Manual

FINE DSP™

Digital X-Over and DSP simulation

Contents

1. How to install the LOUDSOFT programs from a link

**The computer must be connected to internet during installation.
You can´t move the program after installation, so make sure to install on the right computer.**

- When you click on the license link, you must download the program. Click on "Download" at the top right corner of your screen. Make sure to save it, you might need it later. We change the links from time to time for security.
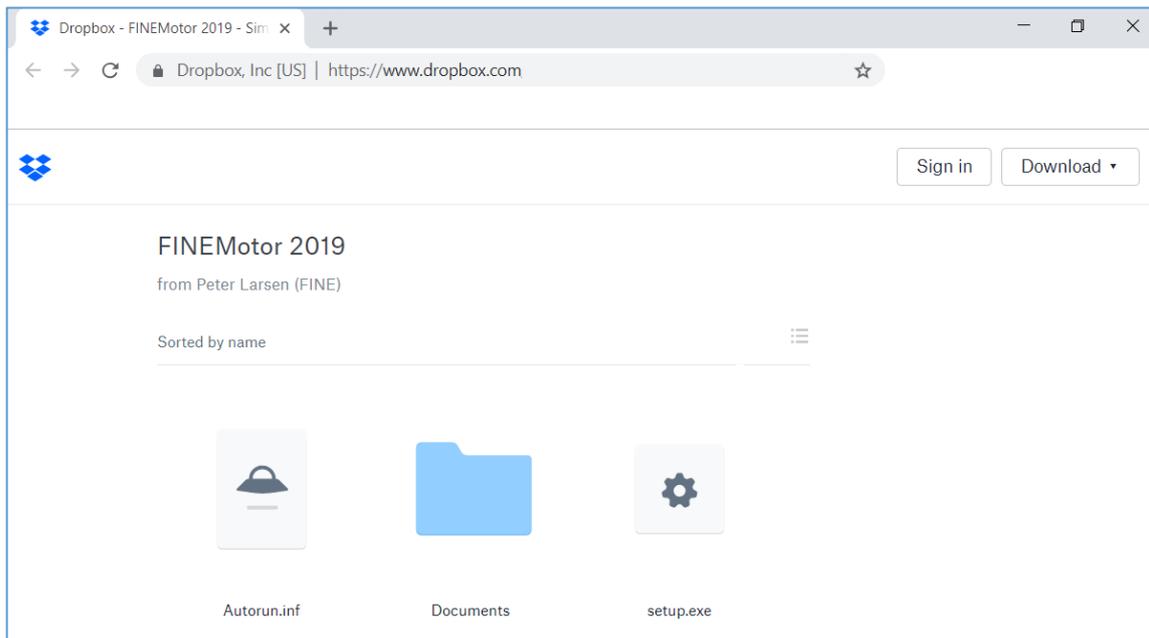
- Now find the downloaded file at the left bottom of your screen. Click on it and you will see the picture below. Click on "Setup" and follow the instructions to install the program.
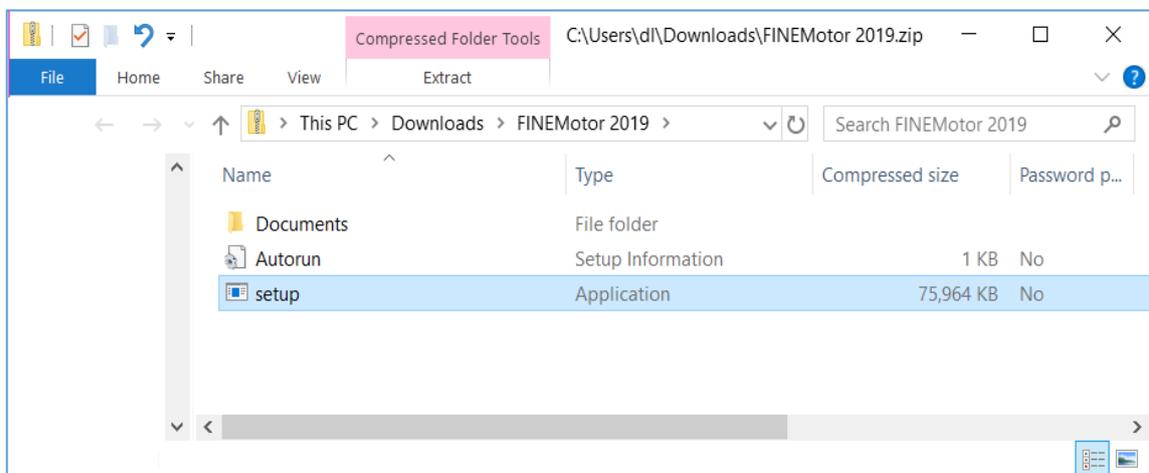
- Once the installation has finished you have to activate (and not later, because it will cause problems). The software will be usable once the activation has been completed.

- Click OK on the first dialog
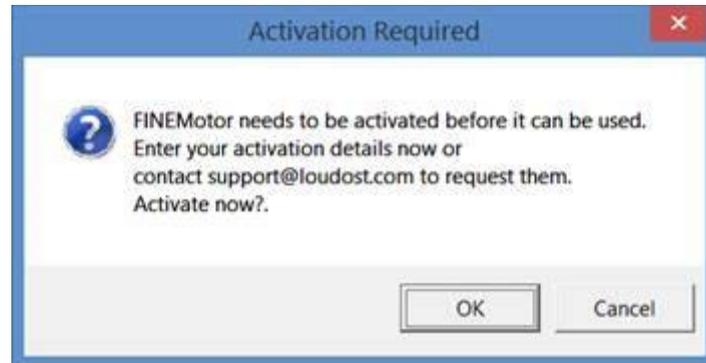
- Click YES on the second dialog
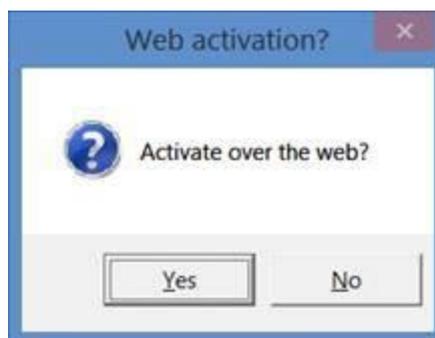
- Fill in your license ID and password in next dialog. Your License ID is XXXXXXXX, and your password is PPPPPPPP

- Click OK. It is all done!

2. Introduction

FINE DSP is the successful software for Digital Speaker Design and Optimization, which is built on FINE X-over while providing interesting new features.

Intuitive Parametric EQ, Bandpass, Shelving filters all with adjustable Q and slope plus high order crossovers and delays are all controlled by mouse wheel / clicks.

Up to 7 measured SPL responses are imported and simulated by DSP simultaneously so that both on-axis and up to 6 off-axis responses are displayed with acoustical phase in real time. Traditional DSP optimizing is only about the on-axis response. With FINE DSP it is now possible to use and incorporate off-axis responses in the optimization of digital loudspeaker systems. The user can drag the x-over frequency up/down aiming for the best compromise between on/off-axis dispersion and power rating.

Both the real power for all drivers and actual Power / Excursion limits are calculated instantly and displayed as the new SOA (Safe Operating Area) for the loudspeaker drivers. Digital and analog levels are calculated based on the SOA, which helps in preventing overload both Analog and Digital levels. This can be based on Max Voltage limits imported from FINEBox .

Moreover, FINE DSP can simulate hybrid crossovers containing both a passive x-over circuit as well as Digital EQ by DSP. This is especially useful when using just one amplifier, thereby keeping costs down. In the hybrid crossover you can design and optimize the passive components, and then use Digital EQ by DSP to perfect the response.

Number of Biquads as well as Digital Biquad Coefficients b0, b1, b2 and a1 and a2 are calculated for import into other software from the IC manufacturer.
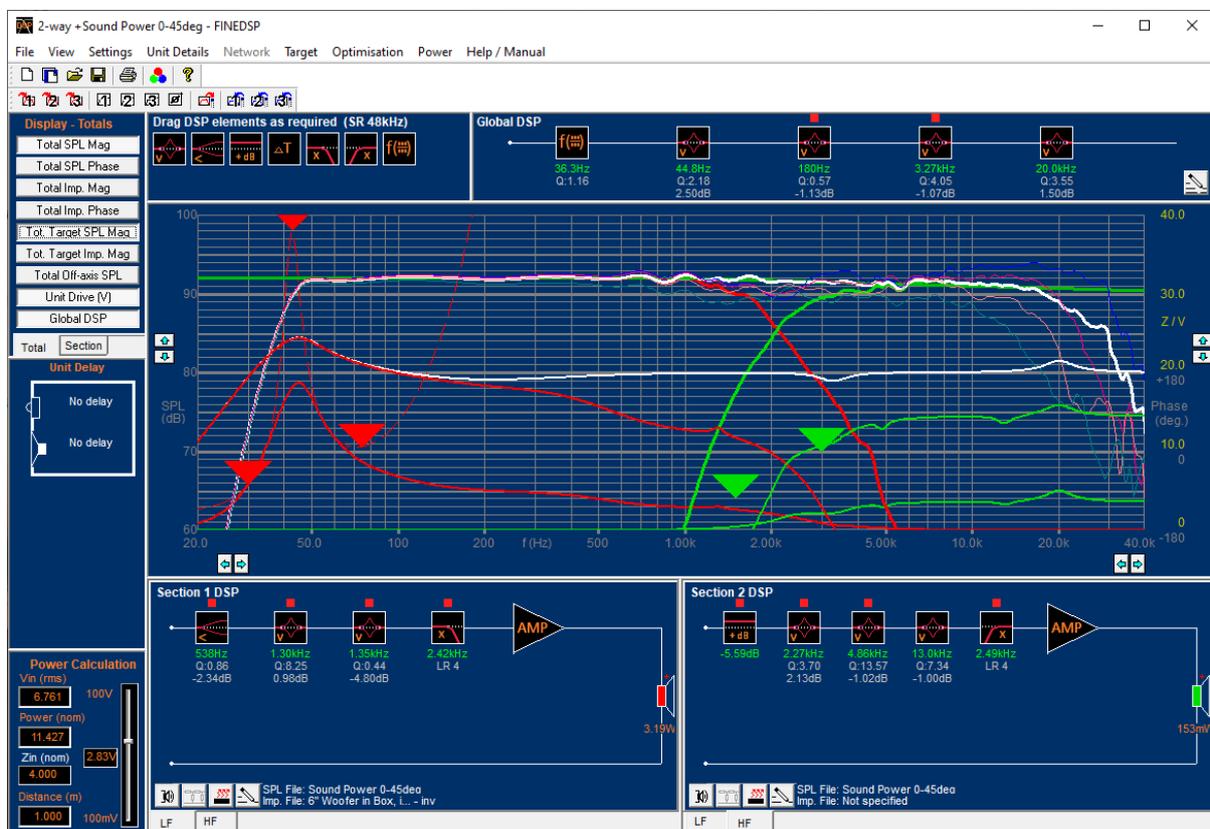
Figure 1 - FINE DSP Main Display

3. High Quality Speaker Design

The traditional way of designing crossovers is about getting a ruler-flat SPL response on axis. There is however no guarantee that this will sound good.

Floyd Toole from Harman has made a very large study of High-Quality loudspeakers including both blind listening tests and comprehensive measurements.

The curves in Figure 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, where the worst had fidelity ratings 6.0 – 6.5.

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.

FINE DSP is specially designed and programmed so this result can be achieved easily based on real loudspeaker DSP simulations.

TOP-TO-BOTTOM: AVG. ON AXIS, AVG. 30-45 DEG., AVG. 60-75 DEG.
SIX LOUDSPEAKERS WITH FIDELITY RATINGS OF 7.5-7.9.

