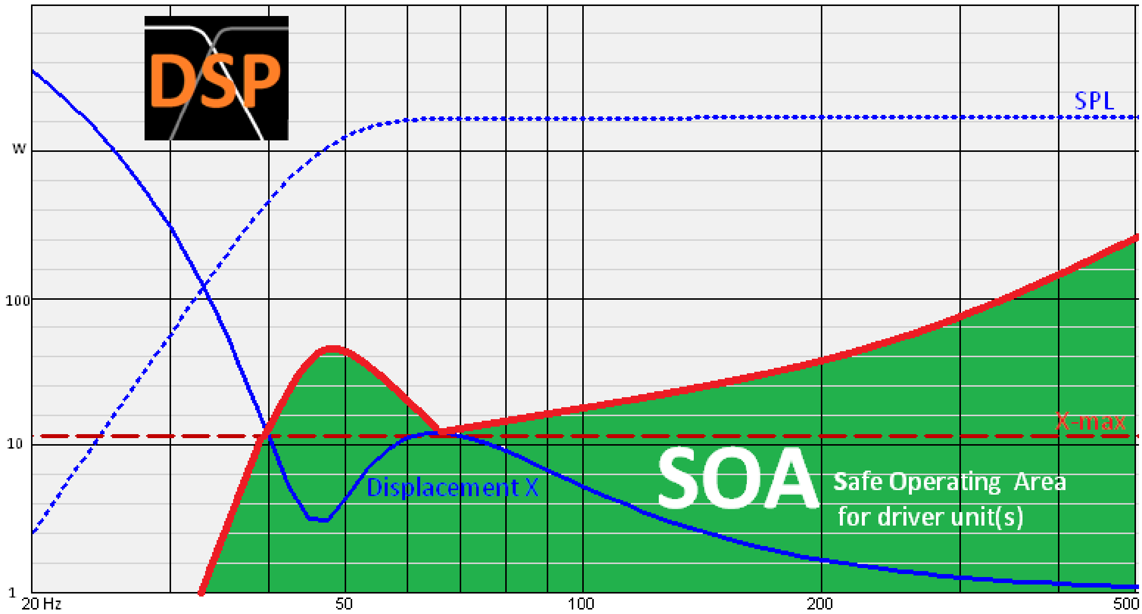


Fig.17 SOA (Safe Operating Area) for driver control with DSP

Fig. 17 shows the typical Voice Coil (VC) Displacement for a woofer in a bass reflex box or ABR system. The upper dotted blue curve is the bass frequency response, and the full blue curve is the typical (VC) Displacement of the woofer. Note that the Displacement is quite high at very low frequencies (<40 Hz), has a minimum around 45 Hz, further shows a local maximum at 55 Hz, and decreases towards higher frequencies. The bass reflex system was tuned to 45Hz (=Fb) in this example, which means that the woofer cone and VC movement is much reduced at this frequency compared to 55 Hz. Therefore we can optimize the low frequency response using this kind of information. In the 2-way system Fig. 14 was inserted at 2nd order High Pass (HP) filter at 44.1 Hz to attenuate the very low frequencies where the total response has dropped anyway, Fig. 18. The HP filter was specified as a general DSP function, which was dragged into the global field。

