## 1 Introduction

Updated 2005-01-26

This document gives a few examples how LspCAD 6 is used in a real situation. This document is meant to expand with more examples as time goes by. All the examples are available as project files in the examples folder.

Driver set

1.000uF

0.200mH

Bass

Treble

0.200mH

# 2 An optimization example, a two way

La Schematic, page 1

Generator 1

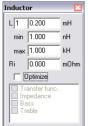
1.000V

Edit Show list

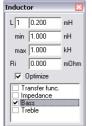
▾

crossover

In this example we have a simple two-way crossover (**Two way tutorial 1.lsp**). We see that L1 and C1 only affect the response of the Bass unit. Open up the advanced settings for L1.

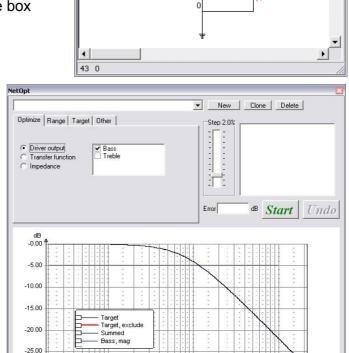


Check the Optimize box and also the box next to "Bass".



We have now instructed the optimizer that L1 should be optimized when we start to optimize the response of the Bass unit. Do the same with C1. In a similar way we see that C2 and L2 only affect the response of the Treble unit. (Two way tutorial 2.lsp)

Open the Optimizer, we choose to optimize the response of the Bass



unit, we therefore click on the box next to "Bass", if we look at the schema we will see that the component text for L1 and C1 has become boldface.

Click on the Range tab and set the Include range to the interval 100 to 6000Hz



Next we select an appropriate target for the optimization, click on the **Target** tab and then on the **LP** tab, check the Enable box and set **Fc** to 2000Hz and **order** to 2, also set the alignment to Linkwitz. With this we are ready to start our optimization.

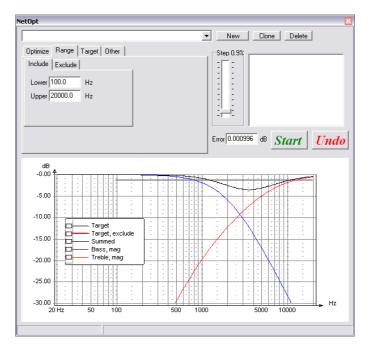
Click on the **Start** button and watch the miracle happen. Optionally one can turn up the step size to get a faster convergence. When the stop the optimization L1 and C1 are roughly 0.160mH and 39uF and the mean error is close to 0.01dB. For some reason it is possible that one did not like the result, the remedy is simply to click on the

▼ New Clone Delete Optimize Range Target Other Zmin X0 points Level+other Box vol LP | Peak/notch | Imported | Slope Fc 2000.0 Hz ▼ Enable order 2 Linkwitz • dB Start Undo dB -0.00 -10.00 -15.00 -20.00 Bass, mag -25.00 -30.00

**Undo** button to get back to the state before the **Start** button was clicked.

In the similar manner we can optimize the response of the treble unit for a 2<sup>nd</sup> order HP Linkwitz alignment at 2000Hz, but instead we test what the lock XO option can do for us.

We check "Treble" in the optimize tab, also wee set the target to flat, and the range to 100 to 20000Hz and the optimizer screen looks like

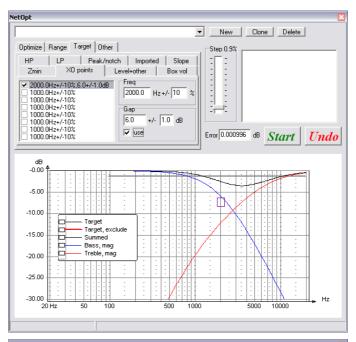


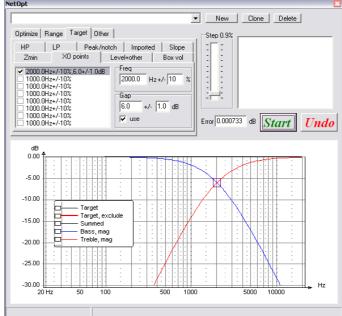
#### LspCAD 6 tutorial

We open the XO points tab and put a lock at 2000Hz with a gap of 6dB with a slight tolerance. And start the optimizer, but before we do so we can disable the optimization of L1 and C1. (**Two way tutorial 3.lsp**)

After a while we have a crossover with a flat system response and a crossover frequency locked at 2000Hz. (**Two way tutorial 4.lsp**)

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# 3 A closed box... and a bass reflex box

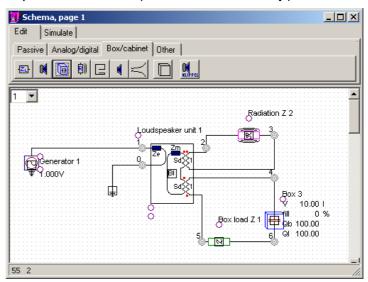
At first glance the modeling of various loudspeaker boxes might look overly complex, the intention is however that it should be possible to model other, more complex boxes than the standard closed, bass reflex and passive radiator boxes. This section describes how modeling of a closed box is performed in LspCAD, this example is then extended with a bass reflex port.

#### 3.1 Closed box

First of all we need a signal source and a Loudspeaker unit, pick this from the component tray, we also need to ground one of the speaker terminals (**Closed tutorial 1.lsp**).

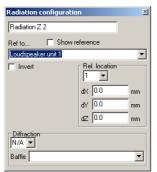
In a closed box we have free air in front of the speaker cone. This is modeled as a Radiation element. Pick a radiation element. This acts a load of the front part of the cone.

Behind the code we have a box and also a load from the air inside the box. This is modeled with a Box load and a Box component. After the components are picked and arranged a little on screen (don't use too little space) we have a schema that looks something like (Closed tutorial 2.lsp).