(d)

Fig. 7. Amplitude response measurements of loudspeakers with fidelity ratings (d) 7.5–7.9.
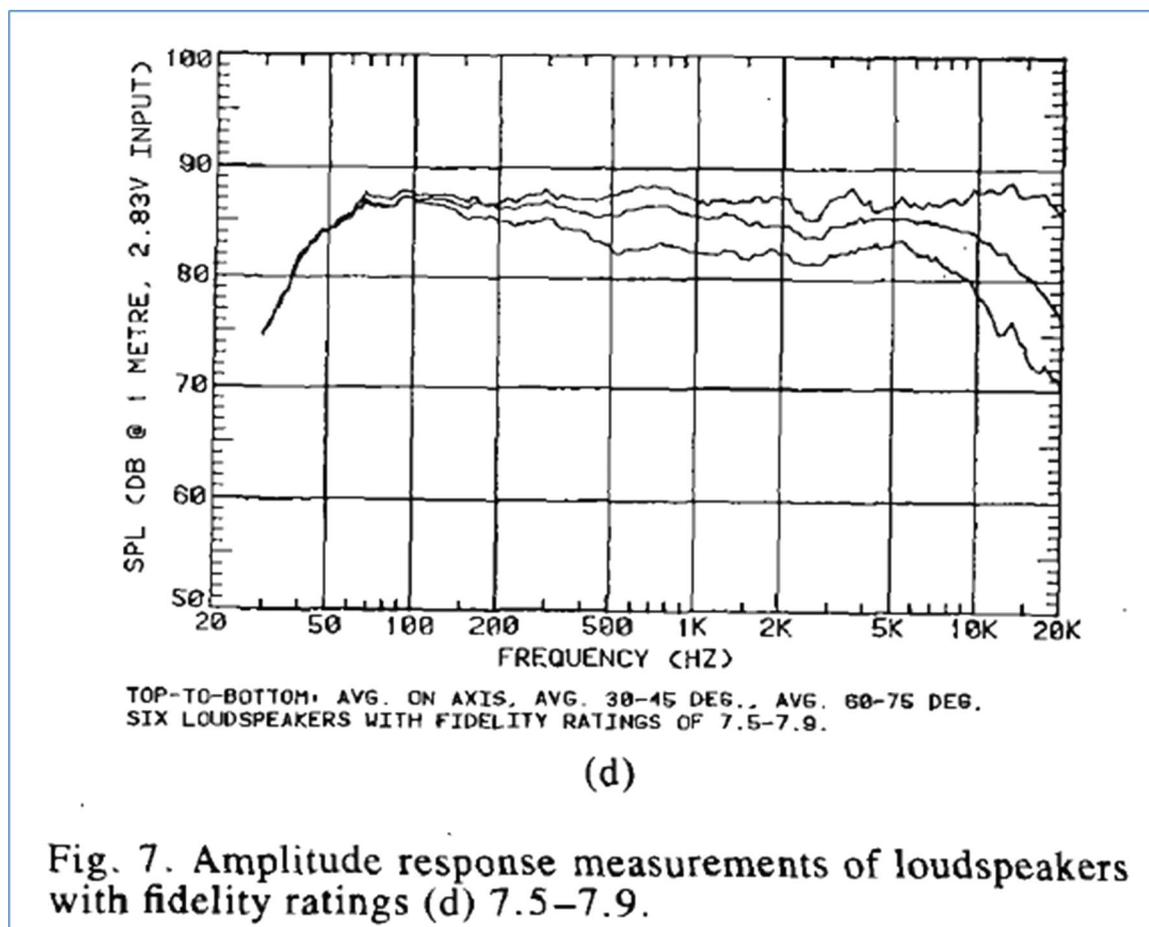
Figure 2 - Amplitude response measurements of loudspeakers with highest fidelity ratings (from Floyd Toole)

4. Simple 2-way DSP X-over example

Let us start with a simple 2-way loudspeaker cross-over demonstrating how easy it is designing with FINE DSP. Thereafter we will show a more advanced example where we use many off-axis responses at the same time.

Start a new design by NEW. The wizard starts with the window shown in Figure 3.
Before using the wizard, you should have measured responses of all drivers with phase one by one. Place the microphone ideally in the listening position if you have a quite large room. If you have a normal room with only 1-2 meters to the side walls you can measure the response without reflections using FINE R+D, which is FFT based with a Time window for excluding the reflections. This way you will automatically get the phase responses included with the measurements.

Place the microphone on the listening axis, which is usually in line with the tweeter as shown in Figure 3. After measuring the on-axis response rotate the speaker around the vertical speaker axis (through the tweeter) by keeping the same distance all the time, and without moving the microphone. If only measuring one off-axis curve then the 30deg off-axis response is a good choice.
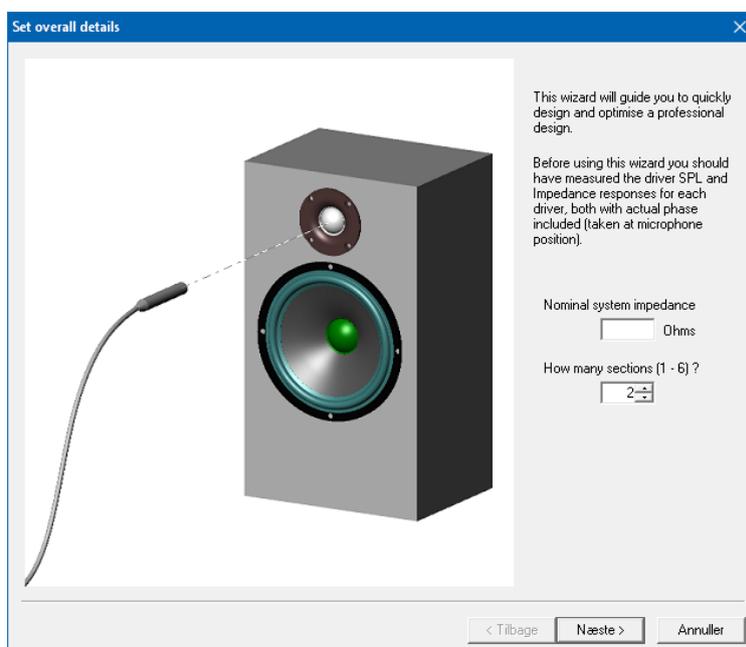
Figure 3 – New : Wizard

Now press next in the wizard, see Figure 4 and 5. Here you can import the measurements which must have phase included (= Transfer functions).

You can drag and drop both on- and off-axis responses directly from FINE R+D by keeping both programs open.

The best is importing the Sound Power curve, which is done by averaging several off-axis responses directly in FINE R+D.

You can also import from many formats, txt (VACS/Klippel), Tim, FRQ, (MLSSA), Lab and FSIM (LOUDSOFT). You can preview the responses as shown in 5.

Note: Be sure to set the DSP Sample Rate, see Figure 39.
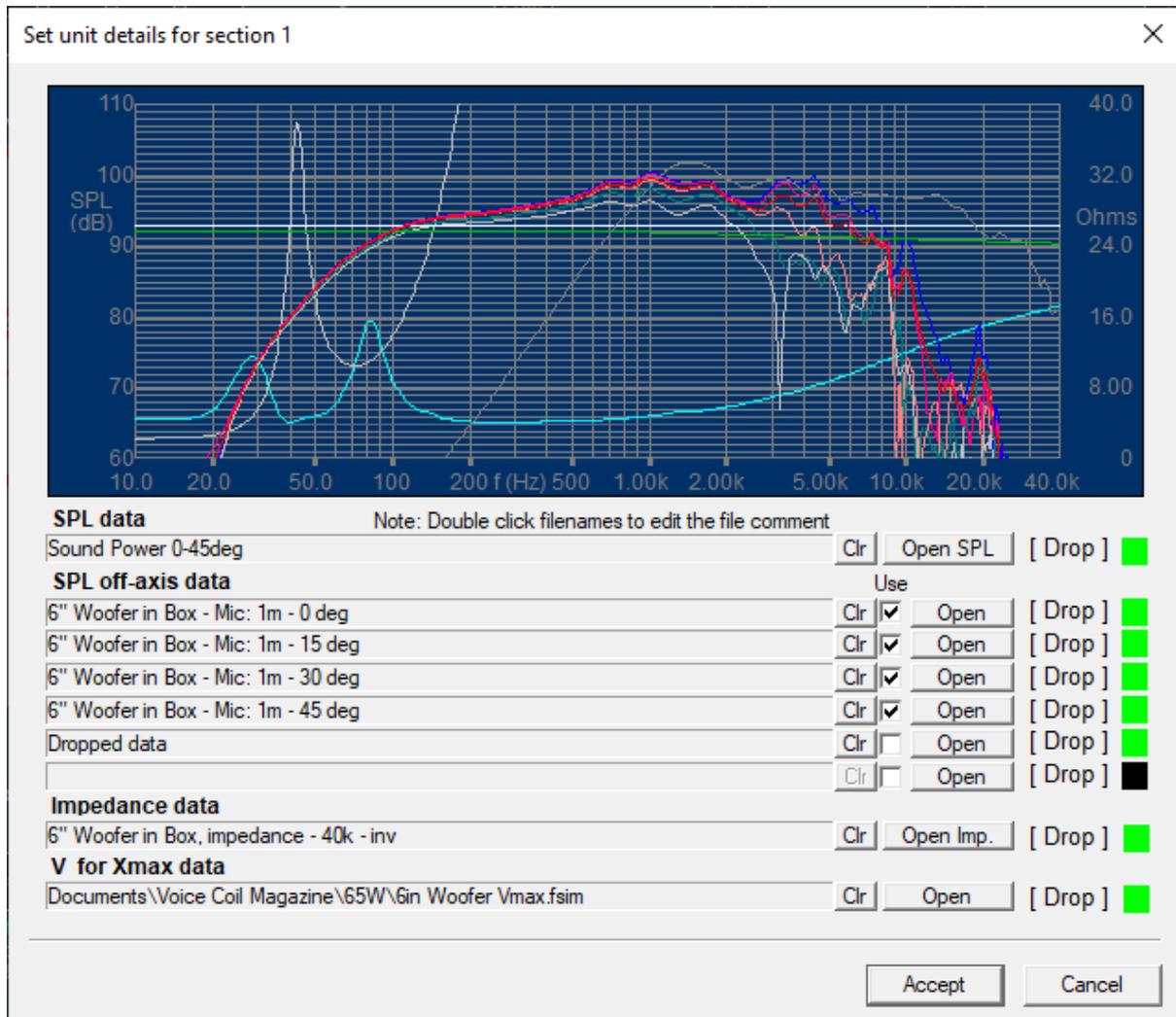
Figure 4 - Drag/drop responses from FINE R+D
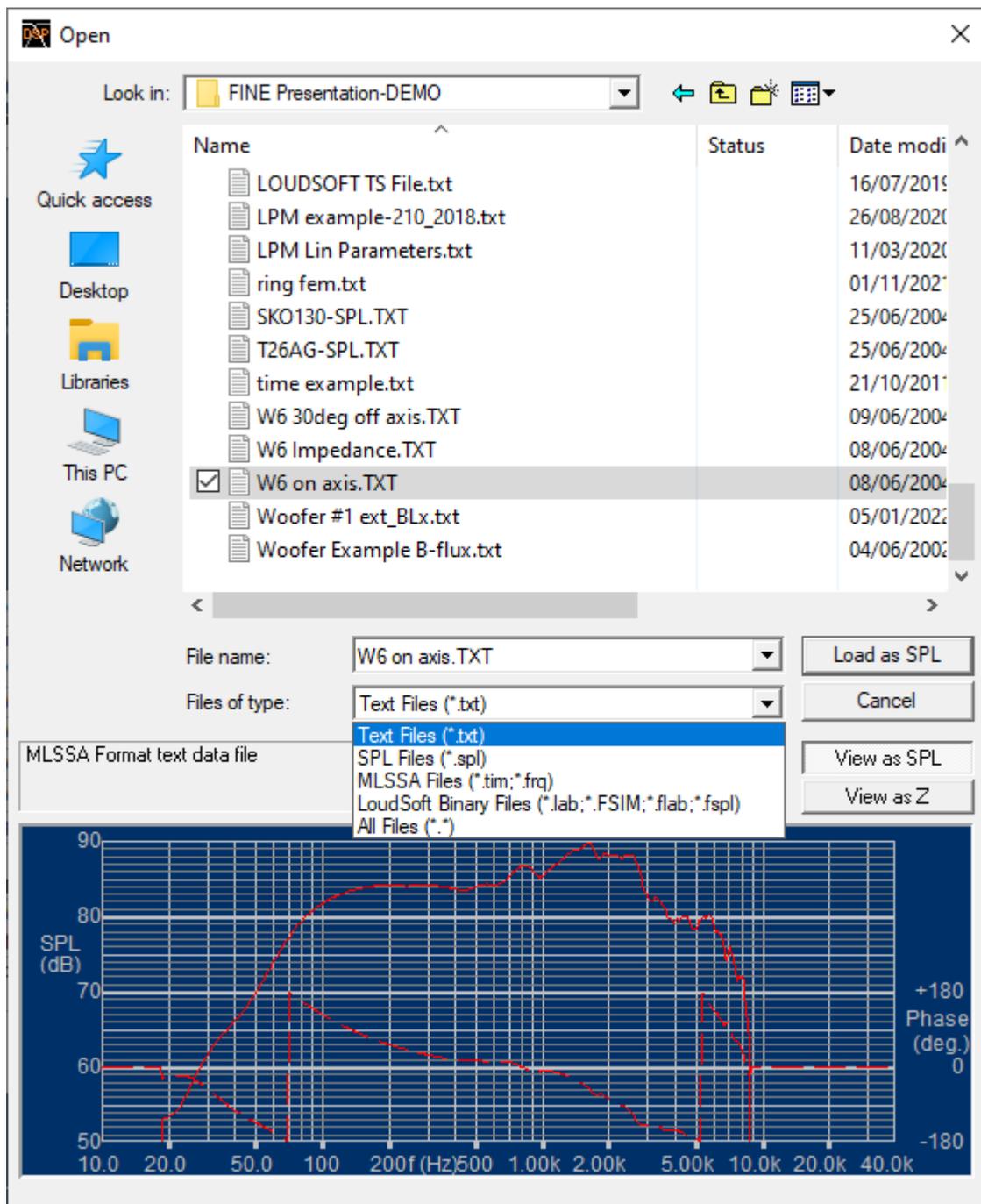
Figure 5 - Import measured responses with preview

