Designing Speakers

Part 10 – WD18BR – Crossover Development

Measuring loudspeaker drivers is the only way to develop a crossover. Peter Comeau shows you how.

L ast month we looked at the options available when designing a bass reflex speaker and this resulted in our first project, the WD18BR. Having investigated, with the help of LspCAD6 lite, the 'virtual' performance of enclosure concepts and suitable bass units, it is now time to do some 'real' measurements.

Again we will be using LspCAD6 Lite (you can download a free trial version from www.world-designs. co.uk) but you can use your own measurement software and still follow along.

We have planned and built a prototype cabinet to give us the 18 litres net volume that LspCAD predicted would work well with the SEAS H1215 or H1217 drivers. (SEAS drive units are available from www.seas.co.uk). Typically bass reflex enclosures give optimum performance with just an internal lining of cabinet damping. So for our first prototype I lined the internal walls with a layer of SONIQS CDF making sure that the route from bass unit to port is kept clear of obstructions.

As is the fashion when I design bass reflex enclosures I have put the port on the rear of the cabinet, and this is more than just a cosmetic fad! An extra hole in the front baffle just weakens its rigidity, as well as looking, to my eyes, unsightly, so the rear baffle is a better place for it. In addition, hiding the port on the rear also hides it acoustically. There is a fair amount of midrange output and mid bass harmonic distortion produced by a port and I'm quite happy to 'lose' this by letting the speaker box 'hide' it from the direct output of the speaker.

Offsetting this is a possible lack of integration between the port output and the bass unit. You may see this mentioned in some treatises on bass reflex design – that the port should be positioned close to the bass unit for good mutual coupling. Actually I've only found this to be true where the system is ported at a relatively high frequency, say 80Hz or more (e.g. a really small box). At the 36Hz frequency that WD18BR is tuned to, the path length difference between the port and bass unit is short compared to the wavelength, and the port area is too small to enjoy mutual coupling from the bass unit, so it really doesn't have an audible effect.

If you want to 'show off' your port then by all means put it on the front baffle – there is room under the bass unit. Again, for ultimate efficiency it is often advocated that you put the port at an antinode where the air excitation inside the enclosure is the highest. However as this also amplifies the standing wave in the enclosure I would advocate placing the port away from the antinode, say at 25% of the cabinet height. You should certainly avoid putting the port near the cabinet walls where energy levels are low and 'muffled' by the internal damping material.

At this point I won't be fitting the final port as I want to tune the exact length by ear. LspCAD will certainly give me an 'optimum' value, but I may find that detuning the system resonance a little may enhance the subjective performance – more on that later. So I have cut a 40mm diameter hole in the rear baffle that is just the right size to accept the 1.5" dia PVC drain pipe readily available from home DIY centres. I can then try out different lengths of pipe quite easily and cheaply.

Positioning the drive units is partly common sense and partly an art. The common sense bit is not to put the bass unit half way up the cabinet, where it will reinforce the primary vertical internal standing wave, and not to position the treble unit equidistant from the top and sides of the cabinet. Reflections from each cabinet edge will cause an interference ripple on the treble unit output so having an equal distance from each edge is just asking for trouble.

If you are making your own cabinet you can even decide to offset the treble units, and thus produce left and right 'handed' cabinets, to reduce

this edge reflection effect even further.

ART AND DESIGN

The 'art' part comes in how you position the treble and bass units relative to each other. Now the common conclusion is that you should put them as close as possible so that they 'integrate' better at the crossover frequency and so that the path length difference between them is minimised. However, if you do this, the treble unit 'sees' a shelf, which is the hollow of the cone of the bass unit, immediately below it. This degrades the treble output signifi-cantly and may be audible.

As this design uses a relatively small bass unit of 18cm, I feel I can provide a substantial baffle area between the two drive units. Where the bass unit is much larger, such as in the WD25A design, moving the treble unit away that extra distance does become problematic. We'll see the effects of the increased baffle area around the treble unit when we measure its response.

