Subject: Crossover optimization for DI-matched two-way speakers, revisited
Posted by Wayne Parham on Fri, 02 Jul 2010 20:02:42 GMT
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I usually fine-tune the position of the forward lobe by adjusting the low-pass circuit slightly. I know within about 10° where the centerline of the forward lobe will be before I start, because of mathematical models or previous design efforts or both. Then measurements guide me the rest of the way, to perfect the loudspeaker and set the nulls exactly where I want them to be, outside the pattern. The process is described in the link below:
Crossover optimization for DI-matched two-way speakersWithout measurements, if it sounds good to you, probably you should leave it well enough alone. You could try and model the system, and give it your best effort that way. For two decades, that's how I did it. I calculated the electrical phase at several frequencies through the crossover region starting below and ending above. I also calculated the delay from baffle spacing. I could then calculate where the nulls were, and set the crossover (phase) accordingly. At first, these were all hand calculations, large charts of impedance, phase and delay. Later, I started using Spice to get the transfer function. But I still modeled the acoustic interaction with hand calculations. Now days, you can use programs like LspCAD to make this process a lot easier.

But even the best models are trumped by actual measurements. I have found that it's really hard to get the forward lobe dialed in within 10° of accuracy without measurements. You can get it within about +/-10° if you're really, really diligent with your mathematical models. If you've been meticulous with your calculations/models, then what your simulations will show as the center axis of the forward lobe may be off by as much as 10° above or below. That's pretty good for a modeled loudspeaker. Then again, since the best-case vertical polars have nulls 20° to 25° above and below the centerline, shifting up or down by half that much can put the nulls unpleasantly close to the listeners.

A little bit of personal history, as I said, I used to hand-calculate all this stuff, essentially making mathematical models. I was quite careful and my calculations were very good so even now, with the best measurement equipment and development tools, my current designs are almost identical to what I did decades earlier. The crossovers are virtually unchanged. But where changes were made, they were done to shift the forward lobe a smidge, to set the nulls exactly where I wanted them to be, above and below the loudspeaker, outside the pattern. That's just something you cannot do without the visibility that only comes from accurate measurements.

methodology. I had been doing DI-matched two-way speakers and constant-directivity cornerhorns for about two decades, and the crossovers were basically fixed in topography. But I would use a variety of midwoofer drivers and even a handful of horns, something like the Econowave guys do now. In fact, when I saw the Econowave thread start about two years ago, it

identical loudspeaker concept, even using the crossover that I designed. I have really enjoyed watching it evolve, because it was like seeing your child grow up and do the same things you did as a kid.

What I did in the 1980's and 1990's, and through the earliest days of the forum, was to use a sort of cookie-cutter approach. It's really not all that different than what I do today but with some (measurement) refinements for sake of accuracy. I always used the same basic tweeter circuit and CD horns, tending to prefer CD horns without sharp edges. The crossover circuit was always tailored for the drivers and horns used, with the appropriate R1/R2 values to match the sensitivity (i.e. SPL) of the drivers. I also chose woofer circuit values, individually set for each woofer model. This usually involved the Zobel and sometimes in the core splitter filter values too. This is the same thing I do today, except I did it with manual calculations in the early days.

Way back then, I identified sharp edges internal to horns as having an edgy sound. I don't guess I ever saw this as a real "revelation", and instead saw it as a kind of personal preference. My approach was unique at the time, in that I was taking what were prosound drivers and making hifi speakers with them. This was not a common practice, but I always did it, ever since the early 1980's, using JBL 2205 woofers and the like. When I needed lesser expensive components, I used Eminence. This is exactly the same thing I do today.

Back in the 80's and 90's, most CD horns were Mantaray or BiRadial types, but I tended to avoid them, preferring older radial horns. The Peavey CH-3 was my favorite by the 1990's, because it provided a good pattern yet lacked the sharp edges of the Mantaray. The only thing I didn't like about it was its two-piece design, which on first inspection made me hate them, thinking they would buzz after a while. Not a single one ever did, but I always thought they looked bad with that horizontal seam. Still, it was my favorite horn for a long time.