So far we have been in edit mode, now it is time to enter simulate mode. Said and done, we click on the Simulate tab.

In this mode we need to do a few extra things before we are in business.



The Radiation component needs to know a little about the loudspeaker unit (such as Sd). For this to happen we click on the Radiation component and a small configuration dialog appears. Click on the dropdown list box below "Ref to..." and select "Loudspeaker unit 1" (nothing else to select by the way). Once you have done this you will see a typical 2<sup>nd</sup> order high pass response of a closed box in the graph window.

We are getting close but we are not quite there yet, recall that the Box load



component models the load behind the cone, for this to happen the Box load component must also know a little about the

loudspeaker unit. Click of the Box load component and set the reference to "Loudspeaker unit 1" as before.

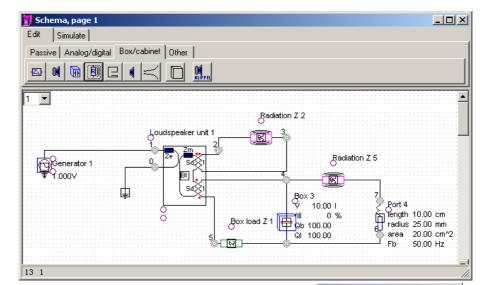
Now we are in now in practice done (**Closed tutorial 3.lsp**). We are free to modify the Box parameters with a click on the Box component. The T/S parameters can be changed via a click on the Loudspeaker unit.

#### 3.2 Bass reflex box

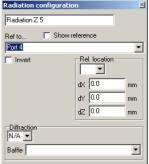
But lets not be lazy, why don't we just make a bass reflex box?. Enter the Edit mode again and pick a Port component and an additional Radiation component from the component

tray. We need a Radiation component as the port is a component that radiates into free air. The schema then looks like (Bass reflex tutorial 1.lsp).

Now we go back to Simulate mode and we realize that the SPL graph has changed (we have a notch in the response) In order

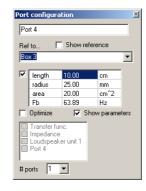


to get the full picture we need to let the extra Radiation component learn a little of the Port, click on the Radiation component and set the reference to the Port component



With this we have the ability to simulate a bass reflex box (Bassreflex tutorial 2.lsp).

It would be cool however if Fb in the schema displayed the correct value. For this to happen the Port must know how large the Box is, click on the Port component and set the reference to the Box. Now we get Fb to update whenever we change the port length or the box volume (Bassreflex tutorial 3.lsp).



## 3.3 Grouping the box parts together

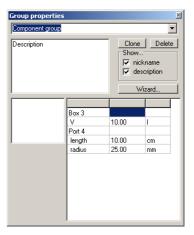
If we group all the components (except the loudspeaker unit, the ground symbol and the signal source) we get an additional benefit.

Enter the Edit mode, select all the components except those mentioned above, right click and select Group. Move the markers that "carry" the text "Component group" and "Description" a little.

Go back to simulate mode.

Nice thing now is that if we left click inside the group box (but not on a component. A dialog pops up that displays the most important parameters.

(Bassreflex tutorial 4.lsp).

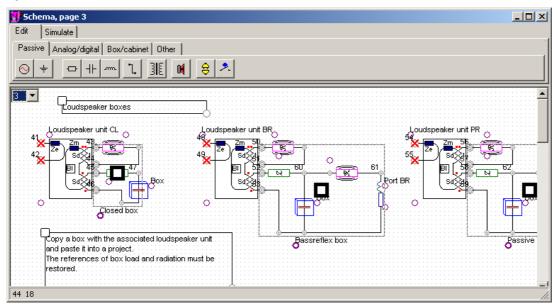


## 3.4 Using the templates

The closed box and the bass reflex box exist as templates, if the templates are used we get an additional nice feature, namely the wizards that help us to get decent values for box volumes and port lengths.

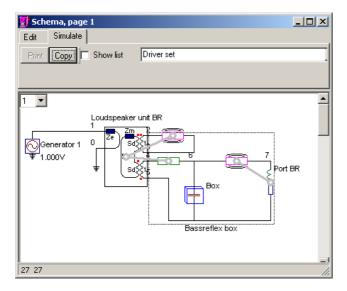
First of all create a completely new project. In the main window, click on the project list and locate the Templates. Select page 3 in the templates.

Copy the Bass reflex box (and the associated loudspeaker unit and paste it into the new project.



Now we must add a voltage source and a ground connection. When this is done. We enter the simulation mode.

The references to the loudspeaker unit must be set for the Box load and the Radiation component closest to the loudspeaker unit (this must be done by hand, hopefully not needed in the future).



Now that this is finally done we are able to simulate our beloved bass reflex box. If we right click inside the group we get a dialog that contains a magic wizard

button, click on the **Wizard** button and a small window pops up that allows us to

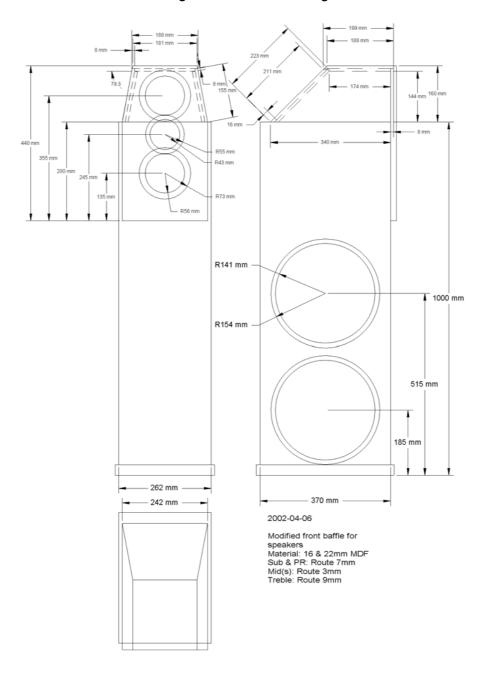


select among a few bass reflex alignments.