Figure 8 shows both an on-axis measurement as well as one off-axis measurement, but we will only use the on-axis in this first example for clarity. Select the target around the ~flat level of the woofer as shown (83.7dB).

Next import the measured tweeter response in the same way.

4.1 SPL scaled with Power Input

When you select a frequency response file for import you will be asked if it should be scaled according to the current power settings in FINE DSP. The scaling is automatic when importing from the latest FINEBox and FINE R+D programs.
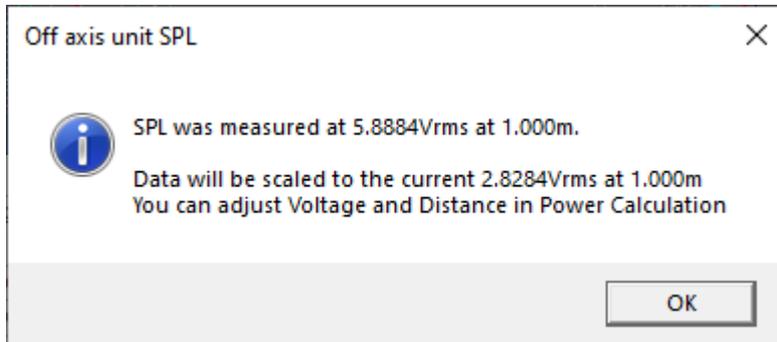
Figure 6 - Imported FINE- responses are automatically scaled to correct SPL

If the measurements come from other systems you will get this message Fig.7, where you can set the voltage and distance used for that specific measurement.
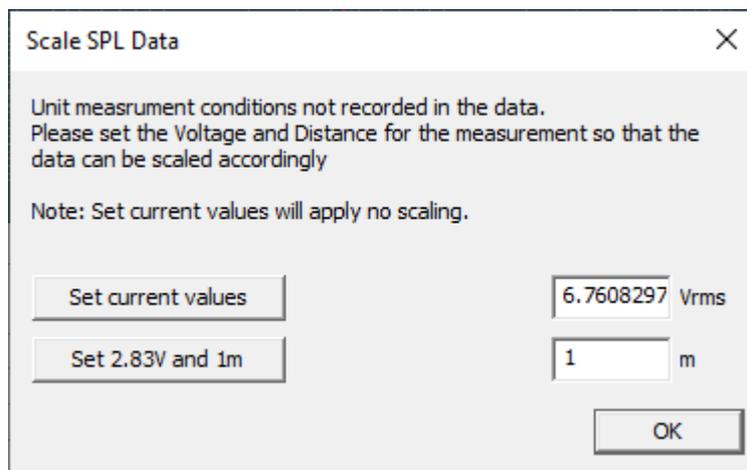
Figure 7 - Scaling of unspecified responses
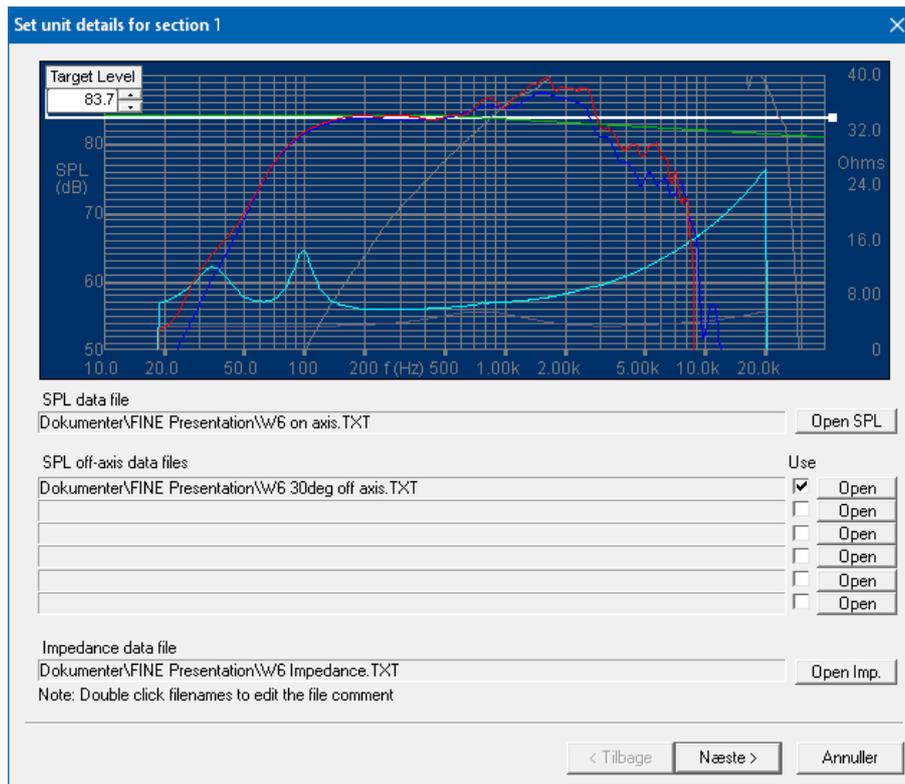
Figure 8 - Imported Woofer responses

FINE DSP now displays the two imported responses in the main window, see Figure 8. First we must choose a X-over for each driver by dragging the X icon (LP=Low Pass) down into the woofer branch.
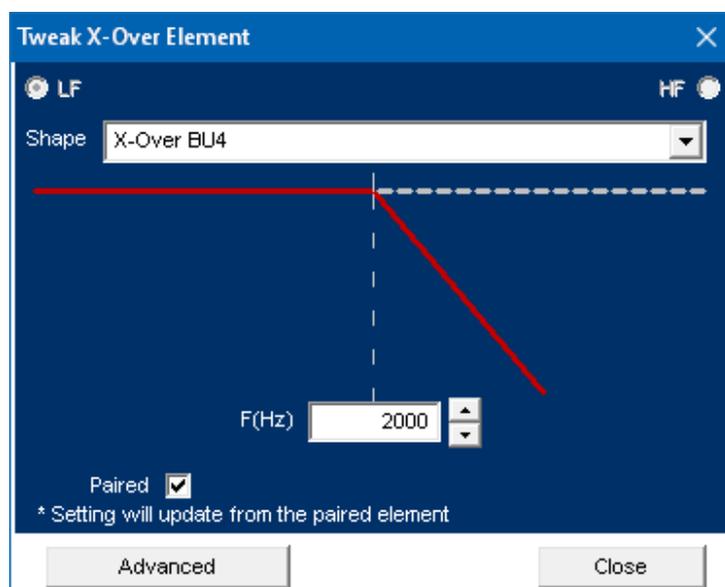
Figure 9 - BU4 (4th order Butterworth) selected @ 2000Hz

Figure 9 shows the LP filter, here selected as X-over BU4. This means 4th order Butterworth, which is a steep slope, ensuring the higher frequencies are well attenuated. Higher slopes are possible, but not generally recommended, because they use more than the 2 Biquads, which are needed for BU4, and they start becoming critical due to the sudden cut-off.

Then we must do similar for the HP X-over BU4 selecting the same frequency (2000Hz). Having done that you may click on the [v] Paired, which means that the two LP and HP filters now will change together in sync.

Try moving the x-over frequency by placing the cursor in the field F (Hz) [2000] (or use the up/down arrows) while rolling the mouse wheel. It is quite amazing watching how well the two driver responses will match together. It is a good idea to drag the attenuator level icon down in the tweeter branch and setting the gain to match the woofer (Fig11. -7dB). This way it is much easier to find a good X-over frequency.

There are two more considerations for choosing a good X-over frequency. You would want to cut off large variations at the top end of the woofer but be careful selecting a too low X-over frequency for the tweeter, because that may overload it, possibly causing a burnt tweeter.

Fortunately, FINE DSP has a very helpful feature, namely Power Calculation. We only need to import the max V curve from FINEBox or measure how many volts each driver can accept at some critical frequencies (see details later in section 8), and we can turn up the input Power in the lower left corner, see Figure 10.
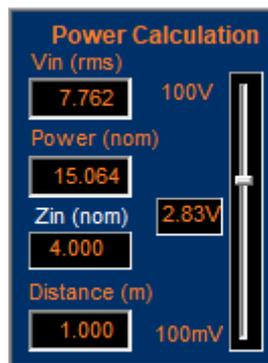
Figure 10 - Power Calculation (Turn up volume with slider)

Selecting [Unit Drive] in [Section] the program displays the calculated Voltage at all frequencies as a dashed red line for the woofer and a dashed green line for the tweeter. The maximum Voltage levels are now shown with red and green arrows.

Figure 11 - FINE DSP main display for the simple 2-way X-over, with red & green power arrows

Figure 11 shows the main display for the final simple 2-way X-over. The woofer response before X-over is shown as the thin brown curve and has the typical rise of ~6dB (due to the front baffle, being only effective at high frequencies). The raw tweeter response is shown in light green (This tweeter was measured in a larger baffle and is very smooth).