So let's get to it. If you missed my earlier article on speaker measurement I'll repeat again that you can easily measure a speaker in your living room by using an MLS 'gated' measurement system and positioning lots of soft furnishing between the speaker and microphone to help absorb the floor reflection.

On that basis I'm using a sofa with cushions and a duvet on top of it between my microphone and speaker. LspCAD6 Lite includes 'justMLS' – a neat program, found on the Tools menu, which allows you to make gated measurements using MLS quite easily and export the data into LspCAD's crossover design section.

JUSTMLS

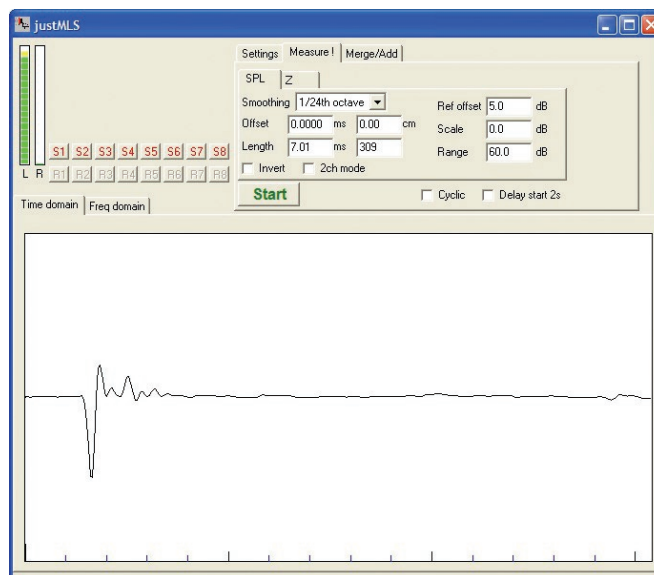The MLS signal is a pulse that contains all the frequencies we want

Fig. 2 - You can see the pulse in the Time domain window. I have selected the window to a width of just 7mS to 'gate out' the major room reflections (you can just see a minor reflection appearing at the right of the window).

to measure and, because it is a short pulse, we can select just the output of the speaker and avoid room reflections.

You can see the results in Figs.2 – 5. The smoothness of the treble unit response is as much due to the baffle area around the driver as it is to this unit's exemplary performance. Similarly the smooth output of the bass driver is enhanced by the wider than usual (compared to modern commercial slimline speakers) baffle. All this is going to make our job easier for good acoustic and crossover design.

Now we can check out the prediction for the port tuning. We could do this by putting the microphone close to

Fig. 3 - When we switch to Frequency domain you can see the result for the H1217 bass unit. I have applied 1/24th octave smoothing to get rid of the 'noise' and we can quite clearly see the overall response together with the baffle shelf effects due to width (575Hz) and height (300Hz).
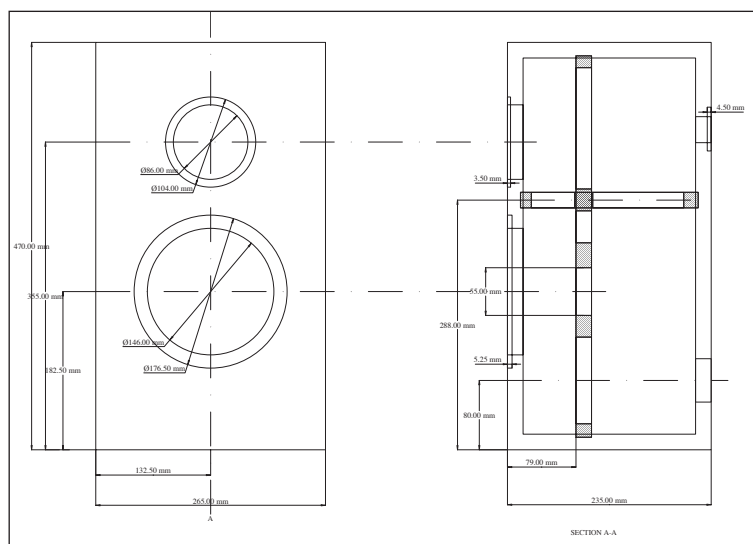
the port and measuring the output. But it is easier, and more accurate, to check the tuning using the impedance measurement of the bass unit. This is because the bass unit motion reaches a minimum at the point at which the enclosure and port are doing all the work. So we can easily see this minimum motion point as the bottom of a 'valley' that intersects the rise in impedance of the bass unit as it approaches the system resonance.