Select an alignment and click on Apply. (Bassreflex tutorial 5.lsp)

# 4 The Ugly duckling revisited

This chapter gives an example how LspCAD is used to create an appropriate crossover for the Ugly duckling loudspeakers, first presented at <a href="http://www.ijdata.com/ugly\_duckling.pdf">http://www.ijdata.com/ugly\_duckling.pdf</a>. This chapter should be viewed as a continuation and will also show how the Behringer DCX2496 is configured.



## 4.1 The subwoofer unit

The bass unit is a Peerless XLS 12 assisted by two Passive radiators

Driver unit		PR	
Manufacturer:	Peerless	Sd:	466.00 cm <sup>2</sup>
Model:	SWR308	Vap:	139.20 l
		Qmp:	15
Sd:	466.00 cm <sup>2</sup>	Mmp:	1000.0 g
Vas:	139.20 l	Fp:	6.80 Hz
Cms:	4.60e-04 m/N	Xsus:	25.0 mm
Cas:	9.94e-07 m <sup>5</sup> /N	Fb:	18.9 Hz
Mmd:	162.39 g		
Mms:	166.40 g		
Rms:	5.12 Ns/m		
Fs:	18.1 Hz		
BI:	17.60 N/A		
Re:	3.50 ohm		
Le:	1.60 mH		
Qms:	3.70		
Qes:	0.21		
Qts:	0.20		
Pmax:	500.0 W		
Xmax:	25.00 mm		
h:	8.00 mm		
1:	33.00 mm		
	(   DD : 405		

The original cone mass of the PR is 425g, therefore 575g weights was manufactured that was attached to the backside of the PR.

In order to model the passive radiator box it is actually a good idea to pick the Passive Radiator box template that is available in the Template section. The template section is actually a preloaded project that contains ready to use templates for a number of good to have building blocks. The template section is accessed from the topmost dropdown listbox in the main window. Select both the TS driver unit and the group that represents the passive radiator box. Right click with the mouse and select **Copy**.

Edit Simulate Passive Analog/digital Box/cabinet Other 3 -Bassive radiator box oudspeaker unit PR 🔘 BI ( 0.437 Ns/m h 10.00 cm 11.16 [ 25.659 g .391e-03 m/N us: 25:23 mm 20.00 cm^2 391e-03 m/r 130.000 cm 32.900 l 10.000 7.00 QL 100 00 26.22 Hz 52.08 Hz 136 0 Schem 

\_ | X

Now go back to your original project and paste the items you have just copied. In order to get going you need to add a voltage source and a ground connector. With this you are done with the schema editing, it is now high time to leave the **Edit** mode and enter the **Simulate** mode. Click on the simulate tab. If you though that "Gosh-it-looks-ugly" you will hopefully find the looks better now. There are a few numbers here and there, if you don't like them you can actually hide them as don't have much use of them. Look for the settings menu pick in the main menu and

deselect the **Show node numbers** checkbox in the dialog that comes up, while we are here we can also set the display range to

Edit Simulate

9.000v

Passive Analog/digital Box/cabinet Other

□ + m 1, 3E M 8 3

Our best buy

ò

10-2000Hz. You can then close this dialog.

There are a few thick gray lines running across the schema. These lines only show that all components are referenced correctly to one another the latter is a key feature in LspCAD 6 and makes it possible to add semi intelligent wizards in an otherwise

Mmp

QI 100.00

391e-03 m/N

130,000 cm<sup>2</sup>

stupid node analysis algorithm. These gray lines can be hidden, click eg on the ABR component and deselect the **Show references** checkbox.

Loudspeaker unit PB

Configuration Parameters

T/S param Nonlinear model Other

Manufacturer Peerless

☐ Show parameters

v\lspcad6\examples\Peerless SWR308.unt ▼

The whole thing looks done, except for the fact that we need to enter the TS parameters,

click on the TS driver symbol (the large box with the text BI inside). In the dialog that pops up you can enter/modify a lot of parameters. First of all click on the **Parameters** tab. Here you should add all the parameters you have

If we go back to the T/S parameter list we see that Sd = 466cm<sup>2</sup>. Said and done we enter 466 in the Sd field, to do this we first double click on the Sd row, then we enter 466 and click outside the enter box. Before we go on and enter Vas we check the small checkbox that is immediately to the left of Sd, with this done Sd will stay fixed when we enter the rest of the parameters. When we have enterer Le we see that all necessary parameters of the parameters have been entered. The important parameters for the computations in LspCAD 6 are: Re, Le, (Reb, Leb), Rms, Mmd, Cms, (LambdaS), Sd and Bl. The parameters in parenthesis are not critical. Once we are done the list looks like the one to the left.

**Important** to know is that the TS parameter list must be saved separately, this is guite natural as one might need

80.00

75.00

70.00

65.00

60.00

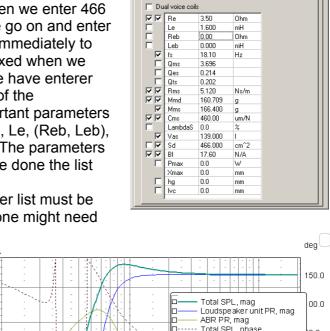
55.00

10 Hz

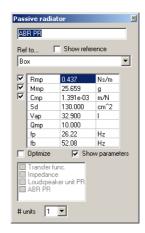
50

the same TS parameters for another project.

Now we should concentrate on the frequency response. If we look on the graph we realize that a cutoff of 60Hz is less impressive given that we deal with an expensive long stroke subwoofer.



Click on the passive radiator component. Here we should first set Sd to 466cm<sup>2</sup>. As we use a passive radiator that has the same cone area as the subwoofer. Here we actually should enter the values that we find in the table some pages ago but lets have some fun with the group properties first!.



