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The Reflection Loudspeaker



PHOTO 1: The Reflection loudspeaker.

I needed two small loudspeakers for the rear channels of my multi-channel system. I had available several Audax AP130Z0 woofers and Audax TM025F1 tweeters from my previous projects, so I decided to design a new compact loudspeaker with these units. The “Reflection” loudspeaker is the result of this effort (Photo 1).

The AP130Z0 (Photo 2) is a 130mm shielded woofer for audio/video use with Aero gel cone and rubber surround, while the TM025F1 (Photo 3) is a 25mm dome tweeter with ferrofluid and a shielded neodymium magnet for audio/video use.

Table 1

Parameter	Audax 1	Audax 2	Audax 3	Audax 4
Fs (Hz)	64.6	61.3	63.9	69.4
Qt	0.54	0.51	0.51	0.53
Vas (liters)	9.5	10.5	10.6	8.3

The passive crossover consists of an acoustic fourth-order Linkwitz-Riley low-pass filter and an acoustic fourth-order Linkwitz-Riley high-pass filter with the crossover frequency set at 2500Hz.

THE BOX

According to the data provided by the manufacturer, the AP130Z0 woofer has the following Thiele/Small parameters:

$F_s = 57.6\text{Hz}$

$Q_t = 0.41$

$V_{as} = 10.85\text{ liters}$

I measured the Thiele/Small parameters on four of the units that I had available¹.

Table 1 shows the results. The averages of these measurements are as follows:

$F_s = 64.8\text{Hz}$ (about 12% higher than the manufacturer value)

$Q_t = 0.52$ (about 27% higher than the manufacturer value)

$V_{as} = 9.72\text{ liters}$ (about 10% less than the manufacturer value)

With these parameters in mind, I chose a box with a net internal volume of 4.5 liters, which gave a resonance frequency at 115Hz with $Q = 0.75$. I constructed the box (Fig. 1) with 12mm MDF and covered it with a wood imitation varnish. I also loosely fitted it with fiberglass.

CROSSOVER DESIGN

Selecting the crossover frequency is an important decision for the design of every loudspeaker. Choose the crossover frequency as low as possible to push the unavoidable rolloff in the power response to the higher frequencies². I chose a

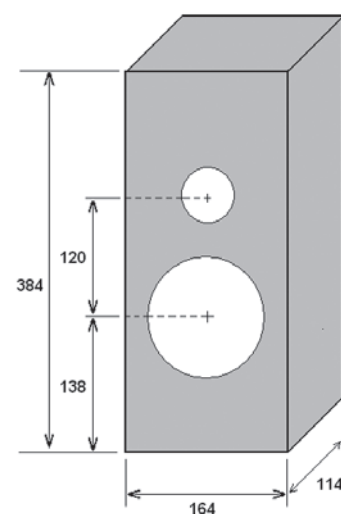
fourth-order Linkwitz-Riley filter with the crossover frequency set at 2500Hz. The 24dB/octave slope will give



PHOTO 2: The Audax AP130Z0 woofer.



PHOTO 3: The Audax TM025F1 tweeter.



All dimensions in mm	
MDF 12 mm	
2 pieces 140 mm x 90 mm	
2 pieces 164 mm x 384 mm	
2 pieces 90 mm x 384 mm	
V _{internal} =4.5 lt	

FIGURE 1: The box dimensions of the Reflection loudspeaker.



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AP-016



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The AP-016 is available in three versions. The AP-016A crossover point is at 2KHz, the AP-016B is at 2.7KHz and the AP-016C is at 3.5KHz. All of them are 4th order Linkwitz-Riley crossover.

Specifications:

HF Power Output: 30Wrms

LF Power Output: 80Wrms

THD: 0.03%

S/N ratio @ rated W: 90db

Input sensitivity: 1V

Input impedance: 22Kohms

4th order X-over: 2KHz AP-016A

2.7KHz AP-016B

3.5KHz AP-016C

Weight: 2.6Kgs (5.7lbs)

Dimensions W x H x D:

W: 137mm (5.4")

H: 218mm (8.6")

D: 81mm (3.2")

Cut-Out W x H:

W: 108mm (4.25")

H: 190mm (7.5")

AC Voltage: 115V / 230V

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sufficient attenuation to the woofer resonance which exists in the region of 5 to 6kHz and will protect the tweeter from high excursions at the lower frequencies. The crossover frequency is a little higher than I prefer, but, unfortunately, the TM025F1 tweeter is not able to provide enough volume capacity for the lower frequencies due to the limited volume displacement.

