

## 9 A small tutorial

This section is a very small tutorial about different aspects of loudspeaker construction. As explained before the intention is not to explain everything. Instead the idea is to only to give a brief introduction to the fantastic world of loudspeaker building. Also the level of this tutorial demands that you are not totally new to this subject.

### 9.1 Loudspeaker boxes

The figure below shows the frequency response when the PEERLESS 6.5" driver unit 832732.unt available in the examples subdirectory is put into different boxes. In all cases the table lookup optimization in LspCAD has been used.

The resulting total volumes are quite moderate except for bandpass type 3 where the total volume is 60l, something that definitely is not acceptable for a 6.5" driver unit. On the other hand one can see that this box extends the frequency response to very low frequencies, it must be stated here also that the transient response characteristics of this box are very poor. A compromise is the bandpass type 1 box. This box extends the frequency response a bit, however the efficiency is lower as a relatively wide bandwidth (35-150Hz) was selected.

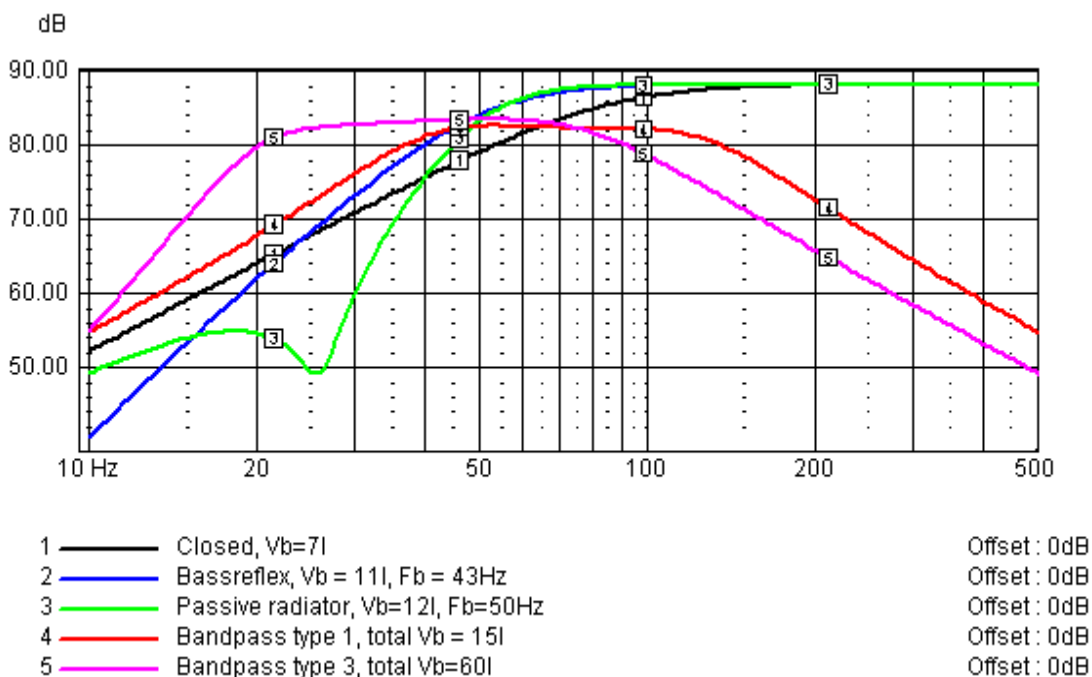


Fig. 23 A Peerless 832732 driver unit in different boxes

### 9.1.1 Transient response characteristics

The difference between the closed box and the bassreflex box is that the bassreflex box can reproduce lower frequencies at the cost of a sharper cutoff slope. As the closed box is a 2<sup>nd</sup> order system the slope below the cutoff frequency is 12dB/octave. The bassreflex box is a 4<sup>th</sup> order system, as a consequence the slope in this case is 24dB/octave. Consequently as the bassreflex box is a higher order system the transient response characteristics are worse than for a closed box.

The passive radiator (or ABR) box has an interesting notch in the frequency response below the cutoff frequency. This notch appears at the free air resonance frequency (Fp) of the passive radiator unit, in this case 26Hz. The passive radiator box is a 6<sup>th</sup> order system (due to the notch), thus one can expect the passive radiator box to have worse transient response characteristics. In figure 24 one can clearly see that this is indeed the case. The passive radiator box shows a considerable longer tail after the toneburst is interrupted.

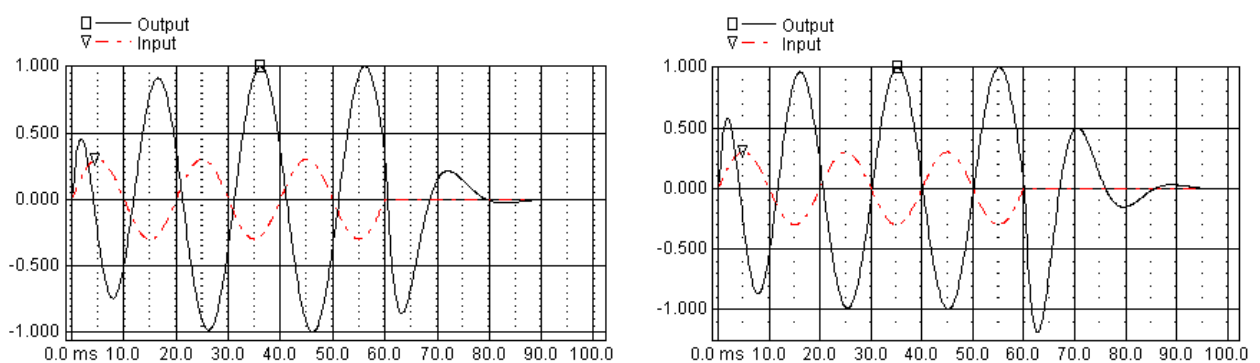


Fig 24 Tone burst response for a bassreflex and a passive radiator box, input signal is shown dashed. Frequency 50Hz, 3 periods.