We have dragged the shelf into the woofer section to compensate for the rise, and the compensation is -6.4dB at 514 Hz in this case, see Figure 12.
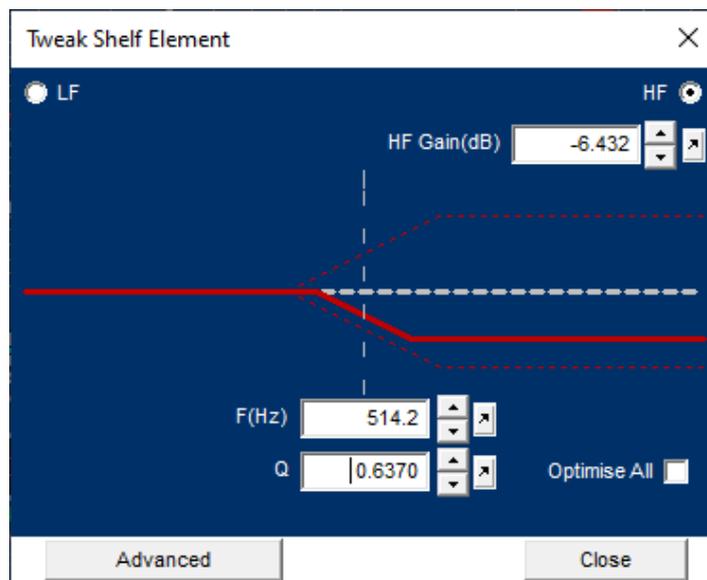
Figure 12 - Shelf at 514 Hz for compensating the rising woofer response

To equalize the tweeter response, we have just dragged the Tweak Peak/Dip Element into the tweeter section at 4166 Hz, and set the Q low (.7721), which nicely makes the tweeter flat, see Figure 11. Finally, we add a 4th order Butterworth (BU4) HO filter @1.81kHz and close to 3 dB attenuation.
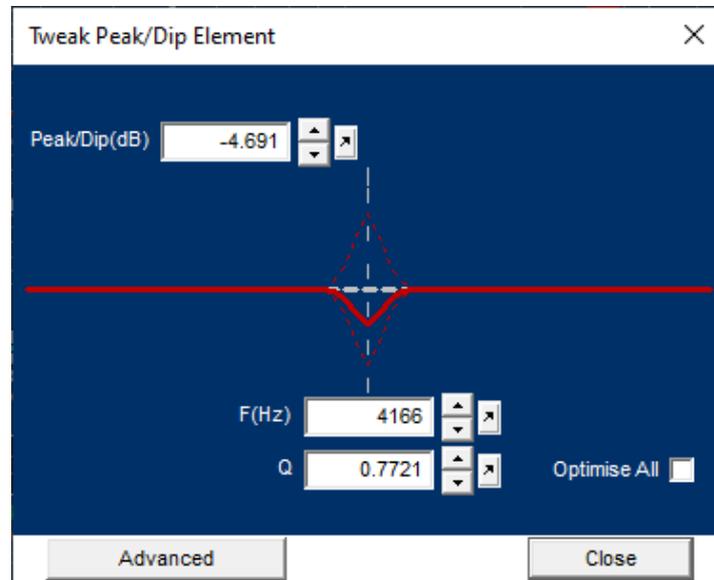
Figure 13 - Tweak Peak/Dip Element in tweeter section at 4166 Hz, with low Q (.7721)

The Biquad settings are shown below each DSP block, and the Coefficients can be viewed by pressing [Advanced] in each DSP element, see Fig. 30.

5. Advanced 2-way / Sound Power + Off-axis responses

The advanced example uses the same two drivers however first with sound power (calculated in FINE R+D) as well as 0-15-30-45 on- off-axis responses measured in the actual cabinet. All response curves are drag/dropped from FINE R+D.

The woofer responses are shown in Figure 14.
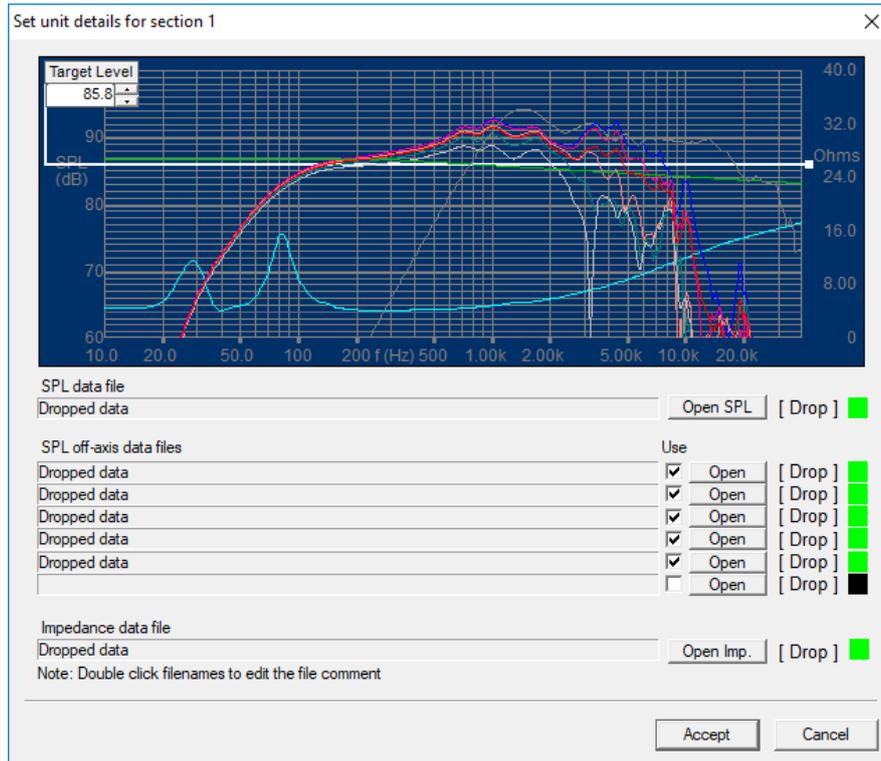
Figure 14 - Woofer responses measured at 0-15-30-45 degree off-axis
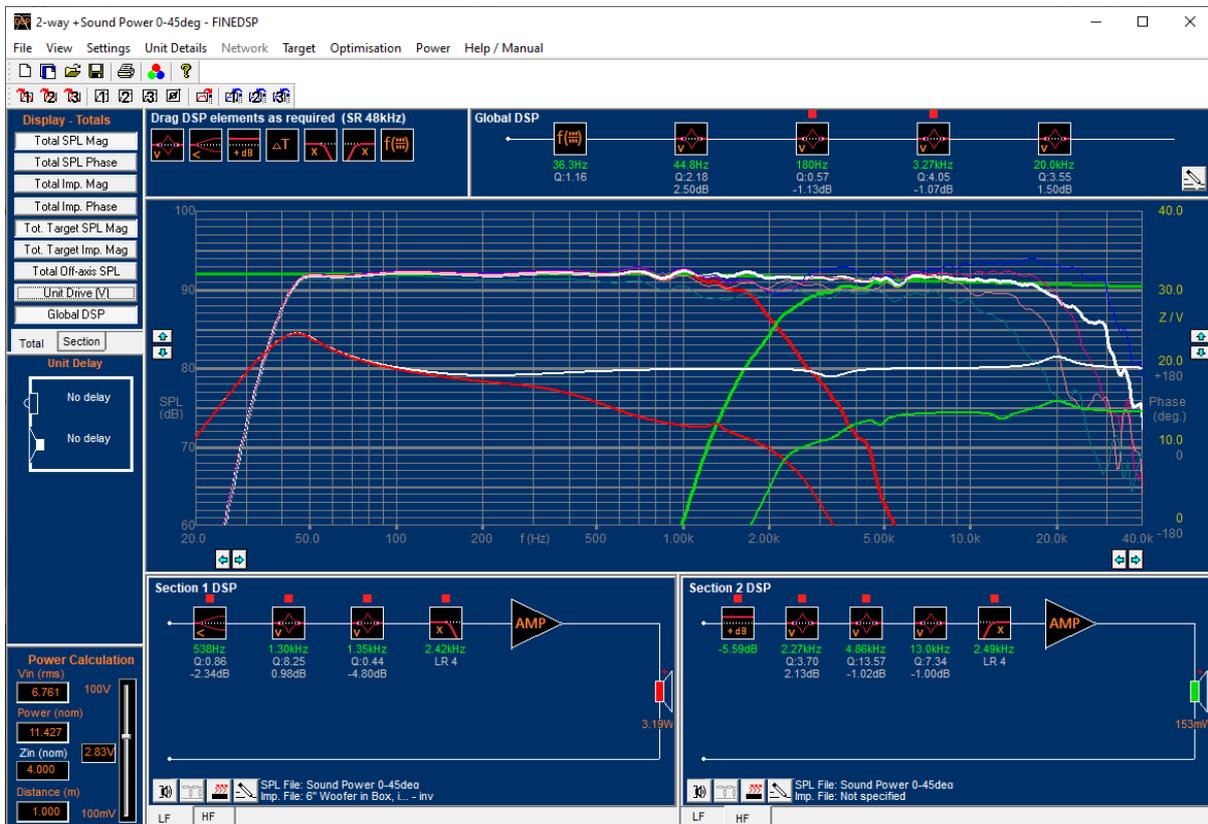
Figure 15 - Advanced 2-way with 0-15-30-45 deg on-/off-axis responses. The white Sound Power curve was optimized. The global and section DSP responses are shown below

Note that the (acoustic) Sound Power response was loaded first because we want to optimize that to control the dispersion, which is also recommended by Floyd Toole, see Figure 2.

This time we are using the full arsenal of DSP elements, in each section and globally Fig. 15.

Both the global (white) and section DSP correction responses (Woofer/Red-Tweeter/Green) are shown below the SPL

Selecting the "Total" button and "Total Target SPL Mag" now 4 responses are displayed as total on- and off-axis responses. The "Tot. Target SPL Mag" button displays the green dashed curve, which is the Sound Power target, gently sloping down. This is specified in Figure 16.
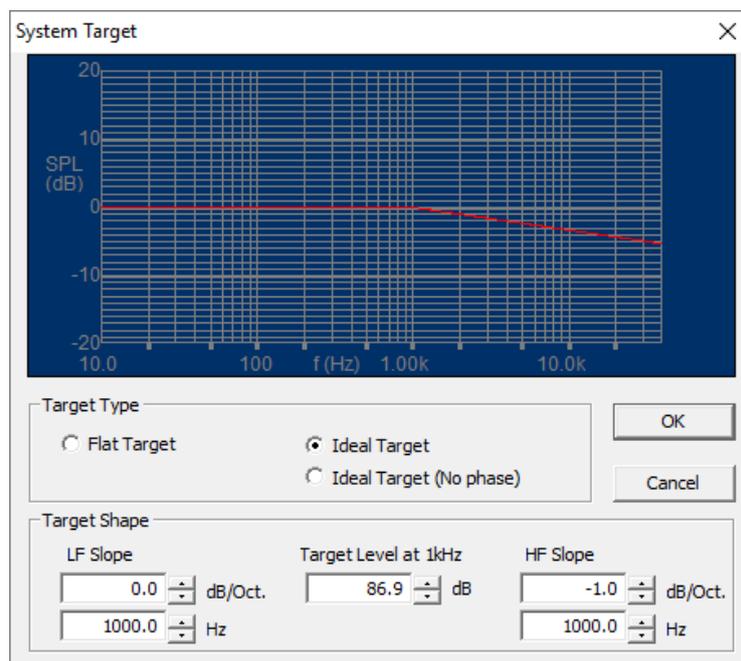
Figure 16 - System Target, with -1dB/Oct. from 1000Hz

Like in the first simple example we have also used DSP elements like LR4 x-over, a shelf in the woofer section and tweeter level (dB). Plus 3 Tweak Peak/Dip EQs were inserted in the tweeter section.