For the design of the crossover, I used the ORCAD 15.7 demo version (www.cadence.com/products/orcad/index.aspx). The program includes a version of a PSPICE simulation program for electronic circuits. But before I proceed with the analysis of the crossover circuit, I need to determine the following:

- a) the topology of the network
- b) an electrical network that will simulate the impedance of the woofer and the tweeter
- c) a measurement of the frequency response of the woofer and the tweeter when they are installed in the box.

First, I measured the impedance of the woofer and the tweeter¹ and developed equivalent electrical circuits that simulate with high accuracy the impedances of the woofer and of the tweeter. These circuits are shown in **Fig. 2** for the woofer and **Fig. 3** for the tweeter. Then I measured the frequency response of the woofer and the tweeter when they are mounted on the box. These frequency

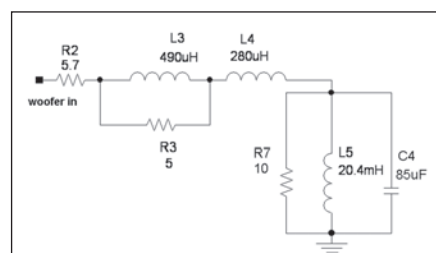


FIGURE 2: The network for the simulation of the woofer impedance.

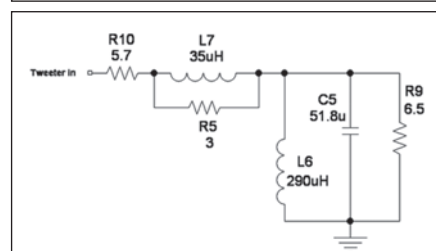


FIGURE 3: The network for the simulation of the tweeter impedance.

responses will be used for the computation of the crossover network.

I took all the measurements in my listening room and this implies some limitations. I placed the loudspeaker on a height 1.5m above the floor and at least 1.5m from the sidewalls. The microphone was about 1.5m from the speaker and its height was about 1.4m above the floor. In all the measurements the microphone was pointed directly at the center of the tweeter. I took several measurements of the woofer in different places in an effort to eliminate the room influence in the low frequencies.

The final frequency response measurement of the woofer, which is an average measurement, is shown in **Fig. 4**, and for the tweeter in **Fig. 5**. Now I am ready to proceed with the design of the crossover network.

I entered the topology of the circuit on the ORCAD Capture Demo editor and, using the special EFREQ device, I also entered the measured frequency response of the AP130Z0 woofer and the TM025F1 tweeter. The circuit as used in the ORCAD simulation program is shown in **Fig. 6**.

The components L1, R_L1, C3, R24, R_L2, and L2 form the low-pass filter of the crossover. The components R2, L3, R3, L4, R7, L5, and C4 form the impedance circuit of the AP130Z0 woofer.

The E2 component simulates the woofer's frequency response, which is entered in discrete frequencies as indicated in **Fig. 7**. For example, the response is entered for 100, 150, 200, 300, 400Hz, and so on, up to 20kHz. Next to the frequency is indicated the amplitude and the phase. Additionally, for the woofer I entered a delay of 60μs in the E2 component, because the acoustic center of the woofer is not in the same plane with the tweeter.

Components G1 and G2 are Laplace functions of the simulation program and compute the theoretical response of a Linkwitz-Riley fourth-order low-pass filter at 2500Hz. The response at the output of these components is compared with the response at the output of the E2 function, which simulates the acoustic response of the AP130Z0 with the crossover. The target of the analysis is to match the two responses as closely as possible.

A similar procedure is followed for

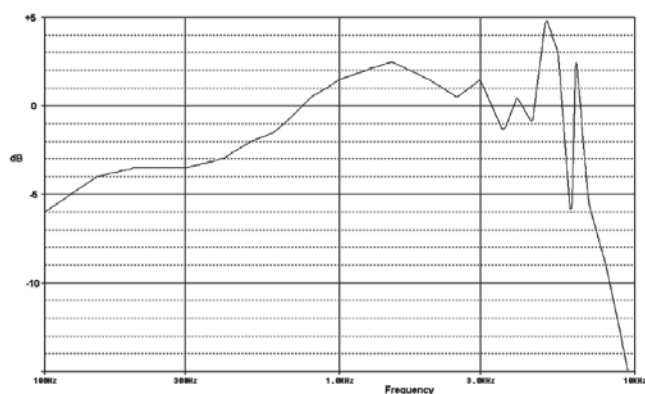


FIGURE 4: The frequency response of the AP130Z0.