Total SPL, phase

0.0

0.0

-100.0

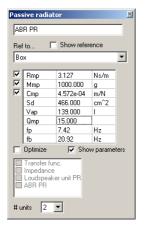
-150.0

+> Hz

Click inside the dotted box with the right mouse button. In the dialog box that pops up we find a **Wizard** button. Click on the wizard button and then on apply and Viola we have an optimally flat alignment for the subwoofer with a box resonance frequency of 37Hz. Close the wizard box.

Now click on Clone. Then change the volume to 40I. Now go to the dropdown list box and click on the arrow. Here you will find that there are two group entries, you can actually switch between the old entry (with 18I box volume) and the new entry with just a mouse click. Take the chance to look at the graphs when you select between the entries. Feel free to change the name Clone – Passive radiator box to something better...





Now it is high time to get back on track again. We open up the passive radiator dialog box and enter the values we found in the table. Note that there are actually two passive radiators. Enter the

values in the same order as they listed

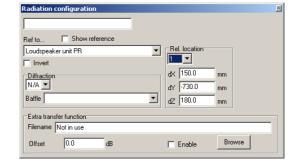
in the table.

We also need to set the boxvolume to the correct value.



We need to set the relative locations of the subwoofers and the passive radiators in order to get a reliable simulation. As we have the treble unit as reference point we need to set dX = 150mm, dY = -730 and dZ = 180mm. Click on the radiation component (looks like the letter R) that is closest to the TS driver and enter the information.

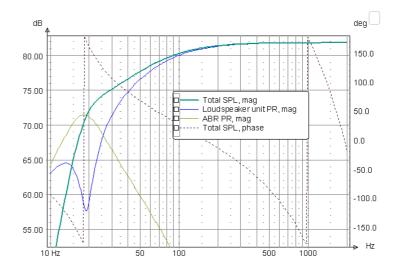
For the passive radiators we enter dX = -150, dY = -730 and dZ = 180 and



dX = 150, dY = -430 and dZ = 180 (click on the radiation component that is closest to the passive radiator component.

The graph begins to look like the simulation we see in the ugly duckling. One exception is however the lack of the bump at 100Hz. To model this we open up the TS driver dialog again and click on the **Configuration** tab. Here we find a box with the name **Inductance simulation**. This is **Off** by default but can be set to two active modes.

#### Load impedance only: In this mode we get almost the same response as in the older LspCAD 5.25 simulation.



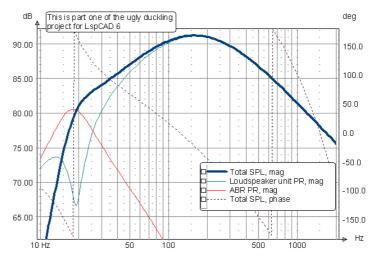
**SPL and load impedance**. This models the impact on the power response, actually it makes more sense to model the power response as the subwoofer is actually side mounted so we simply select this mode.

The colors look quite dull so we take chance to modify these (click on the tiny rectangular boxes in the graph). Also we click on the generator in the schema and set the voltage multiplier value to 2.83.

With this we are done with the subwoofer part for the ugly duckling. If you want to see an overview of the project, click on the small arrow down symbol on the lower left corner of the main window.

The project so far is saved as ugly duckling 1.lsp

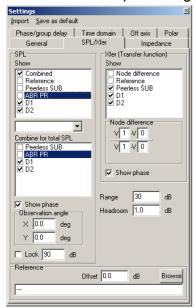
in the examples folder.



#### 4.2 The midrange and the treble unit

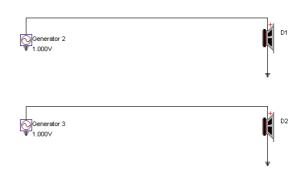
We continue using the same project and add the midrange and treble units. It is however a good idea to put these components on another schema page, therefore click on the small dropdown listbox that is located in the upper left corner of the schema.

First of all we put the driver units on the schema. This is quite straightforward.



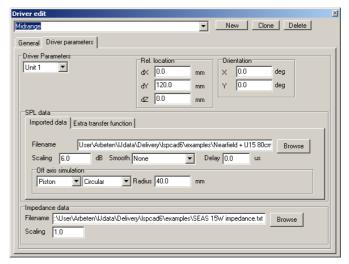
Next step is to import the SPL and impedance data for the midrange and the treble units. The midrange actually consists of two units connected in parallel. The locations of the midrange units are dX = 0, dY = 120 and dZ = 0 and dX = 0, dY = -120 and dZ = 0.

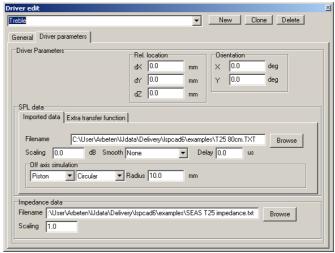
For the treble unit we set dX=dY=dZ= 0.



As we only wish to see

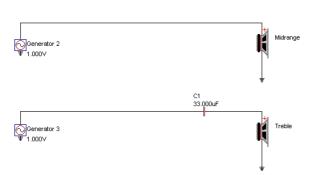
the midrange and treble units for the moment we open up the settings dialog and deselect the Peerless SUB, and the ABR PR from both the listboxes in the SPL field in the SPL/Xfer tab. We can also set the display range to 50Hz-20000Hz

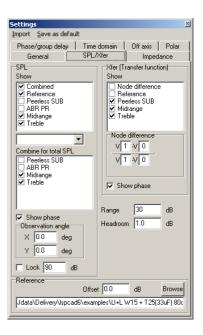




Before we do anything else we recall that we verified that the dZ offset was set correctly in the old ugly duckling. We can actually do the same thing in LspCAD 6. For this purpose we open up the settings dialog again and select the **SPL/Xf**er tab. In the bottom of this tab we have the option to import a reference curve. We import this reference curve and also in the **Show** box we check the reference curve. As this reference measurement was done at 80cm distance we also need to set the distance to 80cm in the **General** tab.