Another Tweak Peak/Dip EQ was inserted as global at 3270 Hz, see Figure 15. A vertical blue dashed line indicates the frequency. Looking at the 4 curves we notice that 3 of the responses are close together, whereas the on-axis (Blue) shows a bump. This was intentional as the off-axis responses are more important than the on-axis.

By moving the X-over frequency up/down this can be further improved.

Finally, the total system may be optimized automatically. Select (System) Optimization from the main menu, and press Optimize. Now a range is shown with green bars. Then all the DSP elements with red arrows can be optimized, see Figure 18.
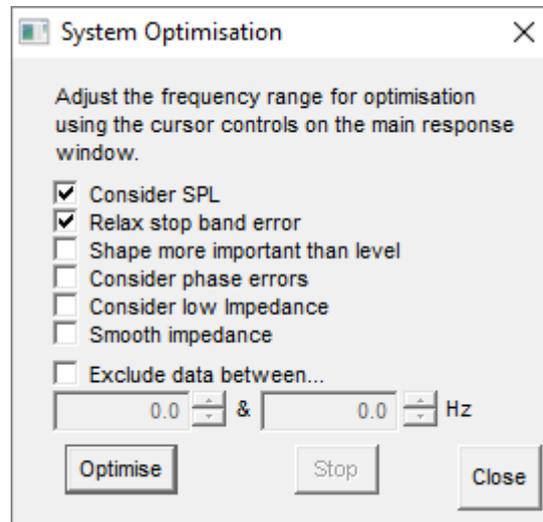
Figure 17 - System Optimization

In this case the Sound Power (White) response was optimized for the dashed (green) target, see Figure 15.

6. DSP Display

The global and section DSP settings produce response corrections, which are shown with the frequency, Q and level (dB) below each DSP block Fig. A red square indicates if a block will be included in optimizations.

Figure 18 - DSP blocks with Freq, Q and level (dB)

You can view both the total response of the global DSP corrections Fig. 20, as well as the DSP section responses by pressing the buttons at left Fig. 19. Press Section to go to sections, for viewing section DSP responses

Figure 19 – View Global DSP (Section DSP with Section button)
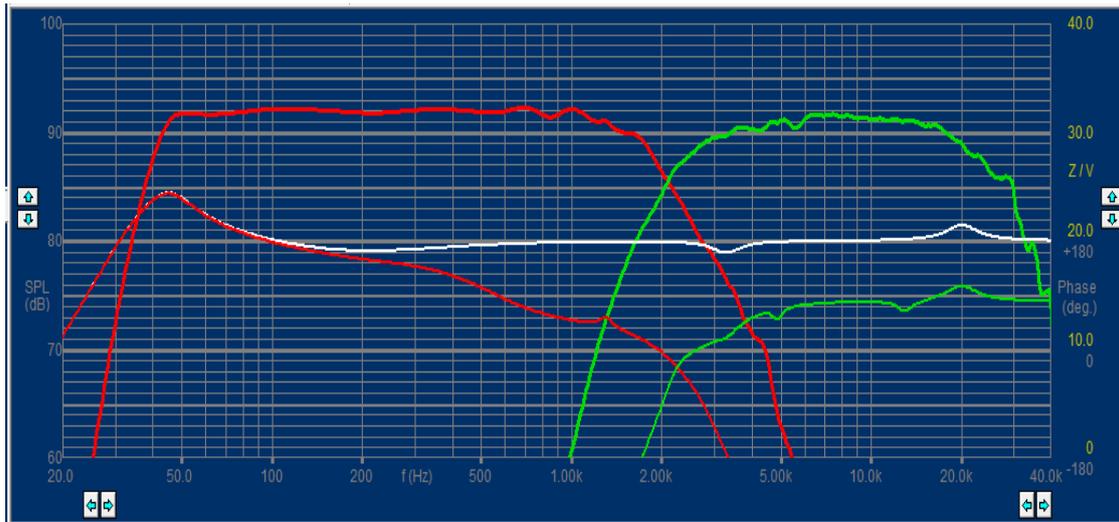
Figure 20 – Center: DSP global response (white) and sections (Red +green)

7. Hybrid- Passive + Digital DSP example
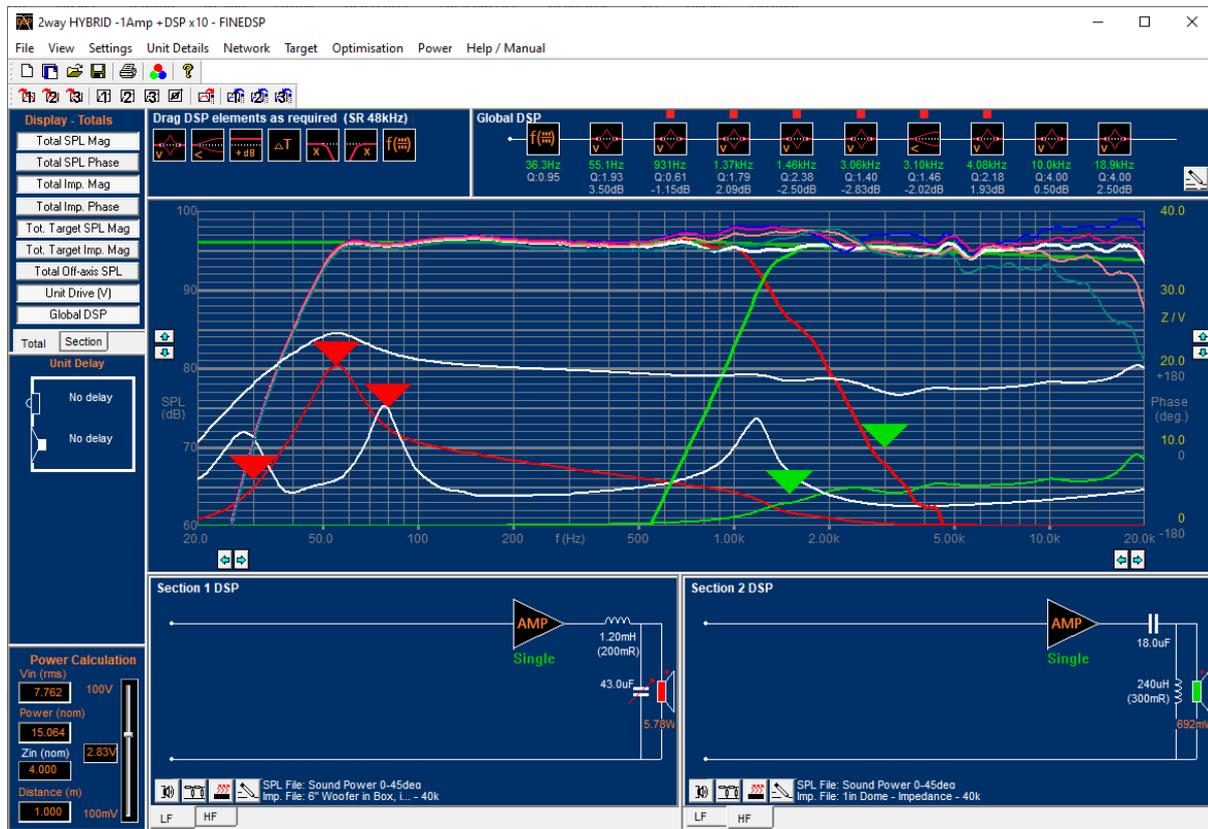
Figure 21 – Hybrid: Passive + Digital Speaker with DSP. Only one Single amplifier is used + passive components

In the last example we will use both passive and DSP components. By using only one single amplifier we can save costs. However, it is then necessary to insert a passive X-over circuit to separate the two drivers, and use global DSP EQ to equalize the total response, see Figure 21.

This time we also imported the measured impedance response (lower white). Using the Wizard, we have chosen a 12 dB/oct. passive filter with a X-over frequency 1500 Hz, giving good separation between woofer and tweeter. Note that we can edit all the passive components as we wish. For example, a large inductor in series with the woofer, could be changed to a smaller, less expensive inductor.

8. Power (V, A) and Xmax in Drivers & passive components

The actual power, shown in Figure 21, in the woofer (5.78W) and tweeter (692mW) is now calculated accurately because the measured impedance was used. Both the calculated acoustical phase, electrical phase and impedance was calculated and displayed.

When considering the maximum power input into the system we need to know how much each driver VC moves. This is called Voice Coil Displacement or Excursion, and the normal maximum is called Xmax.

Unfortunately, the voltage to obtain Xmax for each driver depends very much on the frequency, see Figure 22.
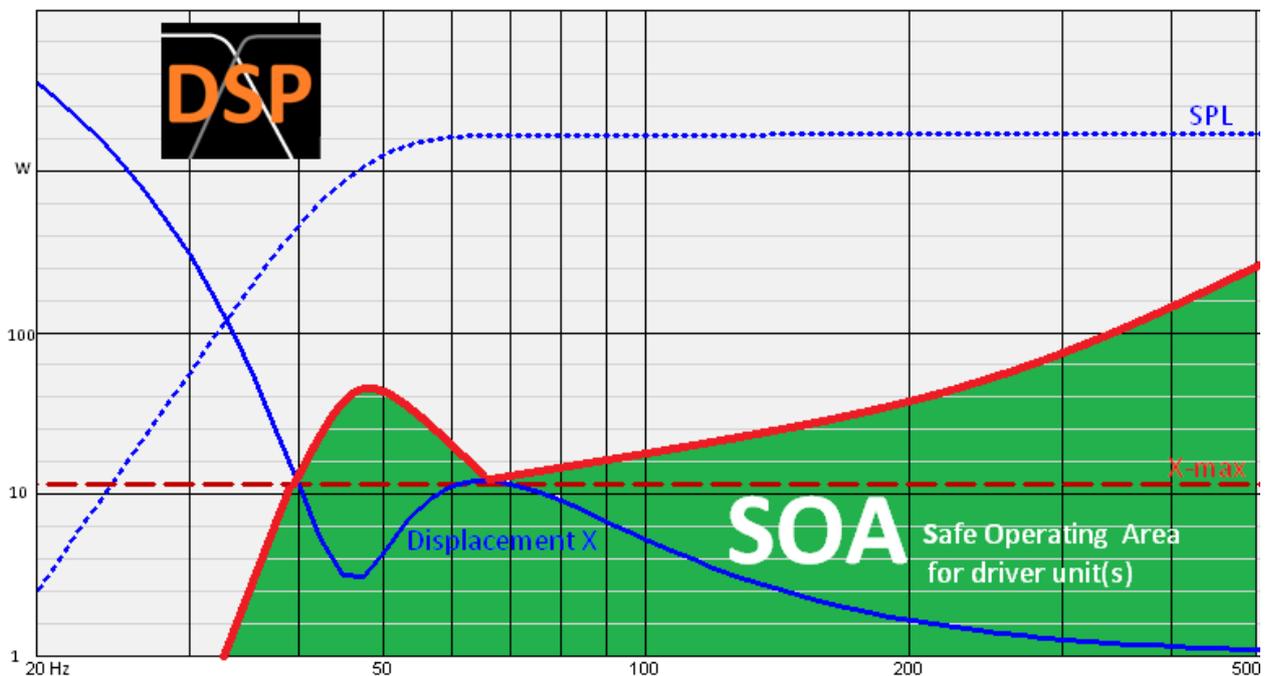
Figure 22- SOA (Safe Operating Area) for driver control with DSP

Figure 22 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, 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 is moving less at this frequency compared to 65 Hz.