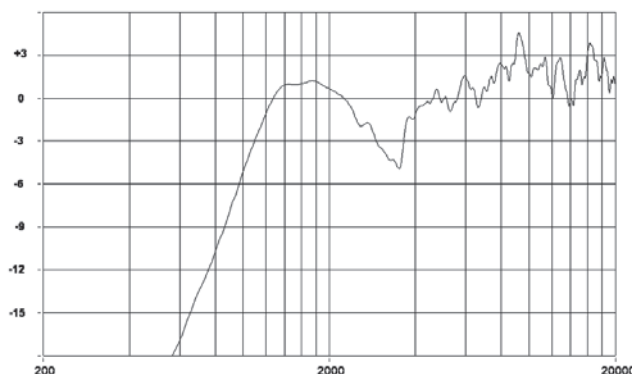


FIGURE 5: The frequency response of the tweeter TM025F1.

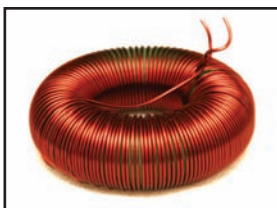
the tweeter part of the crossover. Components R23, C1, L8, R_L8, C8, and C6 form the high-pass filter of the crossover. Components R10, L7, R8, L6, R9, and C5 form the impedance circuit of the tweeter. The E1 component simulates the frequency response of the tweeter as was measured.

As with the low-pass filter, components G3 and G4 are Laplace functions of the simulation program and compute the theoretical response of a Linkwitz-Riley fourth-order high-pass filter at 2500Hz. The response at the output of these components is compared with the response at the output of the E1 function, which simulates the acoustic response of the tweeter with the crossover. Again, the target of the analysis is to match the two responses as closely as possible.

I computed the initial values of the components using the relevant theory for the design of the Linkwitz-Riley filters. Then I modified the values of the components until the response of the theoretical filters were as close as possible with the acoustic outputs of the woofer and the tweeter. The function ESUM is used to add the acoustic outputs of the woofer and the tweeter to simulate the total acoustic output of the loudspeaker.

You can also use the simulation program to analyze the input impedance of the loudspeaker by changing the voltage source V1 to a current source of 1A. Then the voltage as measured at the input of the crossover (junction R23 and L1) is the impedance of the loudspeaker. This input impedance simulation is shown in Fig. 8, which is very close to the real impedance of the loudspeaker as measured in Fig. 14.

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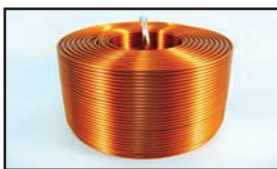


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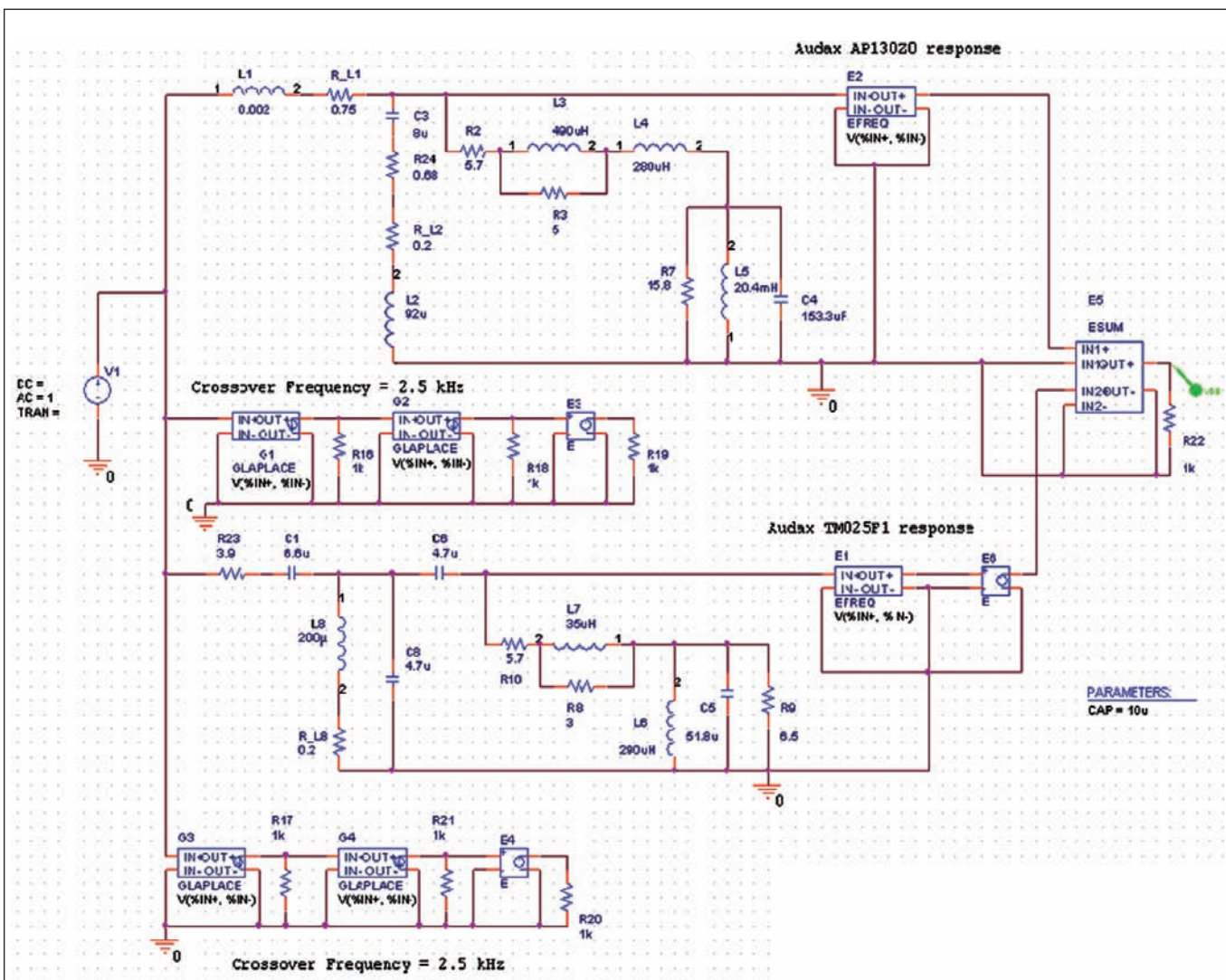