In order to protect the treble unit in this reference measurement a 33uF was put in series with the treble unit. Therefore we also need to break up the schema and do the same thing.





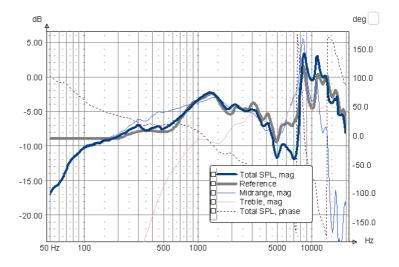
#### LspCAD 6 tutorial

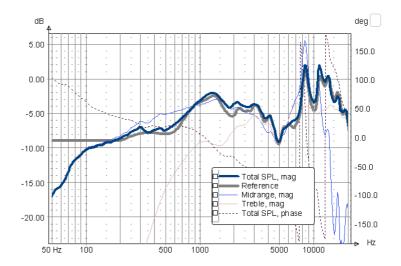
With this done we can have first look at the SPL graph. We can see that we need to adjust the offset of the reference up by 2dB to match the simulated response. Still there seems to be a misalignment between the treble and the midrange units. In the older ugly duckling project we adjusted dZ for the treble unit in order to get a good match and we also see here that dZ = 3mm we get the best match.

With these adjustments we can start doing the crossovers it is however advisable to set the simulation distance to 3m and deselect the show reference curve.

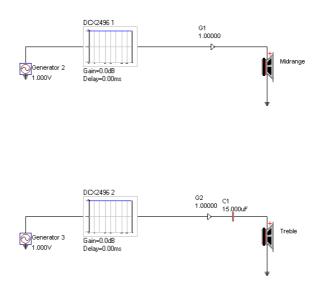
The project so far is stored as **ugly duckling 2.lsp** in the examples folder.

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The real ugly duckling of today use the Behringer DCX2496 digital crossover. Also a 15uF capacitor is in series with the treble unit to protect from possible power on transients. Modeling crossovers for the ugly duckling can be done in many ways. One alternative is to use the G(s) G(z) component but as the DCX2496 is actually used in this case we pick the DCX2496 components here. With the necessary components inserted into the schema we get the looks to the right. It is a good idea to waste space as it may be needed later. The observant eye may see the small buffer amplifiers, there are needed as the DCX components cannot directly drive the driver units, neither in reality nor in LspCAD 6.



For the midrange we need to determine what kind of filter blocks we need. After some thinking we need roughly:

2<sup>nd</sup> order HP filter 100Hz

3<sup>rd</sup> order LP filter 3000Hz

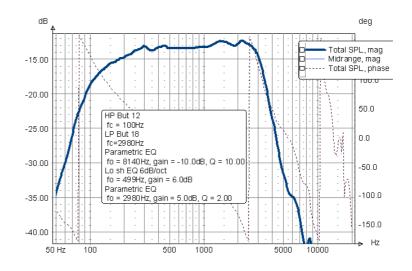
A shelving filter for the baffle step compensation

A notch filter for the 8200Hz resonance peak.

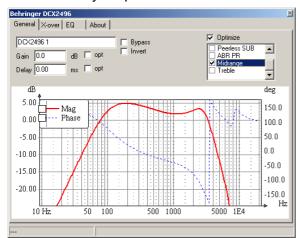
First we deselect Treble from the **Combine** and **Show** list box so we can concentrate only on the midrange.

The first, unoptimized attempt is shown to the left.

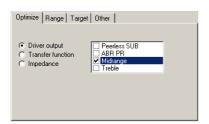
The project so far is stored as **ugly duckling 3.lsp** in the examples folder.



With this it is time to optimize the response of the midrange, as target we select 4<sup>th</sup> order Linkwitz-Riley response with 120Hz and 3500Hz.

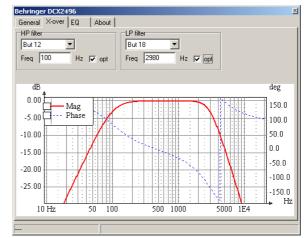


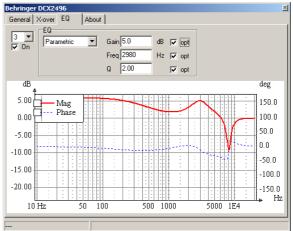
In the settings for the midrange filter we need to set the optimizer to optimize the parameters of the filter, (cutoff frequencies for the LP/HP filters, gain, Q and center frequencies for the EQ's). Also we need to enable the optimization. It is more convenient here to not include the 8100Hz EQ in the optimization.

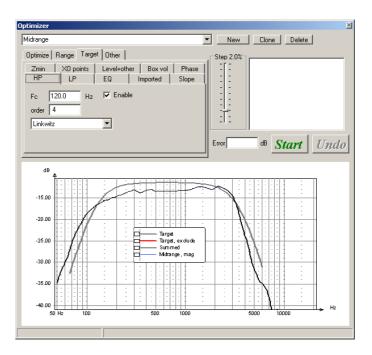


In the crossover optimizer we first select that we wish to optimize the midrange response, also we set the range to 70-6000Hz.

When one click on the Start button one may experience that the optimizer is slow. The reason to this may be that we have too many analysis points. A way to increase the speed is to set the working range to a more limited range and decrease the number if analysis points, another means is to select **the Fast Iteration** in the **Other** tab.

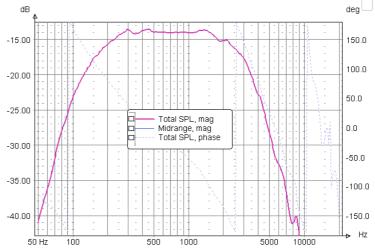






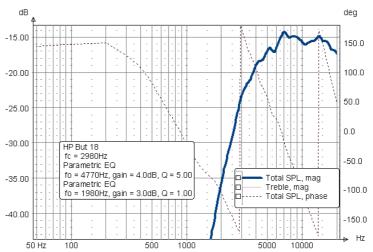
After the optimization we have a midrange response that follow the target pretty well.