Therefore, we can optimize the low frequency response using this information. In the 2-way system a High Pass (HP) filter was inserted at 36.28 Hz to attenuate the very low frequencies where the total response has dropped anyway, see Figure 23. The HP filter was found in the general DSP function, which was dragged into the global field:
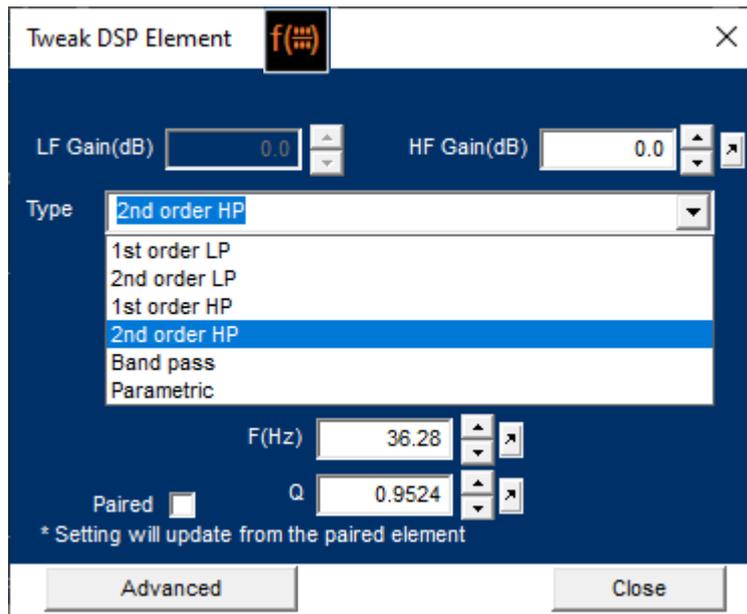
Figure 23 - General DSP Element with 2nd order HP selected

Following a Tweak Peak was placed at 55Hz just where the Displacement is low for this system. We have set + 3.5dB, which will boost/extend the bass. These two features will improve the system by giving more bass, and at the same time protecting the woofer at very low frequency (<36Hz), where the excursion is high, but where the response is dropping anyway.

But how do we know how much Power or Voltage will be used at these frequencies?

8.1 Max Voltage Limit (Import)

Figure 24 gives the answer. After defining your box tuning and TS parameters in FINEBox, you can simply export the max. Voltage Limit from FINEBox.

The Max Voltage was imported into FINE DSP and is displayed as the dashed red line Fig. 24. Then by pressing Unit Drive (V) the actual driver voltage is shown as the red line, which is peaking at exactly 55Hz, but is kept lower than the max V (- - - - -).
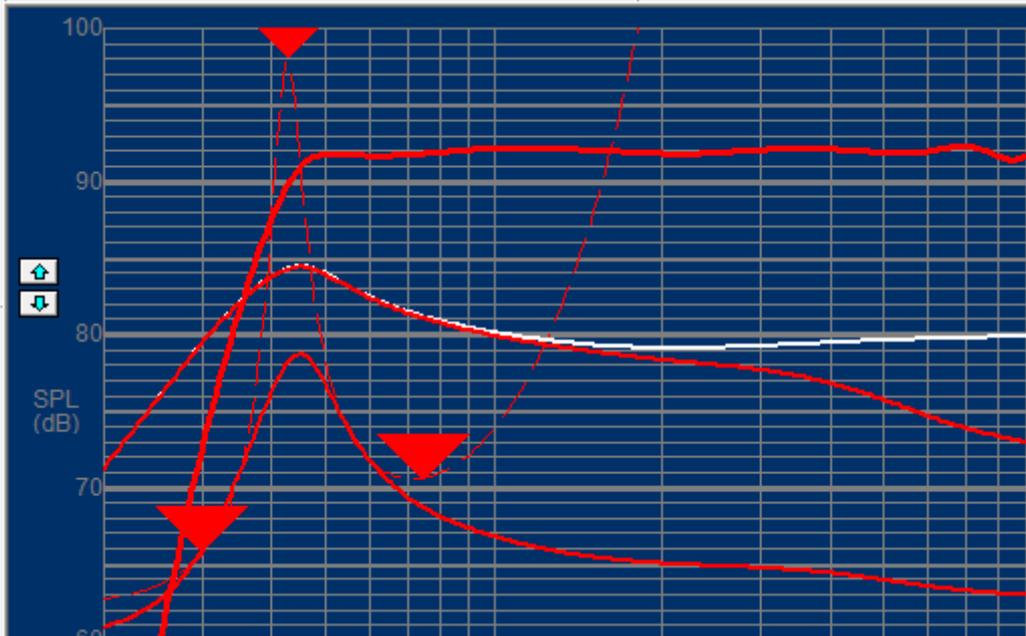
Figure 24 - Setting Bass Boost using imported Vmax limit (- - - -)

It is also possible to set the (red and green) triangles for the same purpose. 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 Figure 25, where we have also measured the voltage for Xmax at the tuning frequency 55Hz. At this frequency the woofer can accept 20.4V, which is very high. But only 15V at 80Hz, which is now the frequency with maximum displacement, see Figure 25.
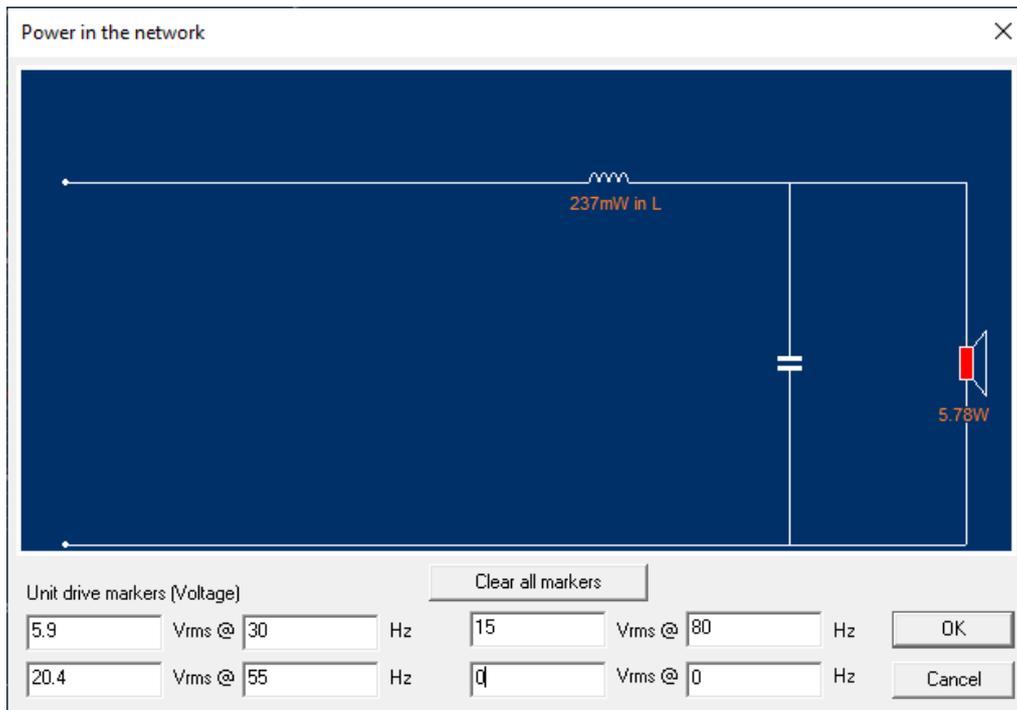
Figure 25 - Input Voltages for Xmax at various frequencies

Select [Unit Drive] under [Section] and the main display with show red arrows indicating the maximum voltage points we have just specified.

If for example the amplifier is 20W or larger, we must turn down the +2dB boost accordingly as well as the high Q (=2.53) of the HP filter @36Hz.

However, the amplifier may not be able to supply enough current. In Settings /General / Calculation we can instead specify the Unit drive current at 4 frequencies. An example is shown in Figure 26.
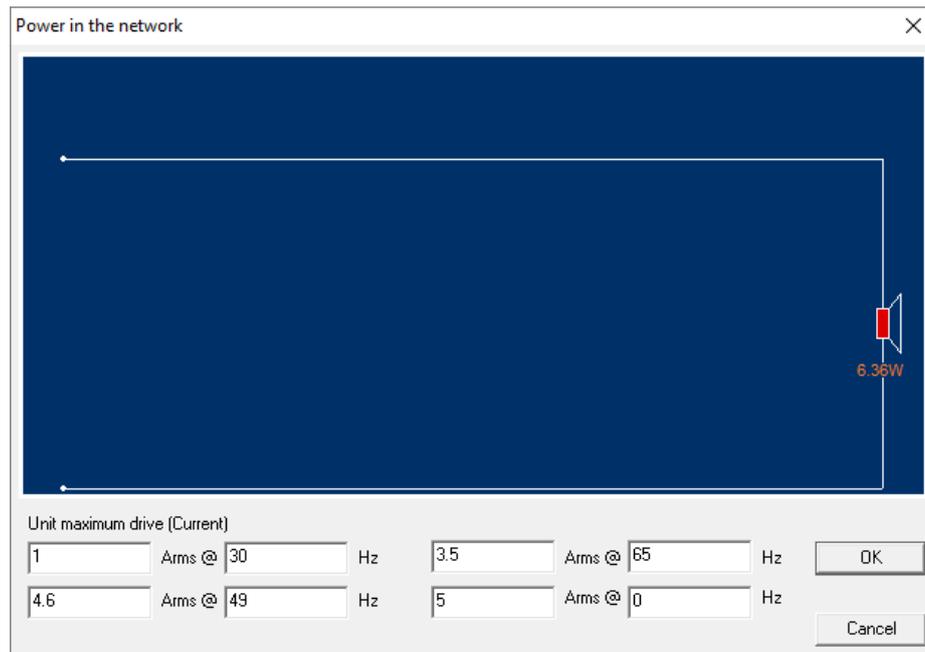
Figure 26 - Max Input Current at various frequencies

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 (see Figure 21).

In this case the tweeter is not reaching maximum yet. But that may change if the x-over frequency is set too low. It is therefore advised to keep the Unit Drive on when choosing the final x-over frequency and slope.

9. Delays

It is possible to introduce delays in two ways, Digital and Acoustical Delays. Figure 27 shows a 3-sample digital delay using the standard 48 kHz sampling frequency.

Note that such delays are common and depends very much on the number of Biquads used. Measuring or specifying this delay is difficult, and we therefore recommend first measuring for example a tweeter response, and then simulating a good X-over with DSP for that. Then re-measure that section response and import it into FINE DSP using the import User1

button. . Then insert a Digital Delay and find the number of sample delays which fit best.
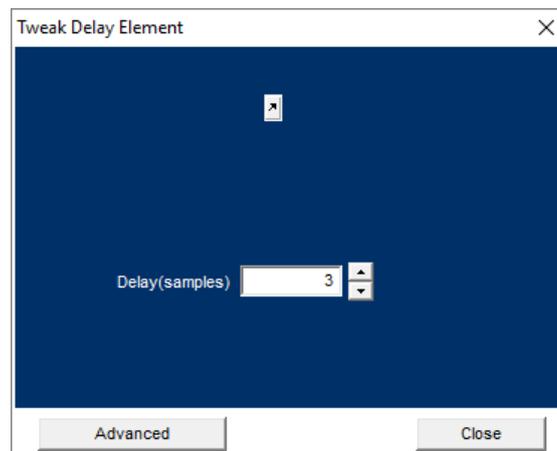
Figure 27- Digital Delay. 3 sample delay @ 48 kHz

Acoustic delays are usually caused by different arrival times for sounds coming from drivers. The Unit Delay, shown in Figure 28, 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 (microseconds). This is done by raising or lowering for example the tweeter on the baffle.
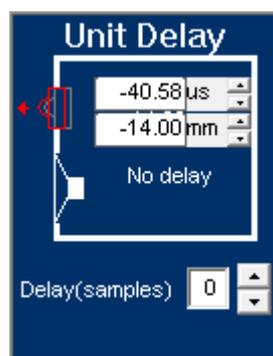
Figure 28 – Acoustic Unit Delay simulation in mm (Distance) or uS (Time)

10. Import / Export of SPL Responses and Biquads Coefficients

a. SPL

Figure 29 - Save / View / Export / Import SPL responses

The user can save and display various SPL responses, see Figure 29. The 3 left buttons (1,2,3) will save the current total SPL curve with a new color. The center 3 buttons will display the saved 1, 2, 3 curves. The [ø] will show the phase of 1, 2, 3.