FIGURE 6: The ORCAD simulation network for the Reflection loudspeaker.

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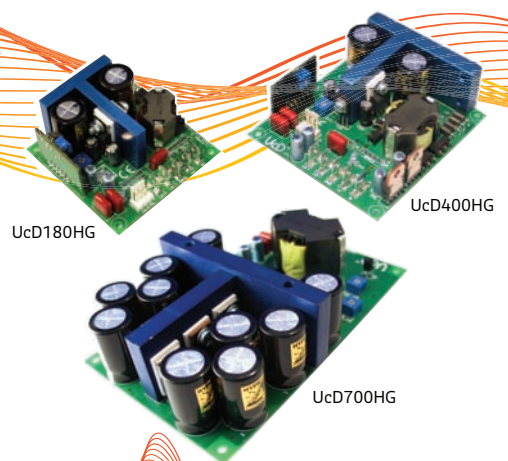
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E_E2      N32214 0 FREQ { V(N105082, 0) } = (
(100,89,80) (150,91,70)
+ (200,91.5,60) (300,91.5,60) (400,92,45) (500,93,40)
(600,93.5,40)
+ (700,94.5,35) (800,95.5,30) (900,96,30) (1k,96.5,20)
(1.5k,97.5,-25)
+ (2k,96.5,-20) (2.5k,95.5,-25) (3k,96.5,-27) (3.6k,93.5,-25)
(4k,95.5,-22)
+ (4.5k,94,-28) (5k,100,-35) (5.5k,98,-60) (6.1k,88,-80)
(6.3k,98,-100)
+ (7k,89.5,-100) (8k,86,-100) (9k,82,-100) (10k,78,-105)
(20k,66,-100) )

```

FIGURE 7: Simulation of the woofer response as used for the analysis of the crossover in ORCAD.

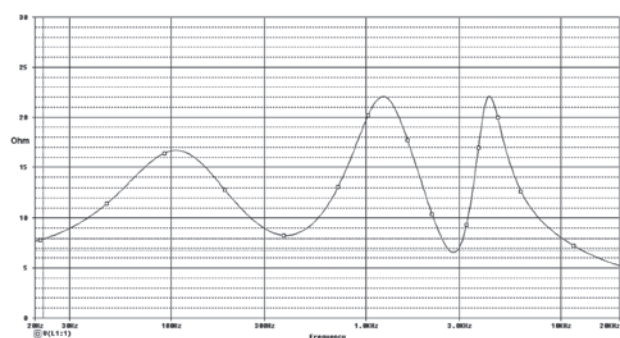


FIGURE 8: PSPICE simulation of the loudspeaker impedance.

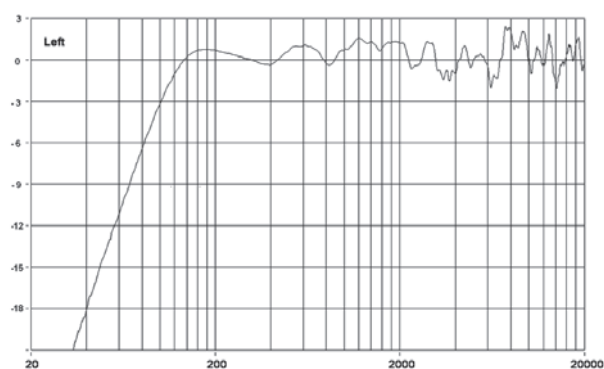


FIGURE 9: The frequency response of the Reflection.

CROSSOVER CONSTRUCTION

The simulation is a very valuable tool, but you need to construct a real circuit in order to confirm the simulation results. So, I quickly assembled the crossover onto a piece of wood and measured the total response of the loudspeaker. I just needed to modify the value of the resistance R23 from 3.9Ω to 5Ω , which reduces the output of the tweeter, to get a flat response. The final crossover of the Reflection loudspeaker is shown in Fig. 12.

For the construction, I separated the woofer and tweeter sections and built the crossovers on two small pieces of plywood (Photo 4). I glued all the components using hot glue. I placed and glued the woofer crossover at the bottom of the cabinet due to the heavy inductor. I glued the tweeter crossover on the rear of the cabinet behind the tweeter.

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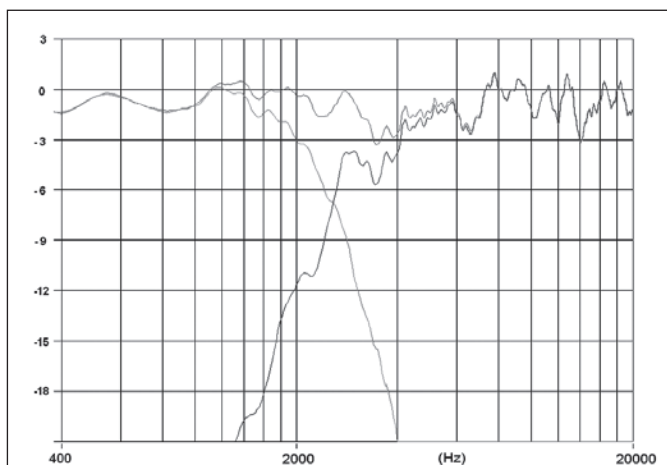


FIGURE 10: The total frequency response with the woofer and the tweeter response.

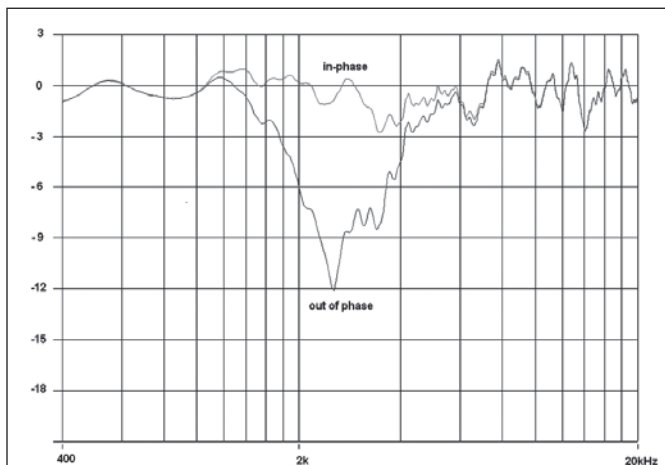


FIGURE 11: The total frequency response with the tweeter in-phase and out-of-phase.

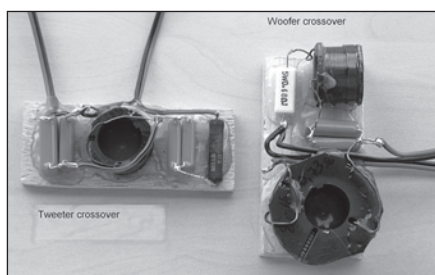


PHOTO 4: The crossover.

MEASUREMENTS