The project so far is stored as **ugly duckling 4.lsp** in the examples folder.



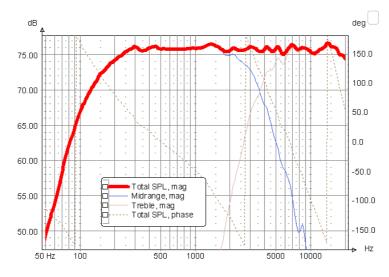
The treble filter is optimized in the same fashion as the midrange filter, the resulting treble response optimized for a Linkwitz Riley 4<sup>th</sup> order alignment is shown to the right.

See ugly duckling 5.lsp in the examples folder.



The combined midrange and treble response can be

found in **ugly duckling 6.lsp**. Looking closer at this response (not shown here) one can see that the response is not entirely ruler flat also there is a peak at 6500Hz, not large but it would be neat to get rid of this. Also it looks better it we add 90dB scaling to the imported SPL data. If we add a dip at 6500Hz and run an additional optimization we get the summed midrange and treble response below (see **ugly duckling 7.lsp**).



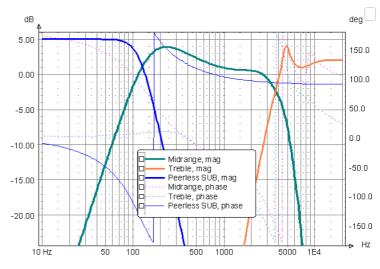
# 4.3 The subwoofer unit (again) and tying things together

After this is done it is high time to add the subwoofer, the signal to the sub needs to be LP filtered, for this purpose we use a Behringer crossover component as well. This is configured as a 2<sup>nd</sup> order butterworth filter with cutoff at 119Hz.

The resulting on axis frequency response is shown to the right. In this case no additional optimization is done (see **ugly duckling 8.lsp** for more graphs).

deg dΒ 150 O 75.00 100.0 70.00 50.0 65.00 0.0 Total SPL, mag Peerless SUB, mag 60.00 ABR PR, mag -50.0 Midrange, mag Treble, mag Total SPL, phase 55.00 -100 O 50.00 -150.0 10 Hz

Also worth a look is the transfer function. This will become handy when we wish to create a passive crossover later on.



The response in the low frequency region drops approximately 4dB per

octave. This is good when the loudspeaker is in a normal listening room, one can always EQ it with a shelving EQ and get the response simulated in **ugly duckling 9.lsp**, this does however sound "too much" in the very low frequency region and can sometimes become really unpleasant

### 4.4 Doing the passive crossover

Doing passive networks is a tedious task. One reason is that the load impedance is complex, this can however be fixed with shunt circuits such as zobel and series resonance circuits. The other problem is that unlike e.g the Behringer XO where one can set the slope of a LP filter and the shape of an EQ , everything in a passive XO is tied together. For instance if one change a component that is supposed to affect the slope of the highpass part of a bandpass filter, chances are pretty high that the lowpass part is affected as well. In short, what is easy done with active XO's can drive you crazy if implemented as a passive crossover.

But lets stop whining. This section described how a passive crossover is constructed for the midrange and treble section. The subwoofer is omitted, is anyway bad practice to construct a passive crossover for a subwoofer as the components can in the end be more expensive than buying a power amp and run the subwoofer actively.

First we open the project **ugly duckling 9.lsp**. To make things easier with the construction of the passive crossover we export the transfer function of the midrange and treble unit filter, this is done from the graph window. Click on **Export**. In the dialog we select



the Midrange transfer function and click on **Export**. We save the file as **ugly duckling midrange TF.txt** in the examples folder. Do the same with the treble unit transfer function.

This makes it possible to optimize the transfer function of the passive crossover to the same target, slightly more simple than having to optimize frequency response all over again, also it is used in this tutorial to show the possibility

One observation worth notice here is that the gain for the treble filter is 1.8dB, this means that we would need to attenuate the signal to the midrange in a passive filter. Personally I don't like this so we need to fix this.

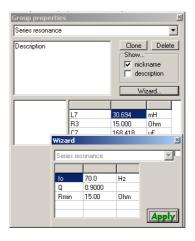
After this we remove the subwoofer and all the other components except the midrange and treble unit driver from the schema.

Our first task is to make the impedance curves of the driver units more flat, this is of great help in our strive to achieve a good passive filter.

In ugly duckling passive 1.lsp a series resonance circuit is used to reduce the

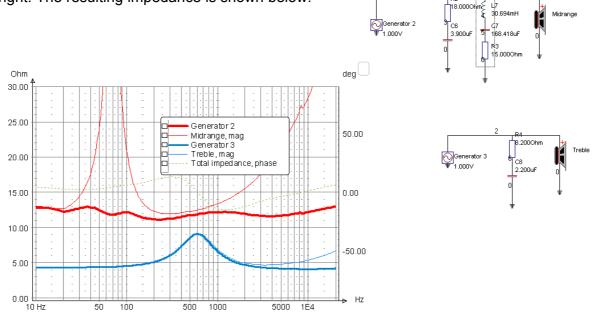
resonance peak at 70Hz while a zobel is used to cancel out the impact of voice coil inductance. Also in this case the midrange is change to a configuration with the two midrange units is series. This reduces the sensitivity of the midrange but makes it easier to match the sensitivity of the treble unit. For the treble unit we settle with a zobel network just to fix the voice coil inductance.