The button will export the current total SPL response as a File.

The 3 buttons at right 1,2,3 will import responses from files

b. Biquads

You can export the Biquads Coefficients of all the Biquads used as a text file. Select Export DSP Coefficients under File.

The Biquads Coefficients can be viewed by pressing [Advanced] in each DSP element, see Fig. 30.

DSP Element Advanced ✕

Consists of 1 biquad
At 48.0kHz sample rate the coefficients are:

b0 = 0.91851173835038 b1 = -1.56422193102309 b2 = 0.66930936398829
----
a0 = 1.00000000000000 a1 = -1.56422193102309 a2 = 0.58782110233867

Close
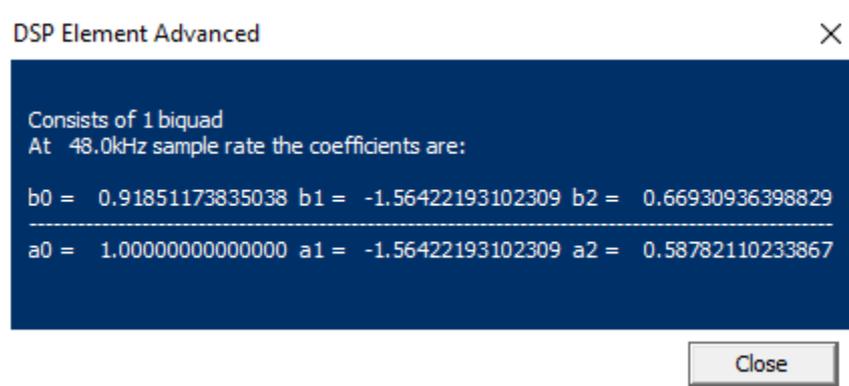
Figure 30 - Biquad Coefficients

11.   DSP Elements
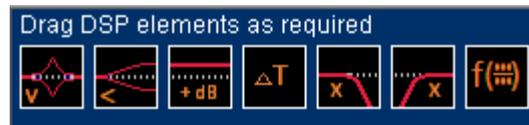
Figure 31 - DSP: Peak/dip, Shelf, dB, Digital Delay, LP, HP and special EQ for dragging into sections or global

a. Peak/Dip Element
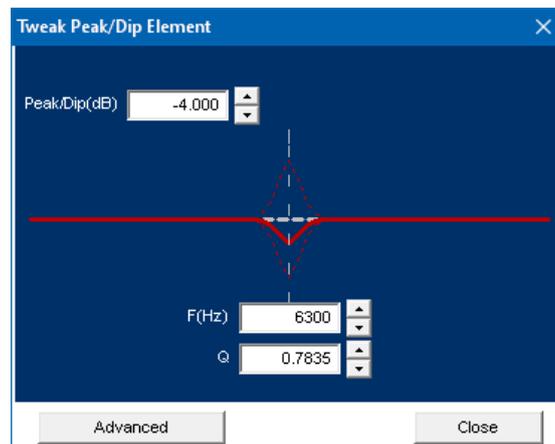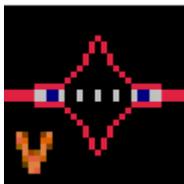
Figure 32 - Tweak Peak/Dip DSP Element

Figure 32 is the Tweak Peak/Dip DSP Element. This is the most used DSP element, which can compensate a peak in the response by inserting a comparable dip. You can also compensate a dip in the response by a peak, however, be careful not to compensate a deep and narrow dip with a sharp peak! Such a narrow dip may be two drivers cancelling each other by being out of phase. If you then insert a narrow peak, that may sound much too sharp!
Rolling the mouse wheel will move the selected frequency up/down, which will be indicated on the main display by a vertical blue dashed line. Set the Q to match the width of the peak. For a narrow peak use Q > 1 and for a broad peak use Q < 1. Finally set the Peak/Dip dB level by rolling the mouse wheel or use the arrows.

b. Shelf Element
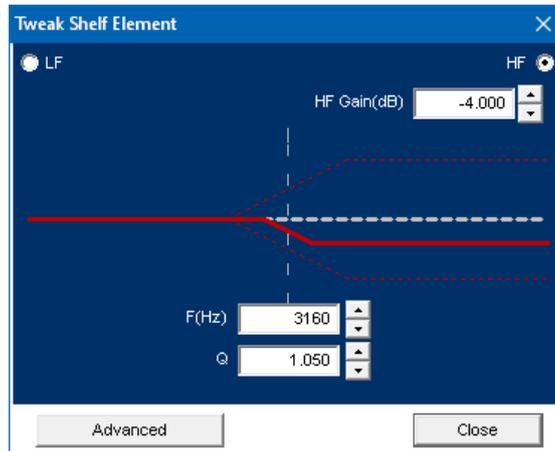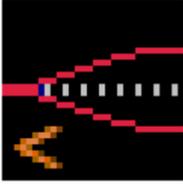
Figure 33 - Shelf Element useful for compensating rising response

The shelf element is extremely useful for compensating the natural (~6 dB) rise in the woofer response. In that case the HF Gain must be set with a negative value, here – 4 dB. The Q indicates how sudden the shelf is inserted.

c. X-over LP/HP
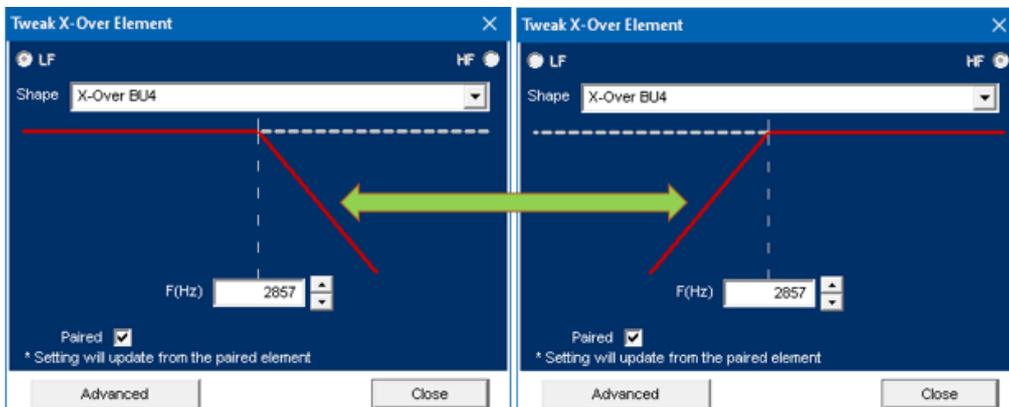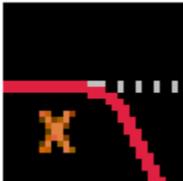
Figure 34 - LP / HP X-over (Cross over) Elements with tracking (Pairing)

Insert a cross over (X-over) LP (Low Pass) for the woofer and HP (High Pass) for the tweeter, or both for midranges. Select the order/slope from the dropdown box, see Figure 35. You can select either Butterworth (BU) or Linkwitz/Riley (LR). The BU is -3 dB at the crossover frequency while LR is -6 dB. The BU is normally most useful for DSP filters.

BU4 (4th order slope) is recommended giving a very good separation between drivers. Higher orders/slopes are possible, but seldom necessary. Increasing from BU2 to BU4 will need another Biquad. The number of Biquads may be limited in your DSP device.

Figure 35 - Cross over (X-over) order/slope selection

d. General DSP Element
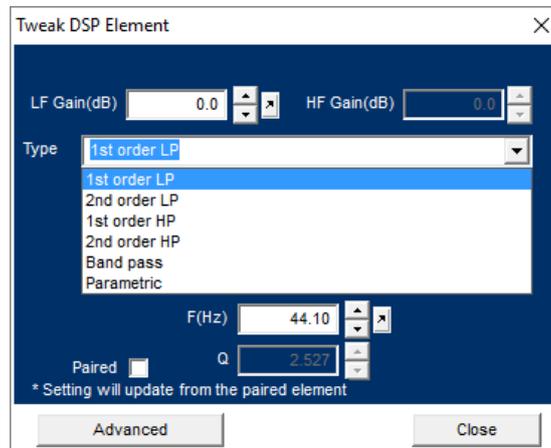
Figure 36 - General DSP Element

The General DSP element will insert a 1st or 2nd order LP or HP. Also, Band Pass and general Parametric filters can be inserted.

e. Level Control (dB)

The level control can adjust the level by a specified amount of +/- dB. This feature is useful for adjusting the level of a tweeter, which is often more sensitive than the woofer in a system.

f. Digital Delay

This will insert a digital delay as a specified number of samples, normally at 48 kHz sample frequency.

12.   Display – Totals / Sections

The display can show a great number of curves. Use the lower tab to view Total or Section responses. The table below explains the function of each button.
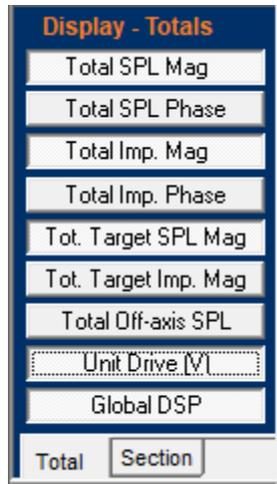
Display – Totals

	Totals	Sections	
1	Total SPL Magnitude	Section SPL Magnitude	1
2	Total SPL Phase	Section SPL Phase	2
3	Total Impedance Magnitude	Section Impedance Magnitude	3
4	Total Impedance Phase	Section Impedance Phase	4
5	Total Target SPL Magnitude	Driver only SPL Magnitude	5
6	Total Target Impedance Magnitude	Driver only Impedance Magnitude	6
7	Total Off-axis SPL	Target Magnitude	7
8	Unit Drive (V)	Unit Drive (V) or (I)	8
9	Global DSP	Section DSP	

13.   Cursor

Click anywhere on the main curve (White) and the cursor is shown for that frequency point, see figure below:

Right-click to enable the relative cursor from the point where you clicked. Right-click to go back (ABS).
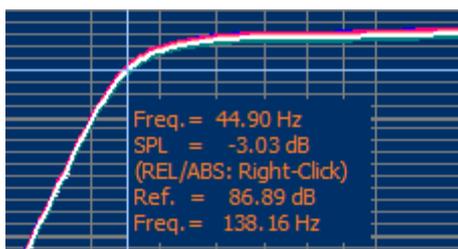
12. Printing

Select Print or Print Preview under File. Figure 37 shows the first page of the print. The power input and Biquads coefficients are also shown.