I took the following measurements for the evaluation of the loudspeaker. The total frequency response of the loudspeaker is shown in Fig. 9. I performed this response measurement in two passes according to the method described in reference 3. I took the low-frequency response up to about 240Hz using the near-field technique and performed the

response above 240Hz using a time selective technique of the far-field measurement. The response is flat within $\pm 1.7\text{dB}$ from about 130Hz to 20kHz.

In Fig. 10 the individual response of the woofer and tweeter units are shown. The crossover frequency is very close to the theoretical value of 2.5kHz. Figure 11 compares the total response with the tweeter in-phase and out-of-phase. The notch indicates that the two units are close in-phase at the crossover frequencies, as they should be. In Fig. 13 the horizontal dispersion of the loudspeaker is shown with measurements taken on the axis of the loudspeaker, at 30° and 60° .

Figure 14 shows the measured impedance of the loudspeaker. The minimum value of the impedance is 7Ω at the region of 250Hz and almost 6Ω at 20kHz. This was one of the design targets, because the length of the cable between the surround loudspeaker and the amplifier usually is very long and this could result in frequency response irregularities, if the impedance of the loudspeaker is low. The maximum impedance value is 21Ω at about 1100Hz. The phase has a minimum value of -40° and a maximum value of $+40^\circ$.

CONCLUSION

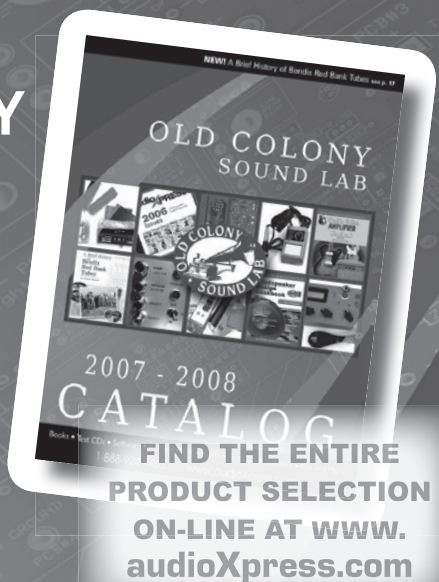
The "Reflection" is a small, low-cost, good-sounding loudspeaker which is very easy to construct. But don't expect thunderous bass from this speaker with its small woofer. Although it was designed to be used for the surround channels of a multi-channel system, you

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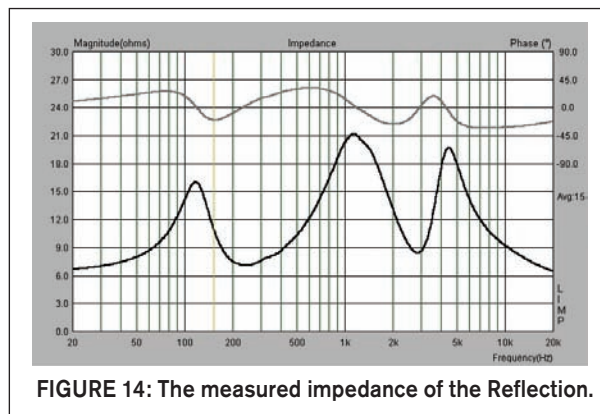
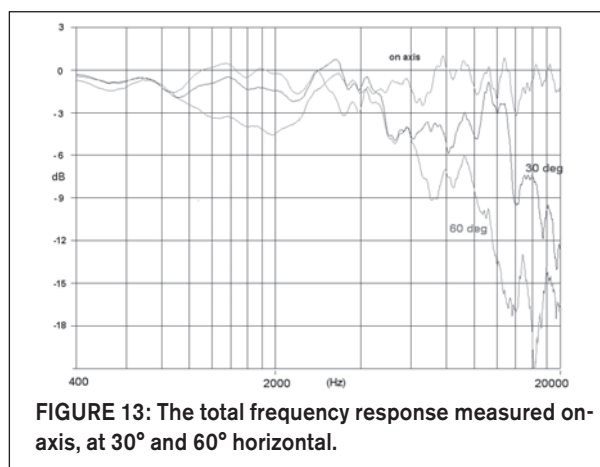
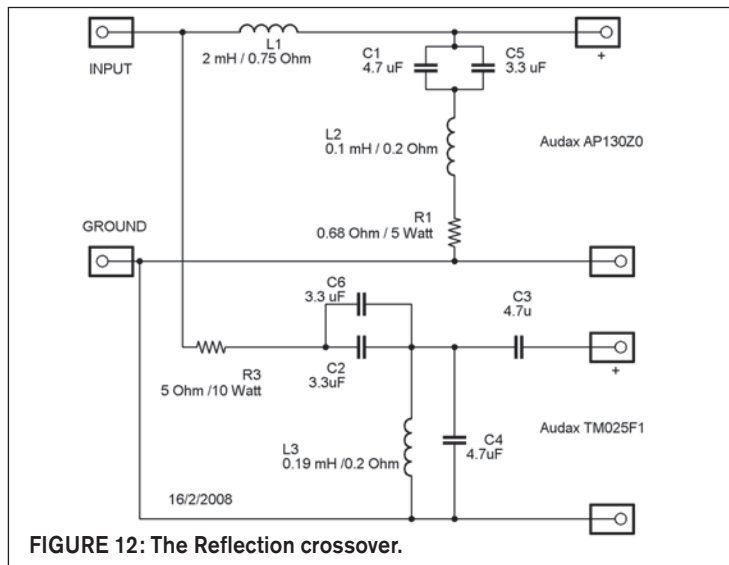
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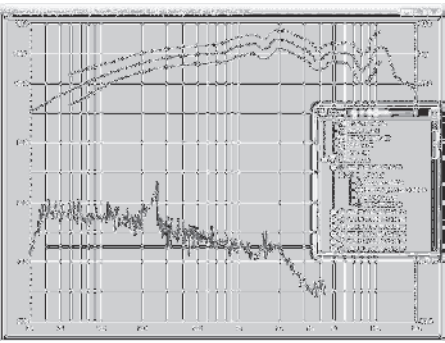
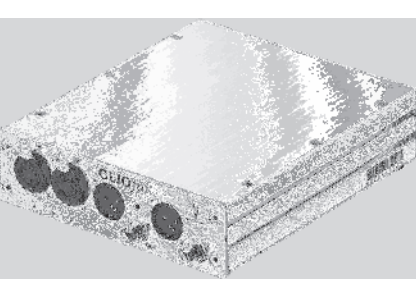
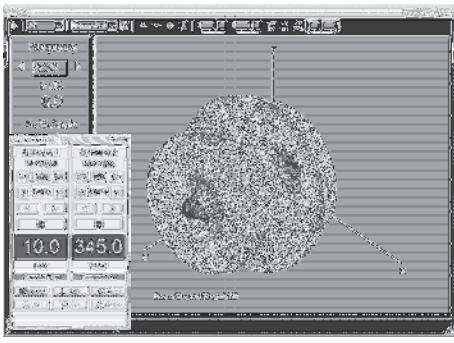
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3. C. Struck and S. Temme, "Simulated free field measurements," Bruel & Kjaer.
4. Measured with the demo version of ARTA software (www.fesb.hr/~mateljan/arta/).



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The Borbely Annals