Note that the series resonance component is a grouped component with wizard properties. This means that one can right click inside the dotted frame and click on **Wizard** in the dialog that pops up. In this wizard one can set the center frequency (**fo**),the Q value and the minimum impedance of the series resonance circuit, when **Apply** is clicked the appropriate components are computed.



After some tweaking with components (actually it is possible to use the optimizer for this).

We get the initial passive crossover show to the right. The resulting impedance is shown below.



The impedance is not totally flat but it is not critical to get it 100% flat as we may anyway use these components when we optimize the transfer functions of the filters. The resonance peak at 600Hz may cause some problems for us when we start to tweak the transfer function of the treble unit but lets hope for the best...

The project so far is stored as **ugly duckling passive 1.lsp** in the examples folder.

Now it is time to add the actual crossover components. For this purpose we can add a HP/LP filter with a right button click on the mouse (in **Edit** mode of course).

As a starting point we select a 2<sup>nd</sup> order HP filter with cutoff at 120Hz and a 3<sup>rd</sup> order with cutoff at 3500Hz. The nominal load is set to 120hm. When we click on create a crossover is dropped on the schema, we move it to an appropriate place.

Perhaps not necessary but good anyway is to make sure that the very aggressive peak at

8500Hz is attenuated as much as possible. Therefore we put a series resonance filter as a shunt across the driver terminals (get the wizardized group from the templates this time also). When we enter **Simulate** mode we right click on the series resonance and configure it so that we get a sharp notch at 8500Hz.



8462.8

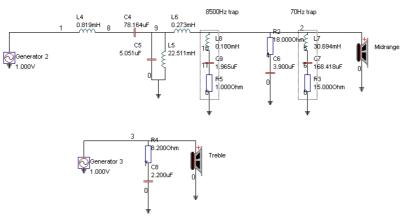
9.5713 1.00 Hz

Ohm

T.

Apply

The crossover begins to look like something useful.



The frequency response is less charming though. We could run hog wild trying to optimize the response but it would probably fail as we do not have anything that can compensate for the baffle step. The existing components may do this job to some extent but most likely it will not work 100%.

The project so far is stored as

# ugly duckling passive 2.lsp

in the examples folder.

deg 70.00 150.0 65.00 100.0 50.0 60.00 0.0 55.00 Total SPL, mag Midrange, mag Total SPL, phase 50.00 -100.0 45.00 -150.0 10 Hz

In order to get something that can handle the baffle step we add a parallel inductor and a resistor in series with the input to the crossover.

Now we tag all the components belonging to the midrange crossover (except the components that belong to the series resonance and zobel circuits) for optimization.

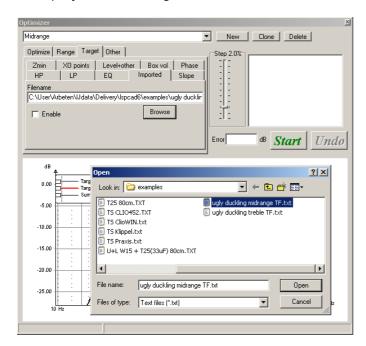
Earlier we mentioned that we should optimize the transfer function of the midrange crossover. The figure to the right we can see what it looks like for the component L4.

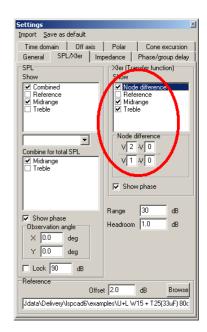


Next we need to measure the transfer functions of the midrange crossover if we look at the schema we see that the input to the filter is connected to node 1 and the midrange is connected to node 2. We open the **Settings** dialog and click on the **SPL/XFER** tab we can set the node difference to be computed as the voltage ratio between node 2 and node 1.

We open up the **Optimizer** and select Transfer function in the **Optimize** tab.

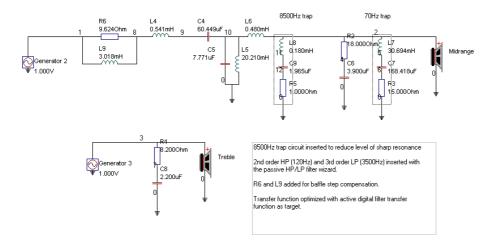
As target we select the file that we exported from our digital filter project. The Range is set to 70-6000Hz.





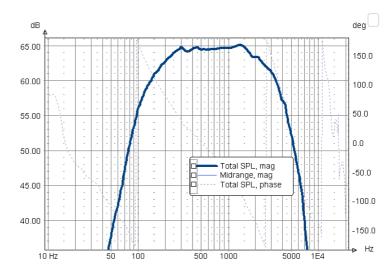
When the optimizer is open the components that are selected become **boldface**, it is good practice to have an extra look at the schema to verify that the right components are optimized before **Start** is clicked.

Once the optimization is done we have the midrange crossover and the response below.



This looks pretty good or ??

The project so far is stored as **ugly duckling passive 3.lsp** in the examples folder.



Optimize Range Target Other

✓ 3600.0Hz+/-5%,6.0+/-0.5dB

1000.0Hz+/-10%

 HP
 LP
 EQ
 Imported
 Slope

 Zmin
 X0 points
 Level+other
 Box vol
 Phase

Freq

6.0

3600.0 Hz+/- 5 %

+/- 0.5 dB

The treble unit is treated in a similar way as the midrange. Here a 2<sup>nd</sup> order HP filter is combined with an attenuator. One special problem with the treble unit is that a dip at 4700Hz needs to be equalized. This is a no-brainer with an active crossover but with a passive crossover some extra brain cells are needed. In this particular case a parallel resonance circuit was incorporated in the attenuator circuit that serves to reduce the attenuation at the resonance.

After some optimization a flat response was achieved one problem though was that the impedance was a mere 2.8ohm at 20kHz!.



This must be taken care of. Therefore we run an additional optimization run with impedance constraint. For this purpose we tag all the components that belong to the treble network for optimization (see example to the right). There is no need to touch the midrange components as we see that they don't cause any impedance problems.