I always thought the thing that made the CD horns with sharp edges sound bad to me was the fact that those edges caused discontinuities that would create acoustic reflections, causing spikes in the impedance curve and ultimately in the acoustic (time and frequency) response. I never focused on internal side-to-side wall reflections, as some now do, but was always concerned about the impedance spikes and the resultant aberrations in response. My main horns of choice were the Peavey CH-3 and Eminence H290, of which I later narrowed down to the H290 alone. I also worked with Martinelli to come up with a wood horn that would provide CD without Martaray style sharp edges, although his horn was not fully what I had envisioned. I later made my own wood horn that was even closer to what I really wanted.

I always made it my goal to provide constant directivity in the horizontal with a 90° pattern. In the cornerhorns, I saw this as being entirely possible through the audio band. Of course, room modes disturb the pattern at low frequency but that's a different subject, addressed with multiple

directivity in the horizontal plane. The free standing two-ways cannot provide CD below the tweeter's passband, but they can be designed to match directivity at the crossover point. This provides some degree of spectral balance, in that the midwoofer pattern collapses as frequency rises up to the crossover point, where it is matched by the tweeter and becomes constant. There is no swell in the power response at the crossover point. This is the design philosophy of the DI-matched two-ways. More properly, these should be called matched-directivity two-way speakers because it isn't DI that is matched, but rather horizontal directivity.
Matching directivity in the vertical and the horizontal planesMy loudspeaker designs also have been given a lot of effort to optimize performance in the vertical plane. That's really the hardest part, in my opinion. Lots of competent designers have put a lot of emphasis on it, and rightfully so. It is important to me that horizontal directivity be matched at the crossover point, and vertical directivity too, in a manner of speaking.

You can't match the vertical patterns, per se, but you can make sure the vertical nulls are spaced outside the intended coverage pattern, and you can use horns with vertical beamwidth smaller than these angles for most of (ideally all of) their passbands. Of course, there are competing priorities, as with all things, and in this case, the balance is in limiting mouth height, which reduces vertical spacing and therefore widens the arc between vertical nulls and increasing mouth height, which can lower the frequency where vertical control is maintained. Generally, a compromise is struck, and the horn size is chosen so that it is not so large the null angles are inside the pattern at HF, but large enough it gains vertical control as close as possible to the crossover point. Usually, the nulls are cutting into a widening (vertical) pattern at MF in the crossover region, and the horn gains control shortly above that point.

It has surprised me, frankly, almost to the point of amazement, to see some designers of DI-matched two-way loudspeakers that place no emphasis on performance in the vertical plane. One that comes to mind is Geddes, who takes great care to match the horizontals and reduce internal reflections, even edge diffraction, but then completely ignores the verticals. As a result of this inattentiveness, the vertical nulls from his loudspeakers in many cases are alarmingly close to the baffle normal, i.e. straight forward.

The Summa, for example, has a vertical null just a few inches from the baffle normal, even several feet back. I believe it was determined you had to sit ten feet back just to have a pattern with nulls outside the dimensions of the speaker cabinet. Basically, this is a gross example of bad verticals. What I think is maybe worse, is the fact that he advertises the speakers as having good polar response, even showing charts of horizontal off-axis response and representing them in a way that purposely hides the poor verticals.

When asked how Geddes represents the data in his charts, his reply is "Its full space, based on only horizontal data, i.e. the polar pattern is assumed to be axisymetric - not perfect, but not bad either. The only error would be at the crossover where there is a vertical lobe that is not represented in the horizontal data." This isn't science, it's marketing, and I'm surprised to see it coming from a man with such impressive credentials. You cannot take a horizontal polar chart and rotate it, expecting the loudspeaker to provide this coverage in all angles. That kind of simplification is unnecessary, it's inaccurate, and even deceptive. As important as the horizontals are, they're relatively easy to get right compared to the verticals, which are also very important.

As I said earlier, most competent loudspeaker designers pay at least some attention to vertical coverage, especially those interested in producing a uniform reverberent field, i.e. constant directivity. The ceiling and floor are usually the two closest boundaries, so what the loudspeaker generates at large vertical angles is a pretty big part of the overall sound field. In my opinion, it would be best if the sound output was minimized at large vertical angles, particularly at medium and high frequencies.