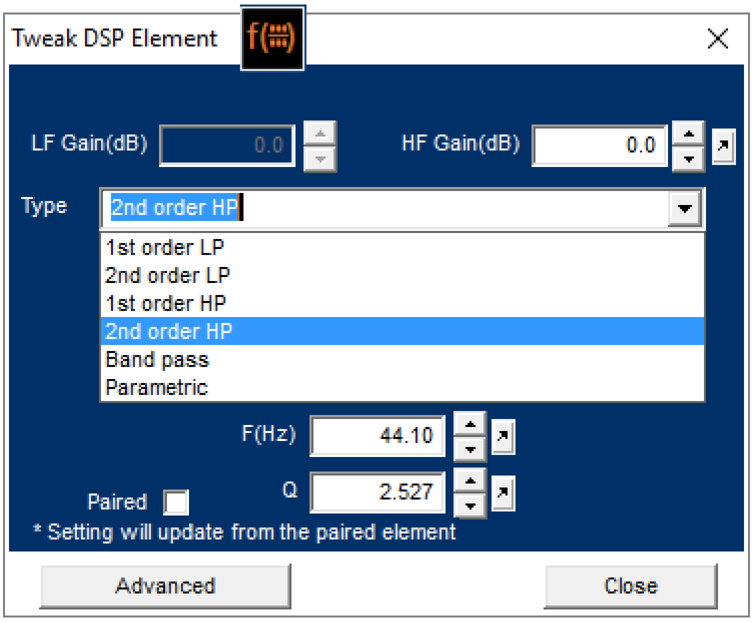


Fig.18 General DSP Element with 2nd order HP selected

One advantage of this HP function is that the Q can be independently specified. In this case the Q is set as high as 2.527. In addition a (Tweak/)Peak was placed at 60Hz just where the displacement is low for this particular system. It was set + 2dB, for boosting and nicely extending the bass together with the HP filter, see Fig. 19. These two features will improve the system by giving more and deeper bass, and at the same time protecting the woofer at very low frequency (<44Hz), where the response is dropping anyway.

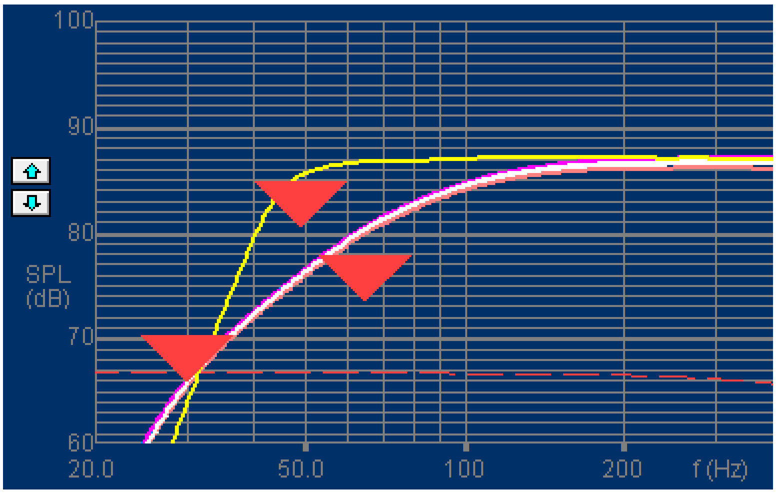


Fig. 19 Low frequency response with Boost (yellow) / without. Red arrows indicating max level

The question is how much Power or Voltage can be applied at the low frequencies? Referring to the general displacement given in the SOA Fig. 17, the maximum unit drive voltages may be specified. Using a sinewave input at 30 Hz, we found that 5.9V is enough for the woofer to reach Xmax. (= The VC displacement/travel where the VC is still in the air gap). This is shown in Fig. 17, where we have also measured the voltage for Xmax at the tuning frequency 49Hz. At this frequency the woofer can accept 20.4V, which is very high. But only 13.5V at 65Hz, which is now the frequency with maximum displacement (see Fig. 20).