Turning to the impedance graph measured in LspCAD6 Lite (Fig. 7) we can place the cursor over the bottom of the 'valley' and, yes, it shows 36.8Hz with the port length we calculated – nearly bang on the target 36Hz that LspCAD calculated in our theoretical design exercise.

Fig. 1 - WB18BR Woodwork

P80-P85§§.indd   81   19/7/07   16:14:41
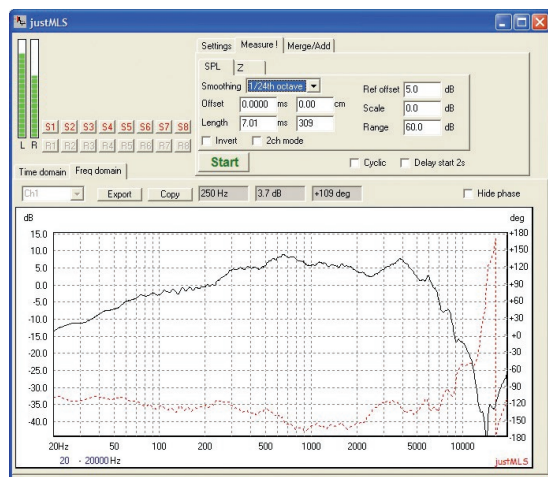
Fig. 4 - Putting drive units in a cabinet usually results in quite a change to the manufacturer's measured response (which is taken on a very wide baffle). If we look at the trace for H1215 we can see that it is much less smooth than H1217, particularly with the peak at 800Hz and the dip at 2.5kHz, both of which will give us problems with crossover design.
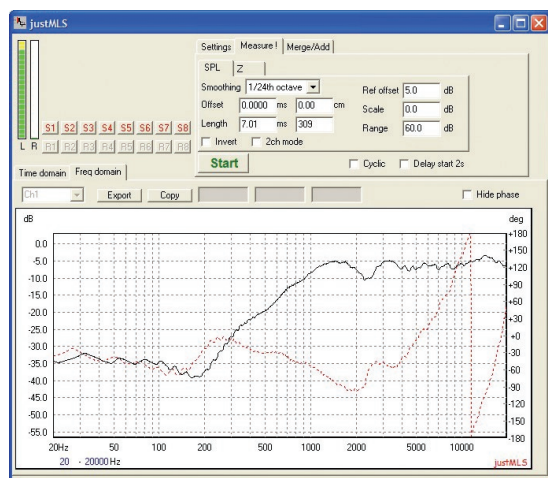

Fig. 5 - Switching to the treble unit indicates that surrounding the treble unit with a smooth baffle area produces a smooth response right down to 3kHz. This is probably going to be our optimum crossover frequency.
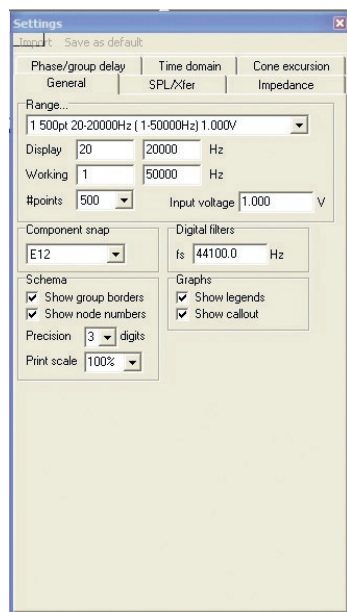

Fig. 6 - Settings

CROSSOVER

Having taken our measurements in justMLS we can hit the Export button to turn the graphs into text files that we can use in LspCAD's virtual crossover designer. I'm going to start off with a standard 2nd order crossover and progress, if need be, from there.

