Subject: Wayne, a question about B&C DE 250
Posted by Russellc on Sun, 26 Jul 2009 19:42:25 GMT
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Wayne, over on the Econowave thread a few of us have began using the B&C DE 250. In measuring its response, the measurers (unfortunately I'm not one yet, soon I hope to obtain measurment ability) are finding a peak around 3-4 khz. Can you shed some light here, and how its gotten around in your design. The two systems crossovers are similar in compensation, but different in 2nd order vs 3rd order. Just wondering about all this.

Another question I've had about the PI4 crossover with DE 250 is where the crossover points for HF and LF are Electrically, I understand they are around 1600 accoustically, just wondered if they were symetrical or "spread" to lessen any such response "humps". As usual, thank for the time and insight.

Russellc

Subject: Constant directivity, compression drivers and crossovers
Posted by Wayne Parham on Mon, 27 Jul 2009 02:54:48 GMT
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There are a few things that may conspire to create a ~3kHz peak in a CD horn with a 1" exit driver. One is that's just below where compression driver mass rolloff starts. Another is the possibility of a pipe mode, some horns have one at the bottom of their passband and sometimes another about an octave up. A third reason is the horn may be gaining directivity control in one or both axes, which causes an increase in on-axis amplitude (and anywhere within the pattern). In fact, most CD horns have some or all of these attributes, causing them to look like they have a peak in the first octave, with drooping response above that. It's so common in horns like these that 10 or 12 years ago, my Mantra about CD horns was "peak at cutoff, with a negative slope" to describe their response curve on various internet discussion boards. They all seem to have an initial pipe mode, followed by mass rolloff, creating a diagonal line of falling response.

About the crossover, I'm not sure if you remember my Crossover Document, but in that paper I examine and describe first, second and third-order filters with various compensation networks. What I found is that second and third-order filters act pretty similarly in terms of how the R1/R2/C1 network can be used to provide specific damping to make the appropriate transfer function, conjugating power response.

I developed that network specifically for CD horns, and it is designed not only to provide SPL matching and top-octave compensation, but also to provide specific damping of the core splitter high-pass filter in order to set the level of the region below mass rolloff as well. In essence, you balance R1/R2 to cause the reactive components in the core splitter filter to double as a peaking coil or notch filter, whichever is needed.

Since R1/R2 sets the damping of the core splitter filter, you can set whatever amount of peaking you want. It can be set for slightly overdamped or slightly underdamped, whatever is needed. That was always the whole intent of the R1/R2/C1 network - to provide these three things: 1. SPL matching between woofer and tweeter, 2. Top octave conjugate compensation for mass rolloff and 3. Specific damping for the core splitter to set the response in the region below mass rolloff, basically the octave between crossover and mass rolloff. Usually, the circuit is slightly underdamped, but if there is a pipe mode and/or diaphragm resonance in this region then it may need to be set for more damping.

About the slope of the core splitter section of the tweeter circuit, the biggest difference in second-order and third-order is the width of the overlap band and the phase relationship with the adjacent driver. The R1/R2/C1 network interacts the same with second-order filters as it does with third-order (and higher) slopes, so you can use this topology on any of them.

The choice of core splitter slopes and frequencies for the woofer and tweeter circuits should be determined by finding the ones that put the vertical nulls in the right place. That's largely dependent on the physical relationship between drivers and their characteristics. You should choose second, third or even fourth order filters and their frequencies solely based on performance in the vertical plane.
Crossover optimization for DI-matched two-way speakers

Subject: Re: Constant directivity, compression drivers and crossovers
Posted by Russellc on Mon, 27 Jul 2009 13:32:41 GMT
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Yes, I just got through reading a bunch of the posts you made on the diyaudio forum horns vs waveguides and answered a bunch of my own questions, many in the link you provided there, as well as here. Thanks again.

Russellc

Subject: Re: Constant directivity, compression drivers and crossovers
Posted by Wayne Parham on Mon, 27 Jul 2009 14:54:09 GMT
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I've made a few comments on other audio discussion boards but I keep the real meat and potatoes here. Most of what I say on other audio sites is just a regurgitation of things I've said here for years.

On the thread you're talking about on DIYaudio, I've tried to be a voice of reason with respect to performance in the vertical plane. There are some there that have made a point that verticals don't matter much and I think that is a sophomoric position, one that shows a lack of loudspeaker design maturity. You can use the best horn in the world and if it isn't implemented properly in a loudspeaker system, performance and sound quality won't be as good as it could be.

To me, if you're going to ignore the pattern in the vertical, you're not that much further along than

those that simply optimize on-axis response and call it quits. If you're using a CD horn and choose an appropriate crossover frequency, then the horizontal pattern approximately equals the on-axis pattern. That's the easy part. What's hard is to keep the on-axis response good over a useably tall vertical range, and then to drop off at large vertical angles. This is the most desirable coverage pattern - Uniform in both the horizontal and vertical planes.
Matching directivity in the vertical and the horizontal planesIt's not enough for the horn to to that, in fact, when combined on a baffle with another driver it doesn't even make sense to talk about the horn in isolation. Once two sound sources are stacked vertically on a baffle, their interaction sets the pattern and the response on-axis as well as off-axis is modified as a result. The shape, size and quality of the forwards lobe is set by the interaction of the sound sources, which is largely determined by their position, (crossover) phase and orientation.