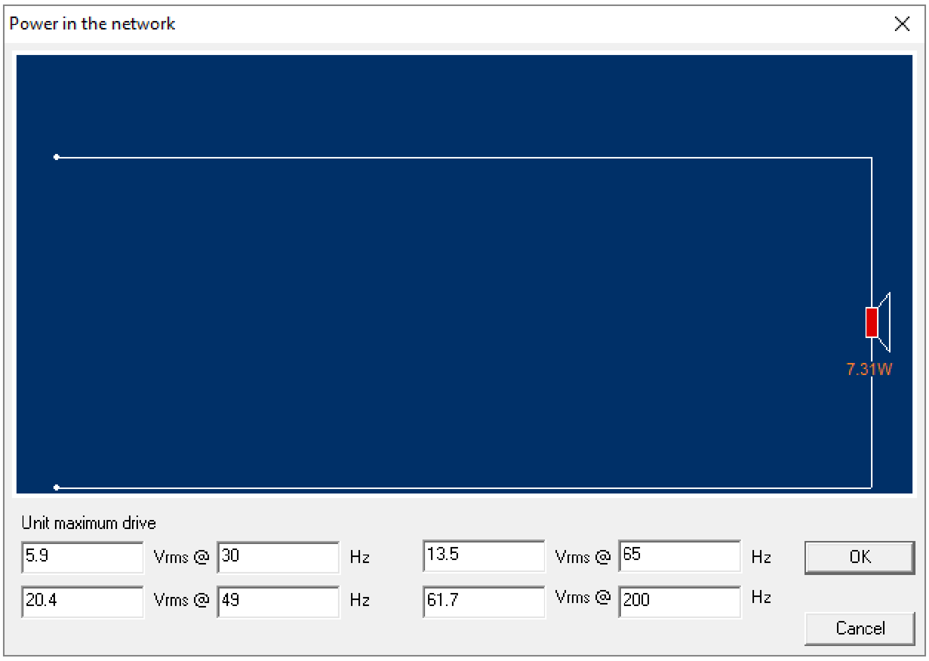


Fig.20 Input Voltages for Xmax at various frequencies

This information can also be derived by simulation software, for example FINEBox [4].Using a special setting, Fig. 20 the main display will show red arrows indicating the maximum voltage points we have just specified. That is already shown in Fig. 14 and Fig. 19, which also contains the output voltage from the amplifier, shown as the dashed red line for the woofer. When we increase the Output Power (see lower left in Fig. 7) this output voltage is accurately calculated by FINE DSP. The display shows that 6.76V (11.4W) will reach Xmax around 49 Hz. If for example the amplifier is 20W or larger, we have to turn down the +2dB boost accordingly as well as the high Q (=2.53) of the HP filter @44Hz when playing loud. However the amplifier may not be able to supply enough current. Instead of voltage the Unit drive can be set to specify current at 4 frequencies. An example is shown in Fig. 21. In this case it is necessary to import the impedance curve for providing an updated calculation.

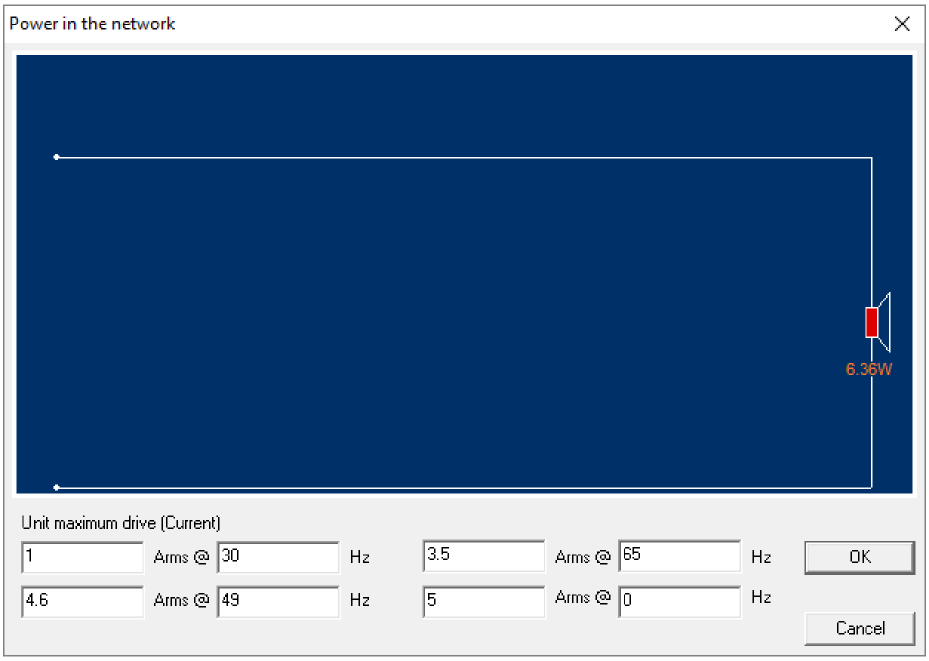


Fig.21 Max Input Current at various frequencies

The same feature is used to protect the tweeter at its resonance frequency 1500 Hz. In general tweeters can only accept a small voltage at resonance (Fs), in this case about 4V. Specifying the voltage at Fs will similarly display green arrows giving maximum voltage drive for the tweeter.