It's up to you whether you want to use first order, 2nd order or 3rd order as a starting point. It is just that I find 2nd order more rewarding, sonically, to work with and I can easily introduce variations on the 2nd order theme without too much complexity.

As soon as you've added the circuit components and imported the measurement files for each drive unit (see panel) you will have to start juggling component values. LspCAD6 lite shows you the changes you make on the SPL graph instantaneously. Initially things will look disappointingly non-linear, with peaks and dips where the drive units overlap that may look insurmountable. My advice is not to be too concerned when you start. The graph always looks terrible in the beginning but it is surprising how easy things are to align with a bit of analysis.

To start with go into the Settings menu and adjust the Component Snap to E12 (Fig. 6). This means that LspCAD will only allow you to use component values that fit the E12 convention, for example 2.2, 3.3, 4.7, 5.6, 6.8 uF and so forth. When you are fine tuning you may like to devise intermediate values but initially it is useful to have these jumps in value.

Now you can select any component and use the up and down arrow buttons on your keyboard to increase and decrease the value. Begin with the series capacitor to the treble unit and parallel coil; 6.8uF and 0.3mH are sensible starting values. Ignore the combined graph at this stage and just look at the treble crossover.

Adjust the values until the crossover looks smooth and not 'peaky' at the point at which it begins to roll-off. Don't be disappointed that it doesn't look anything like those nice smooth graphs shown in the theory text books. Real speakers don't behave anything like

that, at least not in the initial stages of design!

The reason we don't want a 'peak' or sharp corner at the roll-off point is because this generally indicates a 'ringing' filter which will sound bright and edgy. If you find the upper treble output is too high increase the resistor directly before the treble unit to level it out.

Having achieved a treble unit response that looks something like a respectable roll-off you can turn your attention to the bass crossover. You should start with the series coil value, increasing it until the midrange starts to level out. Then you need to juggle the value of the capacitor and resistor in parallel with the bass unit to achieve a smooth roll-off, again without being peaky in nature.

At this point you will probably find that you have a dip in the output between the units. Don't worry about it. Now is not the time to try and hurry towards 'perfection'. Just console yourself with the fact that you can do all this juggling of component values in the comfort of your own armchair and not be stuck in an anechoic chamber fiddling with a soldering iron and a bucket full of components!

OPTIMISER

One way of achieving a 'flat' response is to use LspCAD6 lite's Optimiser from the Tools menu. Here you can set the Target level you want to reach, the Range of frequencies you want to optimise and the components to include in the optimisation process.
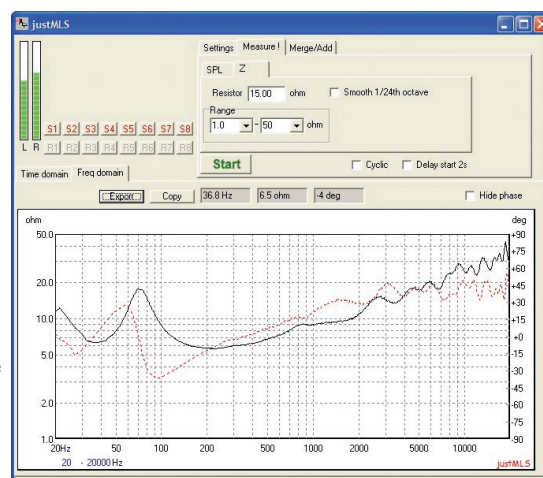
Set the Range to encompass the

Fig. 7 - When we run the impedance measurements we get another view of the drive unit's performance in the cabinet, particularly the bass unit. The two humps at the left of the window shows the system resonance as a 'valley' in between. Centre the cursor at the bottom of this 'valley' and you'll see the port tuning is 36.8Hz, very close to the 36Hz LspCAD predicted in our theoretical design!

crossover region and a couple of octaves either side of it. Don't try to incorporate the upper treble and bass regions as well or the excessive peaks and dips that exist in these regions will give the Optimiser (Fig. 8) an impossible task. Adjusting the crossover to marry with the bass region is best done by ear rather than relying on inaccurate measurement, as accuracy at LF is only possible in very large anechoic chambers, whilst treble above 10kHz is reaching the limit of audibility and is best left to its own devices.