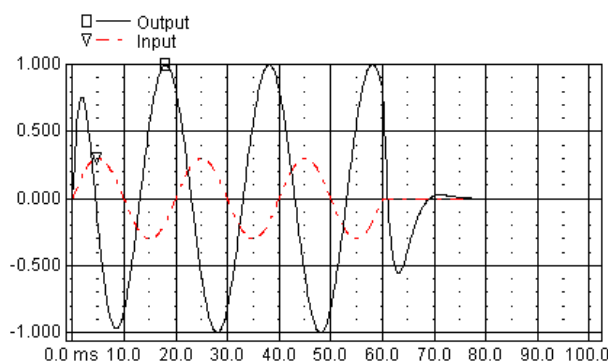


Fig 25 Tone burst response for a closed box.

Speaking in general one can say that the higher the system order, the worse the transient response. With this at hand we can make a list of transient response quality with closed box being the best:

- 2<sup>nd</sup> order systems  
Closed box

- 4<sup>th</sup> order systems  
Bass reflex, band pass type 1
- 6<sup>th</sup> order systems  
Passive radiator, all other bandpass boxes, double tuned bass reflex.

One interesting thing is that a bass reflex box with a bump in the frequency response may in some cases show a better step response than a bass reflex box with a flat response.

A tone burst should be built up fast and die off fast. A closed box builds up and stops this tone burst very quickly while e.g. a bandpass box type 3 may show a very slow behavior.

How does a good transient response sound ?

The answer may be that a good transient response should not be heard at all. The major problem with boxes with poor transient properties is that the punch in the transients is smoothed up. The ringing after the transients are not an equally big problem as the ear is not as sensitive to this due to temporal masking effects in the auditory system.

### **9.1.2 Power handling**

Another aspect, equally important when the type of box is selected is the power handling.

One can compare the cone excursion between a closed box and a bassreflex box. While the cone excursion for a closed box increases up to a certain limit as frequency gets lower the bassreflex box shows a notch in cone excursion at the port resonance frequency. This has implications on both power handling and distortion and makes the bassreflex box able of handling higher power levels than a closed box. The above statement applies only to steady state sinewave signals however. If a toneburst signal of the same frequency as the port resonance is applied to a bass reflex box, the cone excursion will initially be quite high and will reduce to a small value after one half to a couple of periods. This is due to the fact that the port-box system can be viewed as a spring with a weight connected to it. At first we need quite an effort to get things going, but after a while the system will oscillate by itself. With this in respect we realize that for transient rich bass sounds the cone excursion is initially equal to or sometimes greater than for a closed box!. Also if the bassreflexbox is subject to inputs with frequencies lower than the port resonance (e.g. rumble from turntables) the driver is very easily overloaded with large cone excursions even at low power levels.

Fig. 26 below shows the cone excursion for the closed and the bassreflex box, in an ideal situation the cone excursion would increase indefinitely for the bassreflex box as the frequency decreases, this is not the case in fig 26. The reason is that nonlinear simulation is turned on. In this case it is the progressive stiffness of the driver unit that limits the cone excursion.

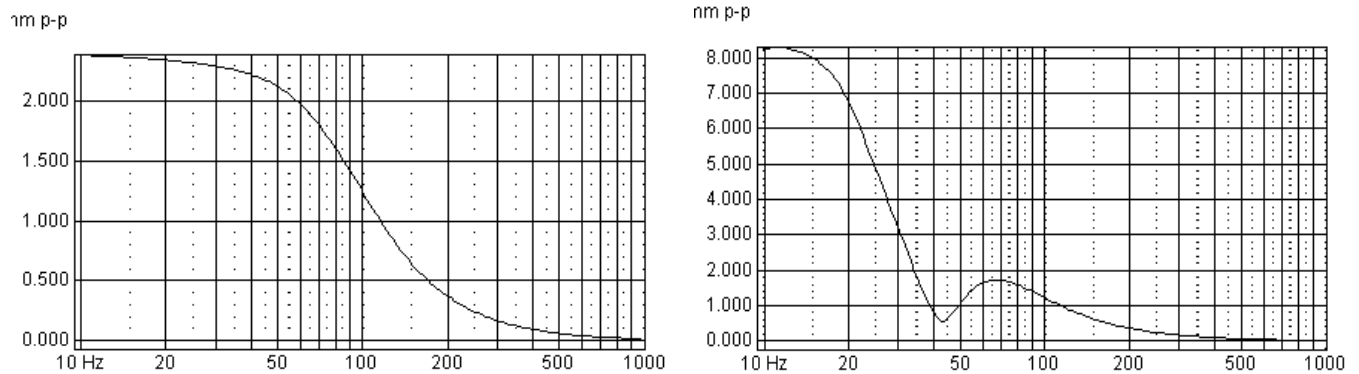


Fig. 26 Cone excursion for the closed and the bassreflex box

One drawback with ported boxes are that the vents are prone to producing standing wave resonances and also chuffing noise at high power levels. Chuffing noise at working levels is solved by increasing the area of the port. A rule of thumb is that the port air speed should not exceed 15m/s.