When using 3-way or more, these will be printed on additional pages.
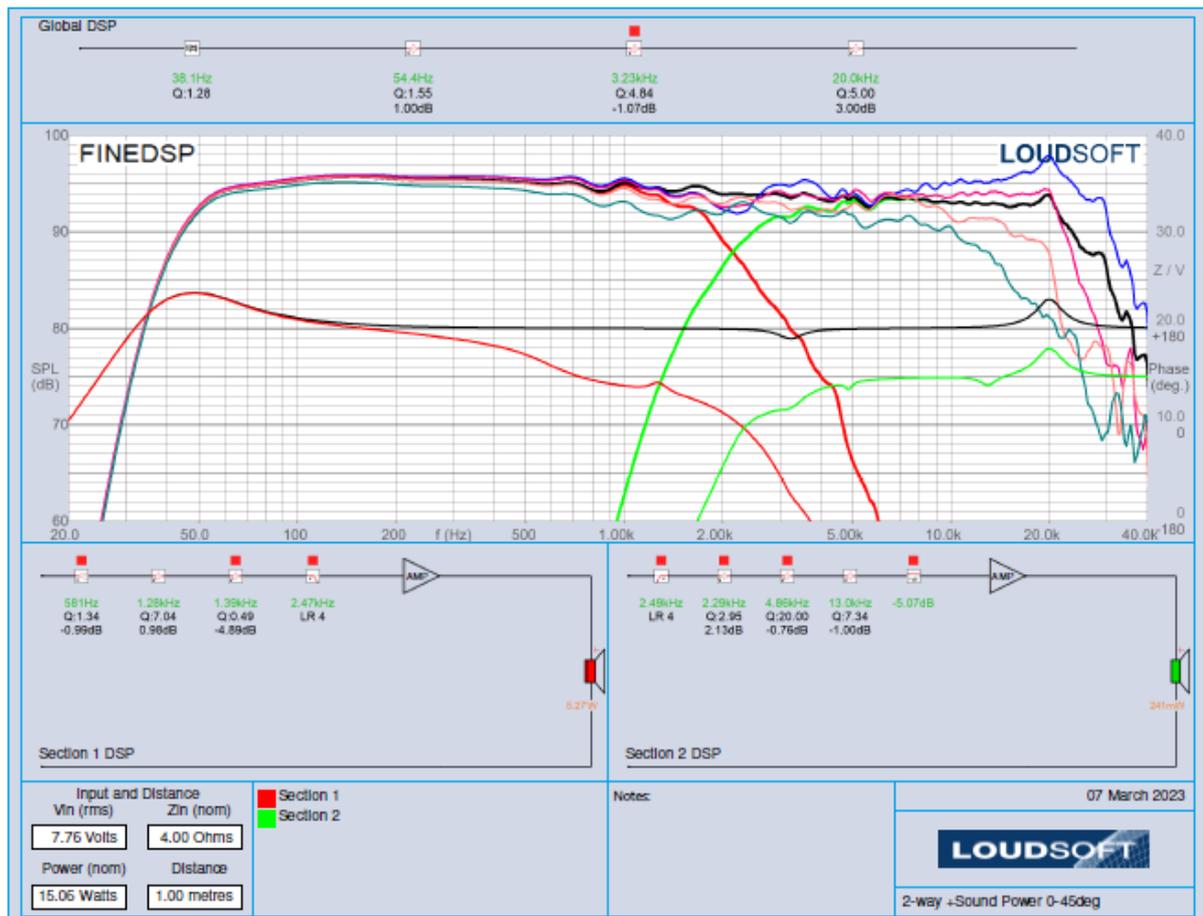
Figure 37 - Print example (first page)

13. Settings
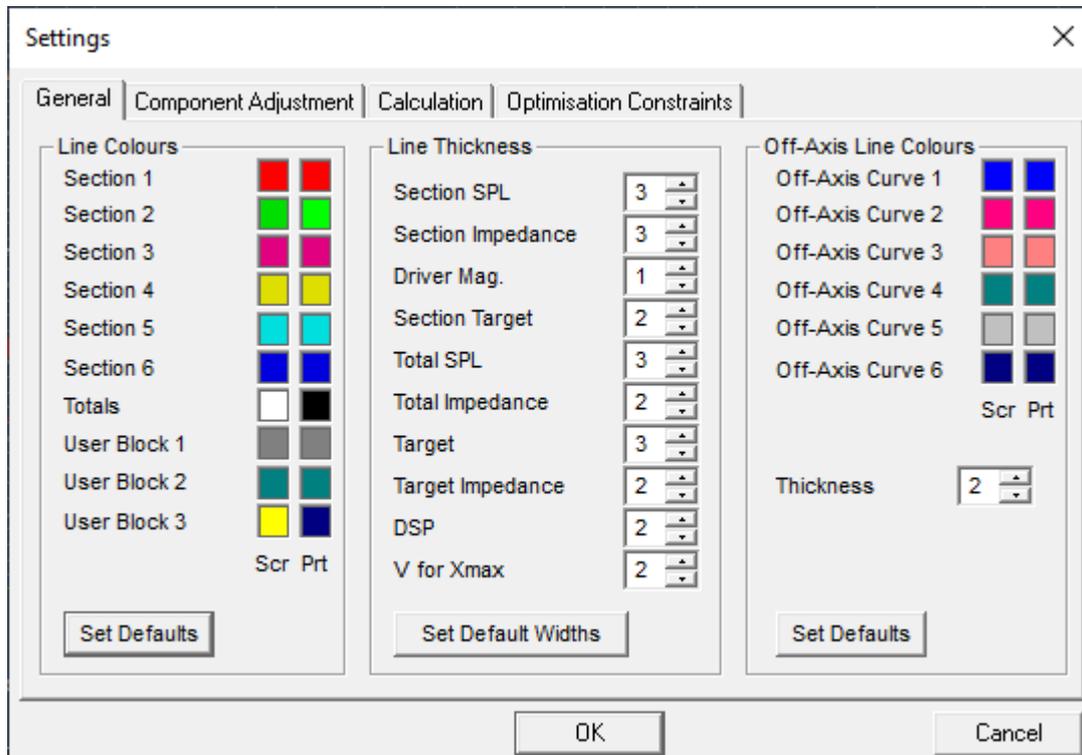
Figure 38 - Settings – General colors and line widths
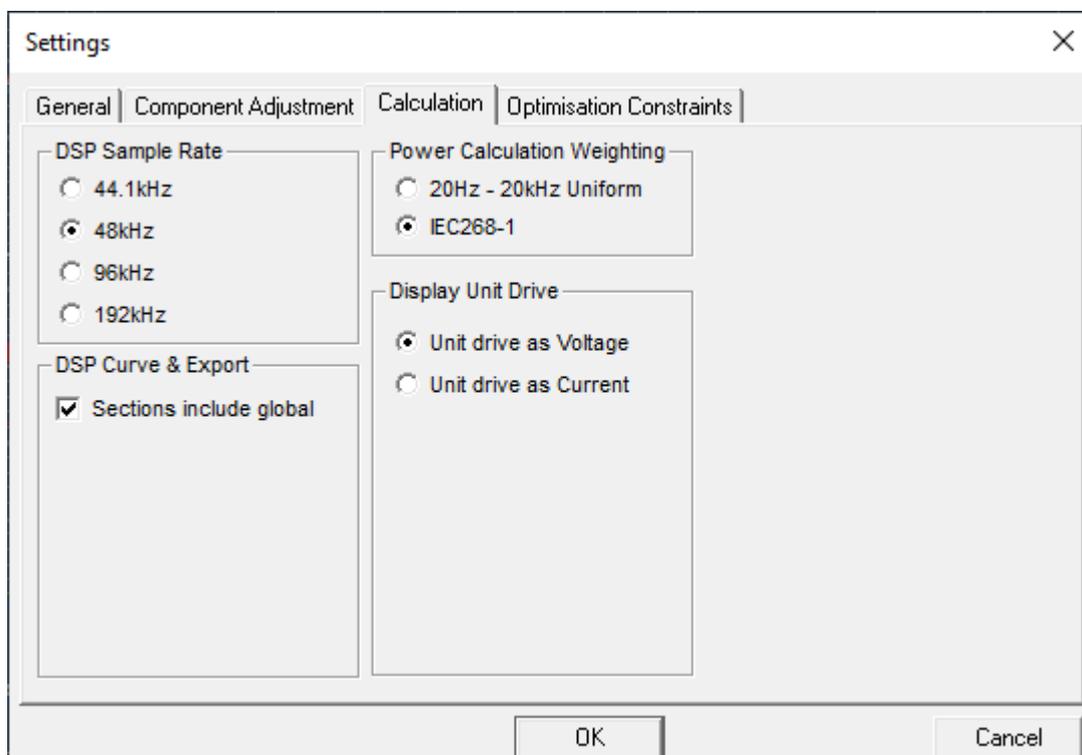
Figure 39 – Calculation settings for Sample Rate, Power, Voltage, Current

Please be sure to set the correct DSP Sample Rate as per Figure. 39
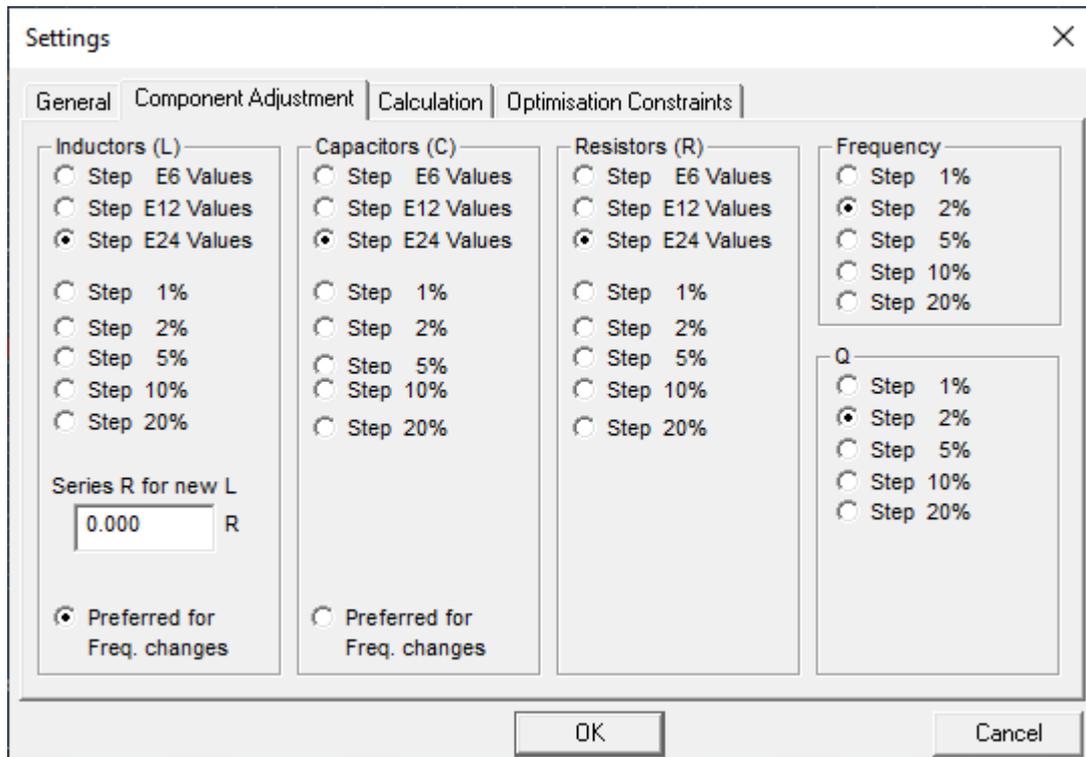
Figure 40 - Passive component step sizes

Optimisation Constraints

Figure 41 - Set Optimization Constraints

Appendix A

FINE DSP uses the Biquad method to implement the various filters available. Higher order filters are implemented by cascading multiple Biquad filters. The Biquad filters are calculated from following equation:

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{a_0 + a_1 z^{-1} + a_2 z^{-2}}$$

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