Erno Borbely is not a man given to bragging about his accomplishments. But from the venerable Hafler DH200 to today's high-performance all-FET amplifiers from Borbely Audio, he has left his mark on 40 years of technological audio advancement. Jan Didden looked him up in his home close to Munich, Germany, and talked with him about audio and other tidbits.

Jan Didden: Erno, when we had our first contacts to prepare for this talk, you told me of the events during the early part of your life. Would you give us some background about that?

Erno Borbely: Well, I was born in that very unstable time, just before WWII, in the Hungarian village of Aszöfö. Because I found some electronic parts during the war, I became interested in electronics and decided I would become an engineer. After two years at the Technical University of Budapest, I had to flee my homeland in 1956 when the Russian military invaded my country. Initially, I managed to get to Austria. I spent a month in a camp until the Norwegian Student Organization offered me passage to Norway and admission to their Technical University in Trondheim. Both the Austrians and the Norwegians were very friendly and helpful with the refugees.

I arrived in Norway on Jan. 15, 1957, and after a short stay in a camp, we students were placed with Norwegian families and studied Norwegian at the University of Oslo from January to June. I was admitted to the Institute of Technology, University of Trondheim, and continued my studies there. I earned my master's degree in electronics in 1960; my thesis was the design of an arithmetic unit for a CPU, with discrete Philips germanium transistors clocked at a phenomenal 250kHz! I worked at the electronics lab of the Institute until 1961, designing instrumentation for the lab. Then in 1961, I got a job at the Norwegian Broadcasting Corp. in Oslo as a design engineer. For the next eight

years I designed studio equipment, both in tube and semiconductor technology. I was also teaching sound engineering courses at the Broadcasting Corp. and at the Norwegian Engineering Society. I have written textbooks and co-authored a book on semiconductor technology in Norwegian (**Photo 1**). I became a Norwegian citizen in 1964 and married Irene in 1966.

JD: How did you get to work for Dynaco in the US?