Interesting thing is the handling of high power levels. Fig. 27 shows the sound pressure level for different input levels for the bassreflex box described above. The curves are normalized, i.e. the reading for curve (3) is actually 20dB higher.

From the figure one can see that at 100W input the output level is compressed about

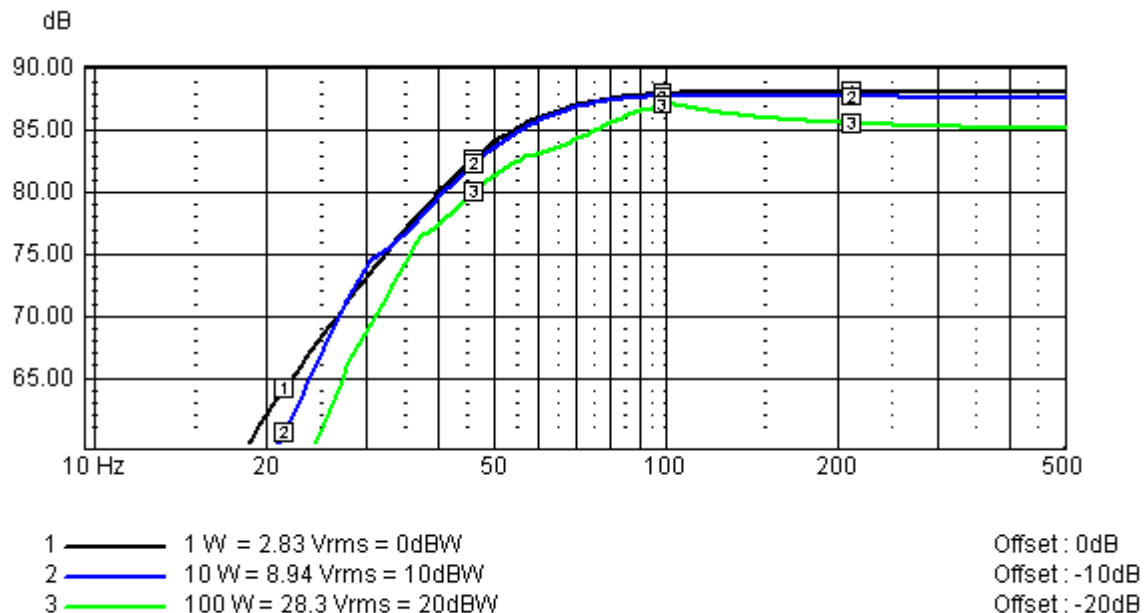


Fig 27. Sound pressure level for different input levels, note that the curves are offset by a value corresponding to the input level. The level for curve (3) is actually 20dB higher than the reading.

3dB at higher frequencies. This compression is due to the heating of the voice coil. At lower frequencies the compression is about 2-5dB and depends also on the non linearity in force factor and compliance. Note that this is only a model that depends on the values  $X_{max}$  and  $P_{max}$  given in the driver unit editor.

### **9.1.3 Some construction tips**

Simulation and modeling is one side of the coin, in the end one also need to construct the boxes. For the cabinet or enclosure it is very good to use Medium Density Fiberboard (MDF). MDF is easy to work with and has very good acoustic properties. For the front baffle it is good to use double 19 mm board for medium sized boxes. Remember though to cut the loudspeaker hole of the inner board bigger than the mounting hole on the outer board. This will limit the risk that the mounting hole forms a long tube that may cause resonances.

It's generally better to use a high and narrow box rather than a square shaped box as the medium and high frequency range is better reproduced if the driver units of the latter is higher above the floor. Of course one does not need to follow this advice if the loudspeaker is mounted on speaker stands.

The most common damping material is fiberglass. The main purpose of damping material is to limit standing wave mode resonances in the box which occur when one inner size of the box is equal to a multiple of a half wavelength. For example if the inner height of a box is 1m then one gets the first mode as low as 170 Hz !. This mode is actually best dampened out with heavy stuffing of damping material halfway up the box as the particle velocity is highest at this point for this resonance mode. Damping material along the inner walls seldom does much for the damping of low frequency standing wave modes inside a box.

### 9.1.4 Cabinet edge diffraction

One property that almost always shows up when a driver unit is mounted in a box is something called cabinet edge diffraction. Without going too deep into the theory about this effect it can be said that the results of it is that the sound pressure level increases 6dB from low to high frequencies. This is referred to as the baffle diffraction step and there is no way of avoiding this effect other than mounting the driver units into the wall of the living room. The worst placement for a driver unit is actually in the middle of the baffle. Fig. 28 shows the baffle diffraction step for a worst case and a best case placement of a 6.5” driver in a normal-sized baffle. It is