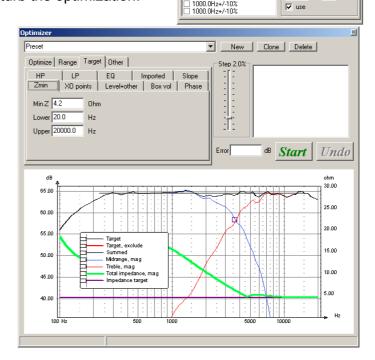
When we run our optimization of the combined midrange and treble response we set the range to 250-16000Hz as we don't want the roll off to disturb the optimization.

One special feature we use is that we lock the crossover point to 3600Hz, thus we ensure that this XO point wont move when we hit start.

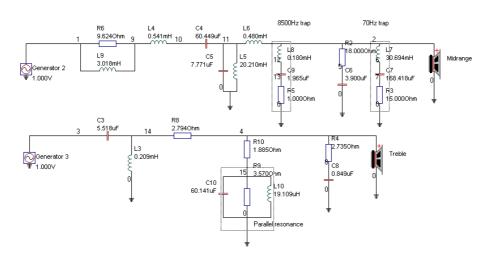
In the **Zmin** tab we set the initial **Min Z** value to 30hm. Then we click on **Start**.

As the optimization is running we click in the MinZ field and increase the value with the "Arrow up" key on the keyboard. Increase this value with care, if you increase to fast the optimization may FUBAR and you need to stop and undo the optimization, the restart again from a low **Min Z**. After a while you have reached 4.2ohm, which is considered good enough.

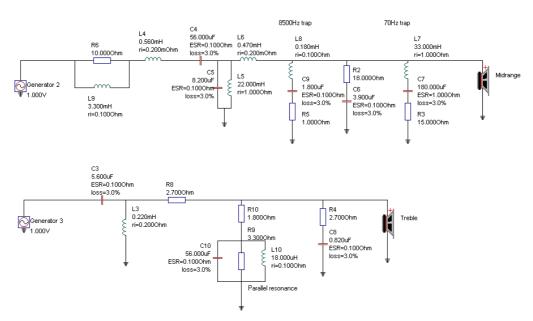
The project so far is stored as **ugly duckling passive 4.lsp** in the examples folder.



The final crossover schema looks like below.



The components are non-standard however. It is possible that we can special order the inductors for the resonance circuits but they are for sure not with zero internal resistance (unless you are extremely rich). Also the capacitors have some loss and ESR. So we add some internal resistance for the inductors and a little loss and ESR for the caps plus force most components to E24 standard values.



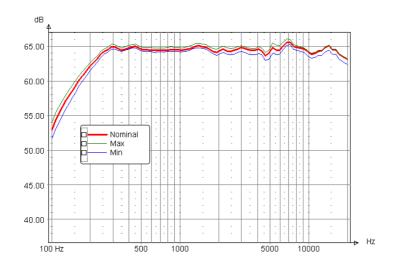
With some work plus rearrangement of components we get a schema like above, the losses and internal resistances are guesswork. The frequency response does not look that good anymore (**ugly duckling passive 5.lsp.** This can be fixed with another optimization run.

After the optimization the response is more flat (**ugly duckling passive 6.lsp**) but the components are non-standard again.

Again we can tweak the values to standard values but it is simply to boring for so I leave it as homework for you.

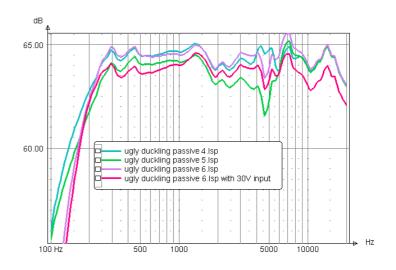
An interesting thing is tolerance to changes in component values. As an experiment all components are given a tolerance of +/-5%. The tolerance analysis feature is used to get an idea how sensitive the crossover is to component variations. The result of this experiment is shown below, there are no big surprises here. Good.!





What about power handling?, it is good if the overall frequency response is kept relatively flat when the input signal increased. For this experiment we assume 5K/W temperature coefficient for the midrange and 20K/W for the treble (this is also a wild guess). When we increase the input voltage from 1V to 30V we will get a slight compression of the output power, this is inevitable, what is more important though is that the frequency response remains relatively unaffected. The figure below displays the frequency response for different parts of the project, the 30V input graph is offset by –29.5dB to fit into the graph.

This project was intended to show some of the features available in LspCAD 6 and also give some ideas how the optimizer can be used.

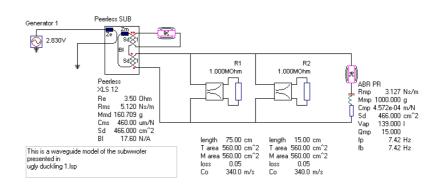


## 4.5 Modeling a column type loudspeaker

The subwoofers in the ugly ducklings are of the long, narrow type. This means that one will experience standing wave resonances at quite low frequencies. These standing wave resonances are not modeled with the lumped parameter model presented in chapter 4.1.

In order to enable modeling of the standing waves we can replace the box component with two wave guide components. The reason why we chose two wave guide components is that subwoofer is not mounted entirely in one end of the column box rather it is located 15cm above the bottom of the box which in this case is 90cm high. We should thus use two wave-guide components with the length 15 cm and 75cm. In order to get a volume of 51 l we need to set the throat and the mouth area to  $560 \text{cm}^2$ .  $V = (0.15+0.9)*(0.056) = 0.0504 \text{m}^3 = 50.4 \text{ l}$ .

After some editing we end up with the schema to the right. R1 and R2 are to simulate the closed ends of the two wave-guides.



Below is shown the difference between the lumped parameter model and the wave-guide model.

This example is stored as **column ABR speaker.Isp** in the examples folder.