In this case the tweeter is not reaching maximum yet. But that may change if the x-over frequency is set too low. The Unit Drive curve is then very useful for avoiding overloading the tweeter when choosing the final x-over frequency and slope.

5 HYBRID- PASSIVE/ACTIVE DIGITAL DSP EXAMPLE

As the last example both passive and DSP components are used. By using only one amplifier there is a cost saving. Then we need to insert a passive X-over circuit and use global DSP EQ to equalize the total response, see Fig. 22.This time the measured impedance response was also imported. Using the Wizard a 12 dB/oct. passive filter with a X-over frequency of 1500 Hz was chosen, giving good separation between woofer and tweeter. Note that all the passive components can be editited as needed. For example the large 2.7 mH inductor in series with the woofer, could be changed to a less expensive 1mH. The changed response can still be optimized with DSP at no extra cost.

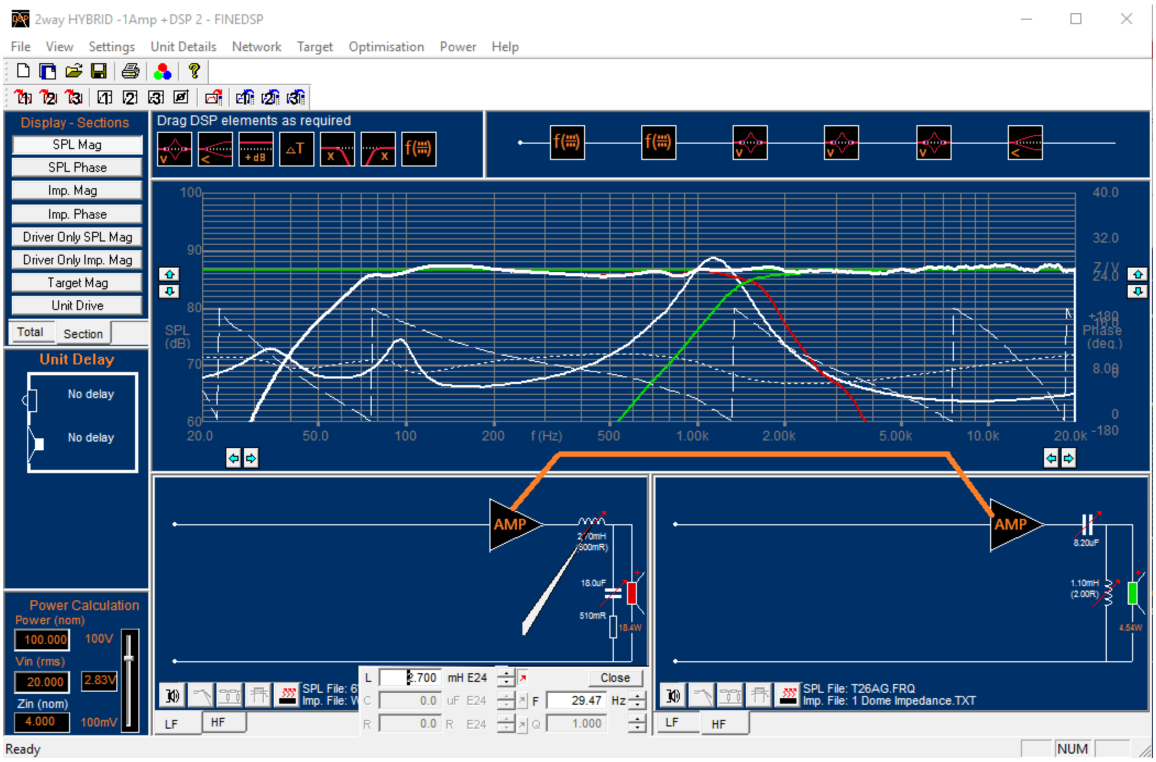


Fig.22 Hybrid - Passive/Active Digital Speaker with DSP. The amplifier(s) are combined by using one only (shown w orange line)