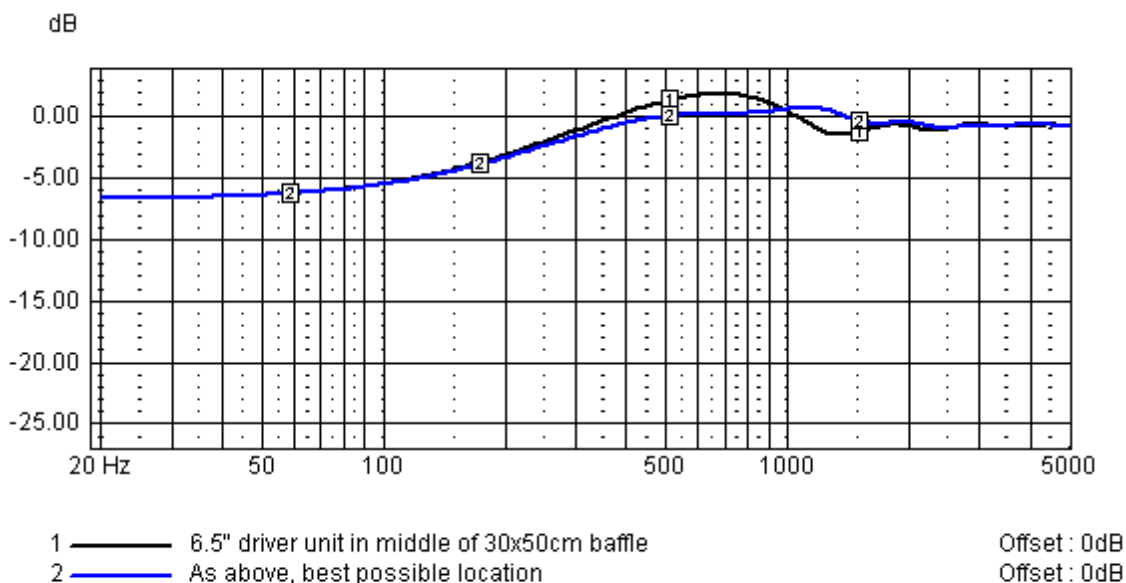


Fig. 28 Baffle diffraction step for a 6.5” driver unit mounted on a 30x50cm baffle, both worst case and best possible case placement is shown.

obvious that one can gain a lot by placing the driver unit correctly. This experiment can be performed by means of the room/cabinet function in LspCAD.

### 9.1.5 Room interaction

A loudspeaker is intended to be placed in a listening room, and this fact may be good to take into consideration. Fig. 29 shows the simulated frequency response of a loudspeaker “measured” in free field (far away from any boundaries) and when

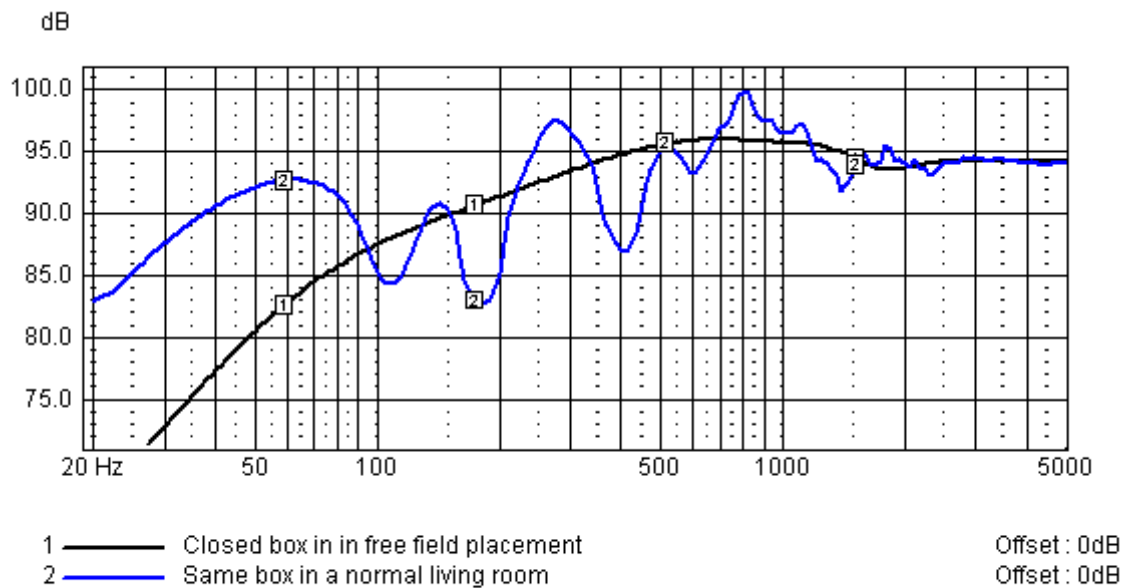


Fig. 29. Simulated loudspeaker response, both free air and in living room.

placed in a living room. The most noticeable effect is that the low frequencies are amplified about 10-15dB. Does it then sound good if one strive for a flat response in a living room. Too often the answer is, no!. I (the author of this text) have the feeling that, for much modern music the producers seem to be aware of this “room transfer function” and they consequently mix the music according to this. Most likely though classic music of live recordings of Jazz music is reproduced better if the room is taken into account when the loudspeakers are constructed.

## 9.2 Crossover networks

In the beginning there was the “one crossover for any combination of loudspeakers”. At least this was what the DIY (do it yourselves) had to resort to when a computer was something that almost nobody could afford. With computers came measurement systems that made it possible to import measured data to crossover design programs and thus the path was open for DIY’s to construct crossovers for loudspeaker. Before we continue, we may as well state that it is necessary to have a measurement system so that one can measure the driver units of a loudspeaker separately, preferably mounted in the box that they are intended for. In the below text it is assumed that the advanced active or passive crossover utilities are used.