If using the Optimiser leads you to a completely different component value solution than the one you had come up with manually then that will give you an alternative to listen to. But don't discard the one you arrived

Fig. 8 - The Optimiser window

at yourself – there is more than one way to design a crossover!

Another way to use the Optimiser is my favourite and that is to use it to align to Target crossover slopes. The Flat Target method can yield crossovers which are quite peaky in their response, pushing the drivers to their limits in order to generate a 'flat' response. Far better, in my opinion, to align the crossover slopes to a precise target. I have found that this gives me the best starting point when beginning to fine tune the crossovers through listening tests.

LspCAD6 lite can handle this happily, allowing you to choose classic alignments like Butterworth or Linkwitz and decide on the slope 'order' e.g. 1st, 2nd, 3rd up to 7th. These are the final acoustic slopes and do not have to align to the electrical 'order' as devised in your circuit schematic.

For example I find it useful to add the natural roll-off of the midrange response of the bass unit to a 2nd order electrical crossover to give an overall 3rd order acoustic slope. This might sound a bit odd but go ahead and try it out for yourself

– you will see what I mean if you experiment with using the Optimiser to target 2nd order and 3rd order acoustic crossovers.

Note that the Optimiser will be troubled by peaks and dips in the drive unit response close to the crossover point. If this is the case it is simple enough to fine tune the crossover component values manually using the Target roll-off as a visual guide.

FINE TUNING

Now it is time to fine tune the crossover by switching off the Component Snap to No Snap. This allows you to select intermediate values which will help iron out any remaining wrinkles i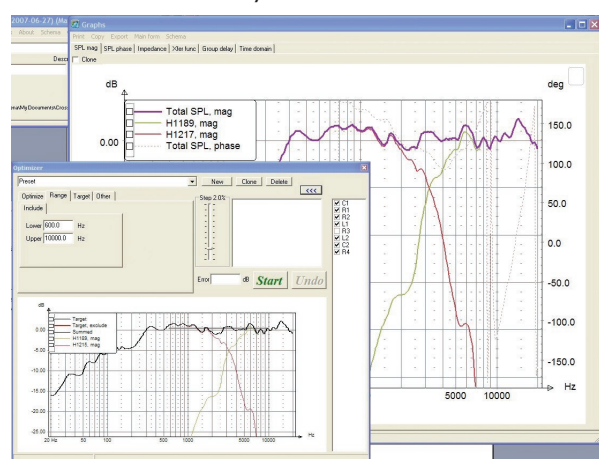n the response. You should be able to select or unwind coils to any value you come up with, while capacitor values can be achieved by paralleling standard value caps, e.g. 2.2 + 1.5 = 3.7uF.

Don't be too finicky about exact values. It is quite difficult to hear the difference between, say, 4.7uF and 5uF and note that these values are close to 5% tolerance anyway (the tolerance that most good quality crossover components are manufactured within).

As you get closer to your ideal 'virtual' crossover it is worth checking both the impedance and phase response. Impedance is obviously important as there is absolutely no point in designing a speaker with a ruler flat frequency response that dips to 2 Ohms. And while there is no concrete evidence that a 'linear phase' speaker sounds 'better' than one which has a shift in phase over the majority of its bandwidth, abrupt changes in both electrical and acoustic phase are to be avoided.

Sharp peaks and dips in the electrical phase response, which you can see overlaid on the impedance graph, are likely to give amplifiers a hard time. It is usually quite difficult to aim for a flat impedance as the natural rise through the crossover region contributed by each drive unit is partly inevitable. I have found that including multiple elements in the crossover to try and flatten

impedance generally just robs the amplifier and drive units of dynamic power rather than helping the performance, so don't worry about a 'rolling hills and vales' look to your impedance trace.