When a constant directivity loudspeaker design is fully optimized you should be able to enjoy the same quality sound sitting in front of the speaker at various heights and not have to be right in front of the tweeter or any other pinpoint spot. That's the whole purpose of constant directivity, to provide uniform coverage over a range of listening positions.

Subject: Re: Constant directivity, compression drivers and crossovers
Posted by Russellc on Sun, 02 Aug 2009 17:42:58 GMT
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Wayne, all of this is very informative and I've been studying the links to the various documents you've included. In terms of the "core splitter" and setting the dampning and so forth, I have a question. Looking at the differences in the 3 PI filter (in some of the links you included) and the 4 PI filter, specifically the resistor (R2) that sets this "dampning" is different in the two, while the other compensation components are the same.

I understand that the 3 PI is a 12 inch model and the 4 PI is a 15 inch model. Does the difference in the size of the two drivers, and there differing responses at the crossover point because of the size have anything to do with the value of R2? I wish I could put this question better, but I think the answer to this (what sets the size of the R2 resistor, or differently. what difference to a given response will raising/lowering R2 have?) will help me see this a little more clearly.
I am trying to down load the video!

"In essence, you balance R1/R2 to cause the reactive components in the core splitter filter to double as a peaking coil or notch filter, whichever is needed". I think this is the portion I am refering to, Wayne perhaps I just dont understand about over dampened and under dampened and what effect each would have on response.
russellc

Subject: Re: Constant directivity, compression drivers and crossovers
Posted by Wayne Parham on Sun, 02 Aug 2009 20:08:21 GMT
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The R1/R2/C1 network is something I came up with a long time ago, and it is designed to produce an initial shelf of flat response followed by a second region where network output increases 6dB/octave to conjugate the mass rolloff slope of the driver (as shown below). This is what I always referred to as "top-octave compensation." I think it's the best approach, better than using peaking coils and/or notch filters.

You're right that R2 is the main resistor setting the load on the core splitter filter. That's what does the biggest part in setting the Q. But R1 and the driver are in parallel with R2, so they have an influence. It's probably best to think in terms of the R1/R2 values as a pair.

When I first came up with this network topology for CD horns, I was using Spice models to come up with the right transfer function. I initially calculated the positions of the lobes and nulls using crossover phase, baffle position and depth. I described the lobes as "windows", with the nulls being cancellation notches from path length differences. The forward lobe is what I described as the "most desirable window", with the nulls being areas that should be designed to be off-axis, outside the tweeter pattern where possible.

My calculations were useful, but all I could really use was driver position and crossover phase. It was an approximation. In hindsight, after using measurement equipment, I did a very good job but was lucky that the drivers I was using didn't have too many "warts" that shifted things, because they can.

I also found with measurements that the transfer function could sometimes be tweaked a little bit depending on driver and horn. Some horns have more quarter-wave mode peaking down low than others. Those will generally need more damping than horns that are smoother down low. A lot comes in to play at the bottom end, from diaphragm resonance to acoustic loading (resistance verses reactance and quarter-wave modes) to directivity (ripples as the horn gains control).

As far as the midwoofer is concerned, there's a lot going on there too. In a matched-directivity two-way speaker, you're pushing a large cone pretty high. It will tend to start flexing up high, and that can make a difference both on-axis and off-axis. Also, if the crossover slope (electrical and acoustic) doesn't create a pretty well-defined stop band, the midwoofer may be making sound high enough that its side lobes may narrow into the pattern. So there are a lot of variables that determine the response and directivity of the midwoofer besides the textbook behaviour of collapsing DI of a round radiator.

There are a lot of moving parameters in a speaker like this, particularly in the critical crossover region. The woofer is getting close to breakup and its off-axis self-interference angle is growing smaller. Breakup is a function of cone flex and damping, and self-interference is a function of collapsing directivity, of frequency verses diameter. The tweeter horn is just starting to gain directivity control, and if asymmetrical (as it should be) then directivity is different on each axis.

resistance verses reactance, and at the transition region, you'll likely find impedance ripple.

As you can see, there is a lot to try to include in a mathematical model for simulation. It's much better to start off with something you know is close and then fine tune using measurements. That includes not only the transfer function set by R1/R2 damping but also the position of the lobes and nulls, by setting the core splitter filter (electrical) frequency and slope. It all comes into play, interacting with each other in a (hopefully balanced) set of competing priorities.

In the end, I think you can do a lot with simulations, manual calculations and other forms of mathematical models. I used them for years with good results, even without the benefit of good measurement equipment. But the crossover region is a tricky one in a speaker like this, because there are a lot of competing priorities and a lot of "variable drift", meaning that things are moving and changing.