### 9.2.1 Filter order and alignment

When selecting filter order and alignment one must bear in mind that the individual driver units of a loudspeaker sum up in a way that the total frequency response is as flat as possible (not always true). Important is that the driver units overlap well at the crossover point. In many crossover software one can select from a great number of alignments, Chebyshev, Butterworth, Linkwitz-Riley, Bessel, Elliptic with their own magnitude and phase properties just to mention a few. What is easily forgotten is that the individual driver units has its own characteristics. Fig. 30 shows the magnitude

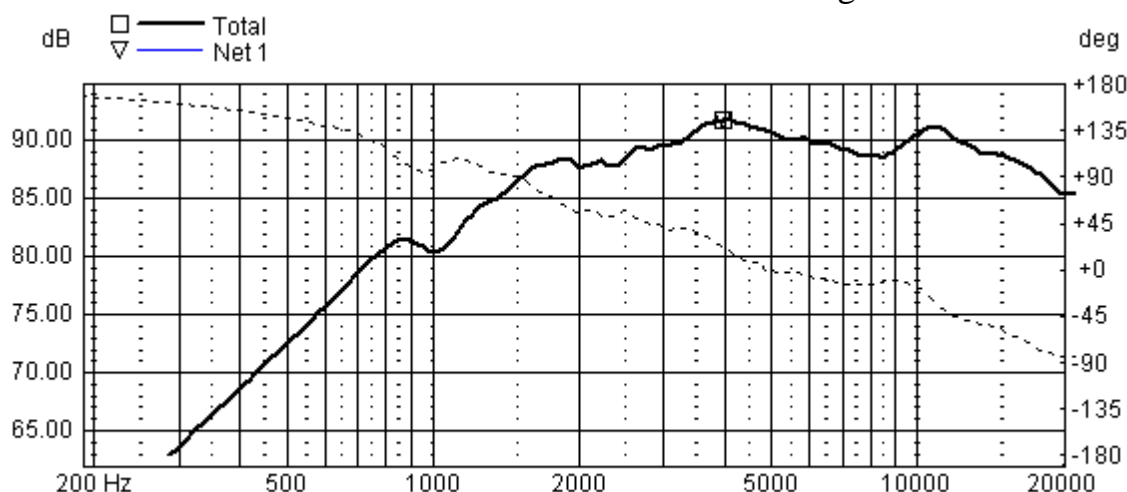


Fig. 30 Magnitude (solid) and phase (dashed) of treble unit Dynaudio T-330D mounted in a box

and phase response of a treble unit mounted in a box. One can see that not only is the magnitude variant but also is the phase.

Many crossover design tend to ignore this fact, and that is like constructing a car without steering wheel assuming that the road is straight with no curves!. The conclusion is thus that the important thing is not the response of the crossover, what is important is that the driver units overlap. Fig. 31 shows the frequency response of the crossover network and the driver unit+crossover when LspCAD has optimized it for a 4<sup>th</sup> order Linkwitz-Riley alignment with cutoff frequency 1500Hz



Note that while the total frequency response (driver unit+crossover) follows the

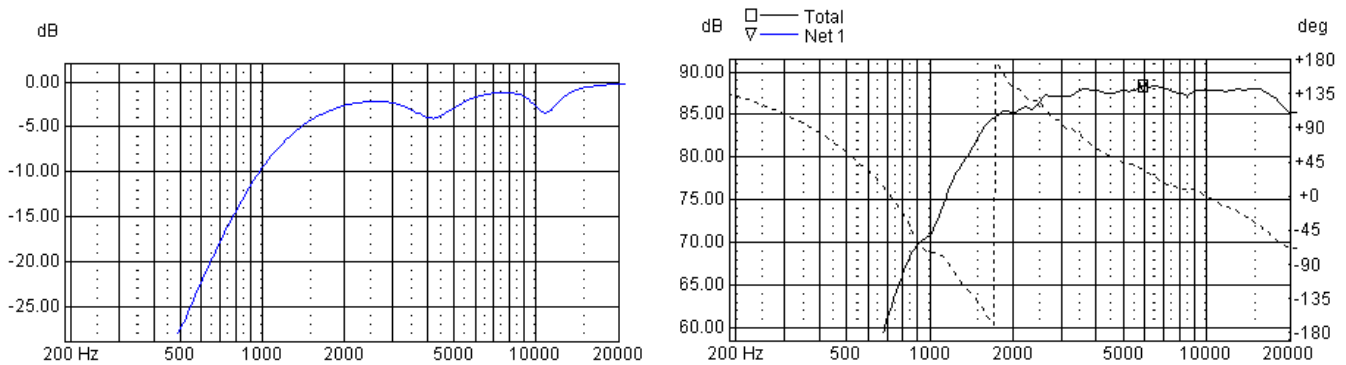


Fig. 31 Frequency response of crossover and crossover+driver unit.

Linkwitz-Riley target (-6dB and almost -180degree phase at 1500Hz) the crossover network definitely does not do it. The above solution was realized using an active crossover, this means that the impedance data is not needed as the driver unit is fed by a low impedance power amplifier. Should a passive crossover have been used, then of course impedance data would have been needed also. Conclusion is then that driver unit data is needed, period.

As the driver unit characteristics is an important part of the total result it is now stated that when referring to the term slope order in the optimization phase (and in the text below) is meant the order of the total slope of the driver+crossover. It may well be the case that it order to realize a 4<sup>th</sup> order slope one may settle for a 2<sup>nd</sup> order filter as the driver units response display a 2<sup>nd</sup> order slope itself.