EB: Well, I met David Hafler in Stockholm and he offered me a job at Dynaco in Philadelphia, and we moved there in 1969. Working with Dave was very interesting; I had to learn to design simple, inexpensive circuits which nevertheless sounded good! I designed the DYNATUNE circuit for the Dynaco FM-5 FM tuner, for which I received a US patent.

Another interesting development was the Dynaco QD1 Quadaptor (**Photo 2**).



PHOTO 2: Dynaco surround sound QD-1 Quadaptor.



PHOTO 1: Mr. Borbely shows the book on semiconductor technology he co-authored while working in Norway.

This was a passive quadraphonic unit that you put between the stereo outputs of your power amp, and then connected two additional speakers to the QD1, in addition to the regular speakers. The QD1 extracted some of the ambience sound hidden in the sum- and difference signals and sent that to the extra speakers. Dave had special records produced to demo the effect (**Photo 3**). This again showed that Dave was very talented and creative! Both the QD-1 and the later QD-1 Series II are still available today—for example, on eBay.

At around that time Dave and I also started to discuss designing a big power amp, and he asked me to look into the possibility of a high power output stage.

Unfortunately, Dave then decided to sell Dynaco to TYCO, and he stepped



PHOTO 3: Dynaco had special records pressed to demonstrate the surround-sound produced with its QD-1.

down from daily management. They brought in Frank Johnson as president and he named me Director of Engineering in 1972. So, I was working on that high-power cascode output stage and Frank brought in Jim Bongiorno to tackle the voltage amplifier stage. But progress wasn't fast enough for Frank, and it was decided that I and another colleague, Harry Klaus, would team up together to complete the 400. Anyway, I was wrapping up my work on the 400 at the end of 1972 when I decided to leave Dynaco. I suggested doing a patent investigation into the cascode output stage, but I guess Dynaco never did. Overall, I think the 400 was overkill. Harry Klaus designed a simplified version later on, the 410 (simpler heatsink, no Dynaguard, and so on), which was also a better sounding amp.

JD: That was when you moved back to old Europe?