6 DELAYS

Most practical DSP IC's use Biquads which are based on IIR filters [5]. IIR filters have the advantage of being able to implement designs with sharp features with much less computation than the equivalent FIR filter, but with the drawback of not being able to implement linear-phase designs. Often this shows as (unwanted) digital delays. Such delays are common, and depends very much on the number of Biquads used. High order slopes requering many Biquads are not necessary (see page 7) when designing high quality loudspeakers systems. Measuring or specifying this delay is difficult, and it is therefore recommended first measuring for example a tweeter response, and then comparing with a simulated X-over with DSP for this.It is possible to introduce delays in two ways. Digital and Acoustical Delays. Fig. 23 shows a 3 sample digital delay using the standard 48 kHz sampling frequency.

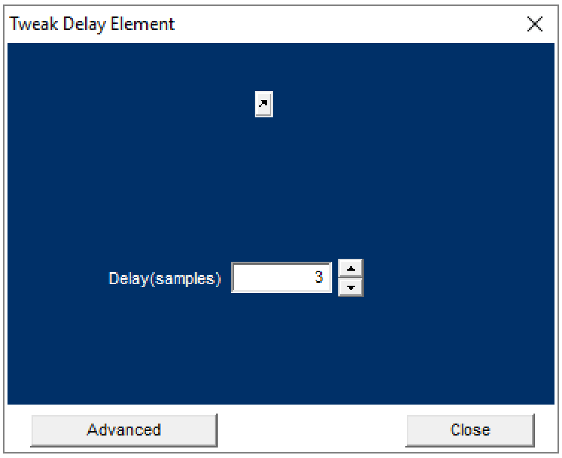


Fig.23 Digital Delay. 3 sample delay @ 48 kHz

Acoustic delays are usually caused by different arrival times for sounds coming from drivers. The Unit Delay Fig. 24, will simulate the acoustical delay from each driver. It is quite easy to simulate the effect of moving a driver closer or further away in mm or uS (micro seconds). In reality this is done by raising or lowering for example the tweeter on the baffle.

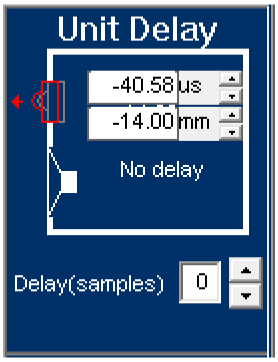


Fig.24 Acoustic Unit Delay simulation in mm (Distance) or μs (Time)

7 SUMMARY

It has been demonstrated that controlling the off-axis responses is an efficient method for obtaining a controlled sound power.

Modern software is very helpful in the design process by being able to simulate DSP Biquads and Digital filters applied to both on- and off-axis measurements of the loudspeaker.

The benefit of using multiple simulated cross-over optimization is a significant reduction in the time needed to investigate design tradeoffs such as different cross-over types, frequencies and/or application of non-standard cross-overs.

The shown method is ideal for obtaining a crossover giving a balanced response and sound power. In addition safe power output capability can be obtained from the design process.

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The Author

Peter Larsen, B.Sc. EE. started his career with SEAS in 1974 and was Chief Engineer for Vifa-Speak 1979-1987. Thereafter Dynaudio 1987-1990 and JBL in USA until 1993

From 1993 Peter Larsen has worked as an independent consultant for leading loudspeaker factories all over the world: Audax in France, KEF Audio in UK, Goldmax in China, Vifa-Speak in Denmark, Peerless Fabrikkerne of India, NXT in UK, Microlab in China, Apple in USA and many others.

During the period as an independent consultant Peter Larsen has specialized in in-depth analysis of loudspeakers and manufacturing techniques, research concerning new components and materials, advanced Acoustic Finite Element modelling, new measuring methods, novel speaker design concepts and development of several customized products (private label).

During the same period Peter Larsen has developed and marketed globally special software for speaker development and design.