The optimizer in LspCAD can achieve both Linkwitz-Riley and Butterworth alignment of order 1-8. The phase properties of the different alignments and orders are shown in the table below. The Linkwitz-Riley alignment is -6dB down at the crossover frequency and the Butterworth alignment is 3dB down at the crossover frequency then steepness of the slope is “6dB per order” i.e. a 3<sup>rd</sup> order slope has a steepness of 18dB/octave. The phase-lag at the crossover frequency is dependent on the order according to the relation

$$\text{phase} = N \cdot 45 \text{ degrees},$$

where N is the order. For example a 2<sup>nd</sup> order slope has a phase-lag of 90degrees at the crossover frequency.

One cannot combine orders and alignments freely. It is recommended to use the same slope order on both sides of the crossover frequency. Note that it is possible to use different slope orders but it is more complicated to succeed with this and not recommended for the beginner. As the Butterworth alignment only is 3dB down, it is demanded that the difference in phase between the highpass and lowpass is 90degrees in order to get a flat response. In case the Linkwitz-Riley alignment is selected the difference in phase should be either 0 or 180 degrees. The table below displays recommended alignments for different orders. In some cases one of the loudspeaker terminals must also be inverted.

<b>Order</b>	<b>Alignment</b>	<b>Invert</b>
1	Butterworth	No
2	Linkwitz-Riley	Yes
3	Butterworth	No
4	Linkwitz-Riley	No
5	Butterworth	No
6	Linkwitz-Riley	Yes
7	Butterworth	No
8	Linkwitz-Riley	No

## 9.2.2 Measuring driver units

Measuring the driver units is a bit complicated, especially for DIY's with limited access to anechoic rooms. Luckily enough many loudspeaker measurement systems makes it possible to measure either with gated measurements or with time windows.

FFT based systems such as LAUD, CLIO or ETF makes it possible to get the absolute phase. In other systems such as LMS the phase response is extracted from the magnitude data by means of minimum phase (Hilbert) transformation.

In order to make for accurate modeling in LspCAD (and other crossover modeling software for that matter), the following method is recommended to use.

1. Start with measuring the impedance of the individual driver units, note that both magnitude and phase is needed in LspCAD.
2. When measuring the frequency response of the driver units. The microphone must be placed at a distance from the driver unit, directly on axis. This is especially important for midrange and treble units as the near field and far field response are very different due to cabinet edge diffraction effects. It is recommended to use a distance of about 1m, in some cases one can use a distance of 60-70cm. It is recommended to use a microphone stand so that the microphone can be raised and lowered so that it is always in front of the driver unit being tested. Remember to keep the distance between the microphone and the **baffle** constant throughout the measurements.

3. If FFT based systems are used, the phase is already available, remember though that the start of the analysis window must be kept constant. It is good practice to put the start of the analysis window so that the phase-lag that depends on the distance is removed. If minimum phase extraction is to be used, remember to do tail correction before.
4. The measured frequency response must be verified in some way. A method commonly used is to do a final measurement with all driver units connected together and the microphone placed in front of the treble unit (at some distance).

The measured response will look awful but don't let that scare you off. Create an advanced passive crossover and import both the measured frequency response and impedance data. Remember also to set the positions of all the drivers to the positions relative to the location of the treble unit, let  $dZ=0$  for all driver units. This of course means that the position of the treble unit is  $dX=dY=dZ=0$ . Moreover set the position the microphone, that is the distance between the microphone and the baffle.

Now take a look at the *Summed frequency response* diagram, does it look similar to the real measurement. If yes then we can continue with the crossover construction. If no, then we have to set the  $dZ$  values for some of the driver units. This should not be needed if absolute phase measurements from an FFT based measurement system is in use. If minimum phase data is used (e.g. LMS data) then we need to put a  $dZ$  value of about 2-3cm for the bass units. The simple reason is that the sound from a bass unit mainly comes from the middle of the driver unit at higher frequencies, therefore this sound will be delayed a little bit extra due to the cone shape. Often it is possible to get  $dZ$  by measuring the depth of the driver cone i.e how recessed the dust cap is relative to the rim of the cone.

Afraid of destroying your treble units with this exercise? Put a capacitor in series with the treble unit, remember though to add the capacitor in LspCAD as well.

### 9.2.3 Constructing zobel networks

The first thing to do (probably), before starting with the real crossover work, is to make the impedance curve more apt for the rest of the crossover, this is not needed for active crossovers. The way to deal with the series inductance  $L_e$  that cause the impedance to increase at higher frequencies is by means of a zobel network, i.e. a resistor and a capacitor in series. It can be stated that it is not always necessary to use zobel networks, but it is experienced that the network optimizer (at least in LspCAD) performs better when a zobel is used. The best way to get an optimal zobel network is to create an advanced passive crossover in LspCAD and import the impedance data. The network dialog for one of the networks is opened and Branch 11 is changed to an R1+C type. As starting value for R1 is selected a value that is about 20% higher than the minimum impedance before impedance correction.

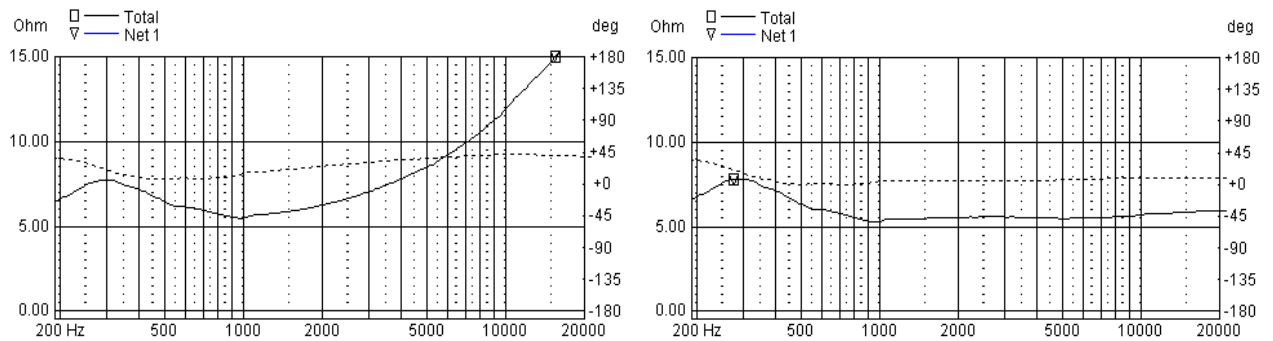


Fig. 32 Impedance curve for a midrange unit before (left) and after (right) impedance correction.