But more exciting is the ability to check drive unit integration by looking at the response with the drive units out of phase. Common sense tells us that phase integration between the drive units will be 'perfect' if the output at crossover is nulled when one of the drive units is inverted in phase.

You can do this easily in LspCAD6 lite. Click on the driver in the schematic to bring up the Driver Edit window, then tick the invert box on the General tab (Fig.11). If you have good phase integration then you will see a deep notch at the crossover frequency. If you have used the Optimiser Target Linkwitz slopes then you will probably find this works first time. Otherwise a bit of manual massaging of component values should get you there.

Again the proof of the pudding is in the eating. Aiming for 'perfect' phase integration is a nicety and does not necessarily mean that the speaker will sound wonderful as soon as you plug your crossover into the real thing. But at least try and get a semblance of a notch occurring.

So here is the crossover I came up with after some 'virtual' fine tuning (Fig. 12). At this stage it isn't ruler flat but it does obey the basics namely:
- The crossover region is slightly depressed, not elevated, from the average level. This helps 'hide' it from the ear.
- There are no big peaks and dips in the overall response.
- The overall trend is fairly level (and therefore neutral in tonal balance).
- Phase integration is good.

Now, at this stage, I could go on to add further 'control' elements to my crossover to even out the lumps. For example I could add a further capacitor to the treble to turn it into an electrical 3rd order to smooth out the little hump at 3kHz.

But you could spend all your time working in LspCAD's 'virtual' environment and never get your speaker built. I find it is better to build the first crossover and find out what's wrong with it (and what's right with it) and then go back and refine it in LspCAD after the initial listening test.

This process of iteration is critical to fine tuning the crossover. You don't have to get it right at your first attempt. Spend some time living with the suggested crossover

CROSSOVER SCHEMATIC

When we have finished the measurements it is time to import them into LspCAD's crossover simulator. But first we have to build a basic crossover circuit.
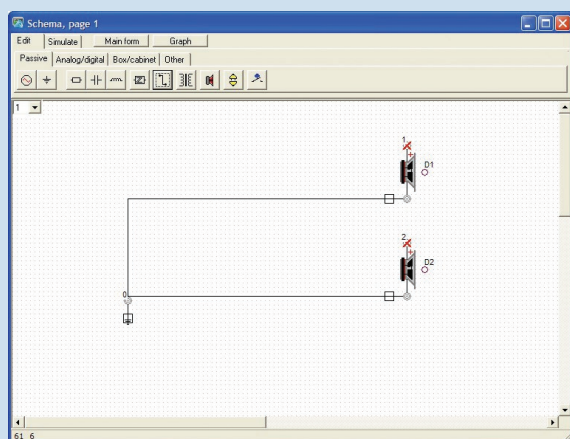

Fig. 9 - Adding components to the circuit (Schema window) is simply a matter of clicking on the component button and clicking in the Schema window. Right click on a component and choose Rotate to rotate it 90 degrees. Start by adding the two drive units and connecting their negative terminals to the source ground.
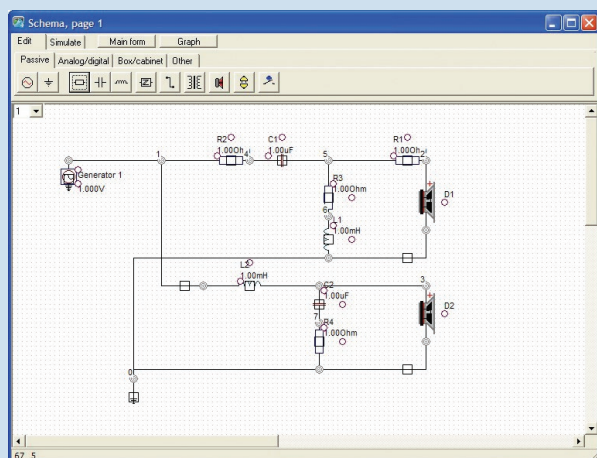