After these initial values it is only to trial and error change the values of both R1 and C. It is good practice at this time to use the arrow keys and change the values while looking at the impedance diagram. This manual method is quite fast, about 2 minutes later one have a good zobel network. Note that it is possible that during the optimize phase one can change the values of the zobel network in order to improve the total frequency response.

### 9.2.4 Constructing the rest of the crossover

After the zobel networks are created it is only to start with the rest of the crossover network. Below is the frequency response of the bass and treble unit for a commercial loudspeaker.

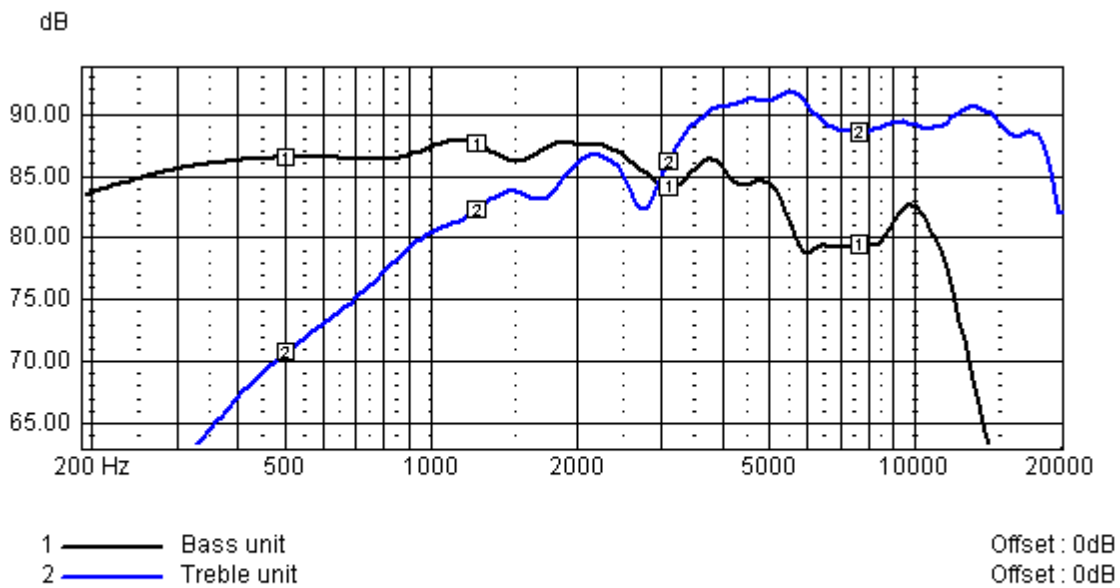


Fig. 33 Frequency response of bass and treble unit

The goal is here to create a crossover that makes for an as flat response as possible in the in the working range. From the frequency response one can see that the optimal crossover point is set at about 4000Hz, this may change during the optimization process. For convenience a 4<sup>th</sup> order alignment seems to be good. Also a series resistor for the treble unit seems to me needed. The network dialogs are opened and the wizard is asked for appropriate starting values. Fig 34 shows the results after the

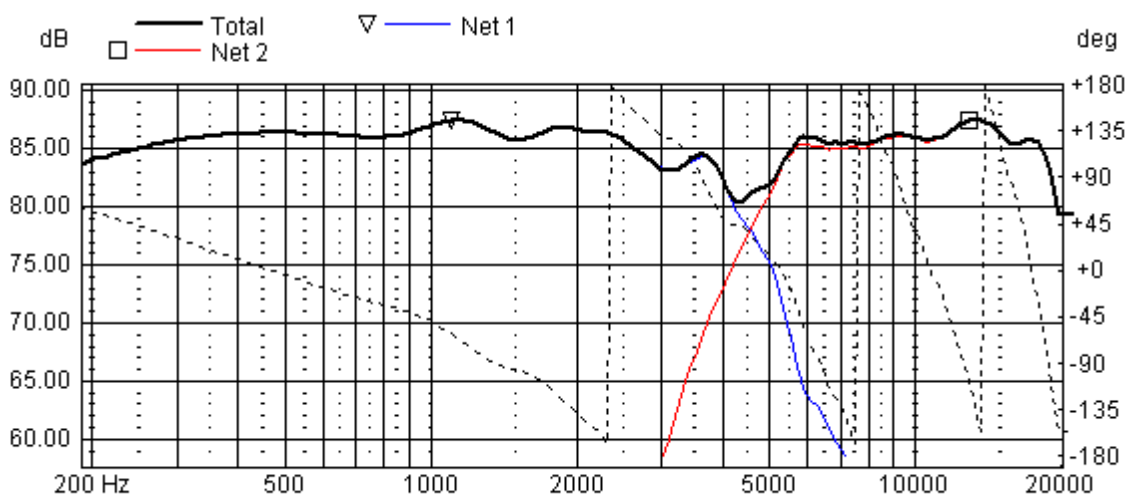


Fig. 34 Frequency response of system when starting values from wizard has been retrieved.

wizard has done its work. The results are quite OK, but not perfect. The next step is to start the optimizer.

In the optimizer the desired alignment and slope order is entered. In this case a 4<sup>th</sup> order slope and Linkwitz-Riley alignment was selected. Also it is important to select the include range so that the error metric of the optimization is only computed over the frequency response of interest. Consequently the include range for the bass network is set to 300-7000Hz and the include range for the treble unit is set to 2500-

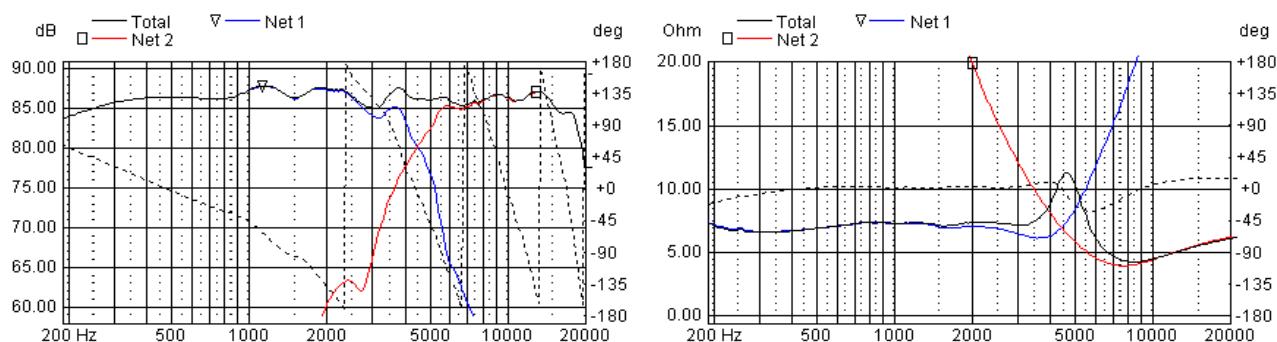


Fig. 35 Frequency response and impedance curve after optimization and some additional tweaking .

18000Hz.

In fig. 35 the resulting frequency response and impedance curve is shown. The results are quite good. One thing that is important to observe though is that the impedance does not get too low. In this particular case the impedance got very low when the optimization was allowed to run, in particular it was the crossover for the treble unit that caused the problems. Therefore some additional manual tweaking was needed in order to get rid of these problems. These impedance problems are quite common when parallel topology networks are used.

Last but not least important, some words about power handling. In fig 36 is shown two frequency responses, one for low power levels and one for high power levels.

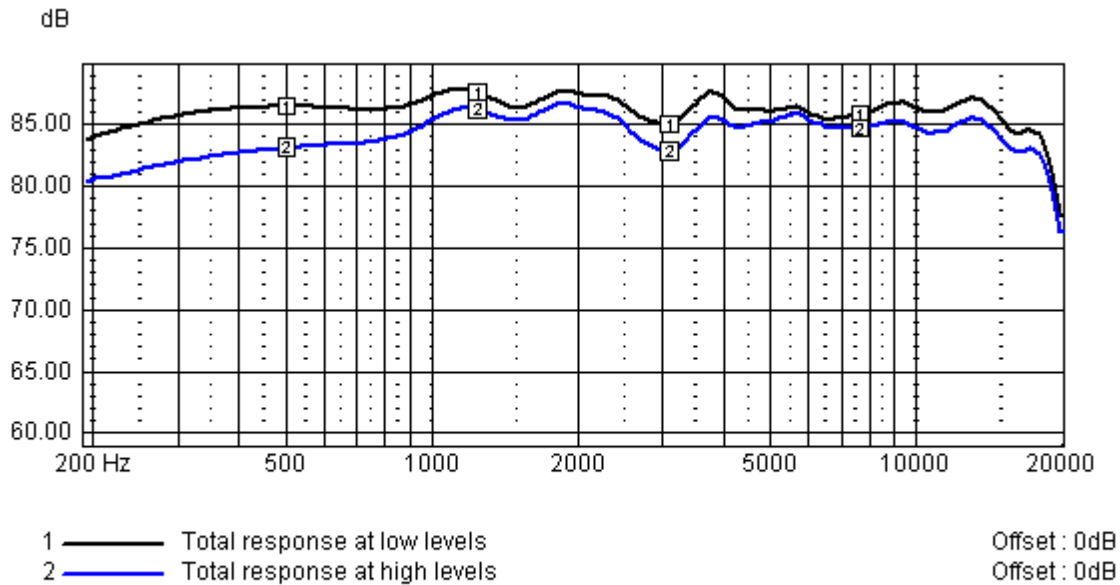


Fig. 36 Simulated (normalized) frequency response for low and high power levels.

The latter is achieved in LspCAD by setting a value other than zero in the Rex field in the driver unit dialogs. For the bass unit an increase of the voice coil resistance of 50% was simulated, while a 25% increase was simulated for the treble unit. The result that can be observed is that the response around 6000Hz is raised. This may give a yelling character at high sound pressure levels. If the loudspeakers are intended for PA use, then it is recommended to do the optimization for the working levels and not for low levels.

This is the end of this tutorial, much longer than originally intended, hope this helps you as a beginner to get started with the loudspeaker building.

And last ..., watch your ears and your thumbs.