Fig. 10 - We'll need some power for the circuit so add a Generator then link in the components of a basic crossover circuit. Here I have added the components for a simple 2nd order circuit. Note that each coil will need a series resistance equivalent to its DCR. For ferrite coils you could leave this out as it is relatively small, but air core coils should definitely have it included. Components are always inserted with a value of '1.0' – you can adjust the values when you click the Simulate tab.
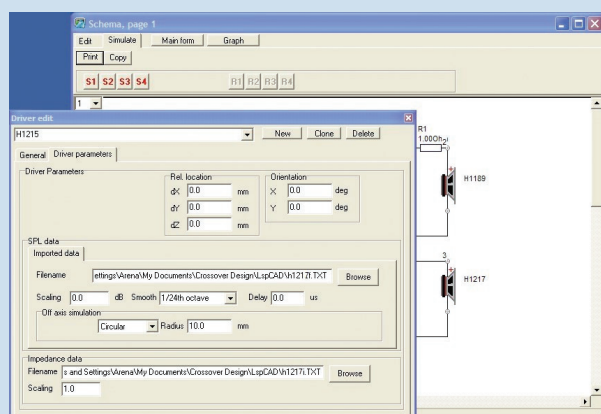

Fig. 11 - Click on a driver to bring up the Driver Edit window. Here you can type in a name for the drive unit and import the SPL data you exported from justMLS.

and trying out a variety of music. When you pick up on something that sounds 'wrong' try and identify whether the problem lies with the bass unit or treble unit (if you have biwired the crossovers then it is easy to disconnect treble or bass to find this out). Then make some changes in LspCAD to 'see' the effect and reflect those in your real crossover to 'hear' the difference.

One thing you will learn by doing this is that small changes to the treble circuit can apparently affect the bass performance, and vice versa. This might seem like 'magic' and part of the 'black art' of speaker design, but it isn't really. This is a psychoacoustic effect caused by the change in balance when you shift the crossover. For example tilting the treble upwards by 0.5dB will make the bass sound leaner and tauter. Similarly reducing the midrange level by 0.5dB can make the treble sound more explicit and detailed.

Clearly there is a lot of experimenting to do here. Give a thought to what it was like before we had tools like LspCAD. My first pair of speakers took 18 months to design by trial and error with occasional visits to an anechoic chamber! With LspCAD6 lite I can now do the same job in a matter of weeks. This is just one of the wonders that the advanced calculating performance of the modern PC brings us. Now you can have that advantage to help you with your speaker design too.

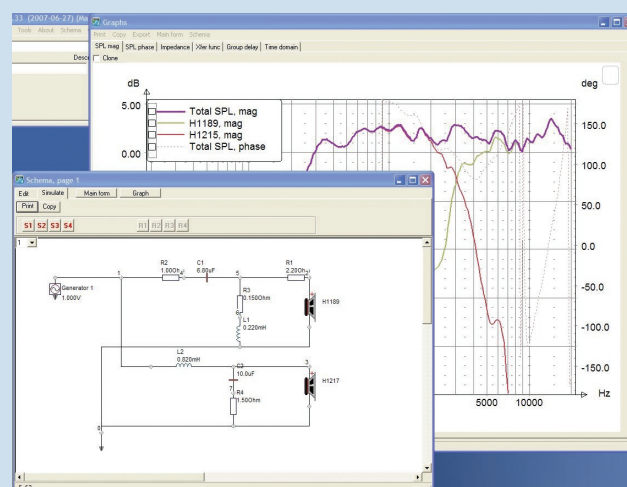**Next month: The floorstanding version and our next project begins**

Fig. 12 - Now we can start adjusting the values in our virtual crossover and see the effects on the SPL graph. To make adjustment easier change the Settings menu Component Snap to E12 so you can use the up and down buttons on your keypad to change values in standard steps.