EB: Yes. We moved to Switzerland and I worked as Applications Engineer/Application Manager at Motorola Semiconductor. I met John Curl there and

we had a good time discussing audio circuitry, and from my position as application manager I was able to provide John with samples of low-noise Motorola semiconductors. I was designing low-noise circuits with bipolar transistors and JFETs, power amplifiers with bipolars, and also an FM tuner with dual-gate MOSFETs. You see, initially, JFETs became available as an alternative to low gain and low Z_{in} bipolars. They were used in very diverse equipment, instrumentation, RF, later also audio. At first they couldn't match the low noise of some of the better bipolars though, but that gap was closed quickly. The early ones had appreciable C_{dg} , so we cascoded them, first with bipolars, later with other JFETs.

I was pretty happy in my job when Dave Hafler came to visit us again in Geneva in 1977 and told me that he was going to start a new company, and offered me a job as Director of Engineering. So, we moved back to the States in 1978 and I worked for Dave again at the David Hafler Company. I designed the DH101 preamp and also the DH200

(Fig. 1) power amp, which was the first one to use MOSFETs in the US.

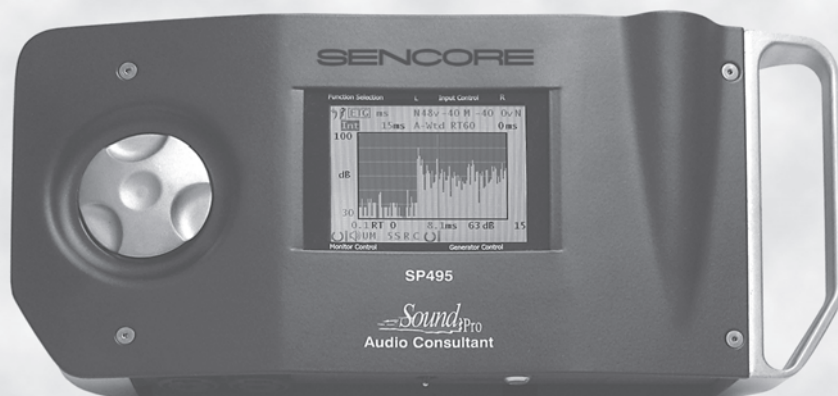
JD: These were the Hitachi lateral MOSFETs?

EB: Yes, for me these sound the best of all MOSFET types. At low bias they sound kind of soft, and come very close to tubes. They have relatively low G_m but the negative tempco is a plus. Vertical types such as the Toshiba's have higher G_m but a positive tempco, which makes it more difficult to stabilize the quiescent conditions. They can sound quite good as well, especially in the bass department, although for full-range I would always prefer the Hitachis. And the Hitachis can sound good with only 100mA, while the Toshiba's would need at least double that.

I don't like the IR-type power MOSFETs, although they have a very high G_m . When I used them I had an extra identical device on the heatsink as the bias regulator to keep the positive tempco under control. They measure well but they are not my favorite for sound quality, unless you go to very high bias cur-

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rents or other topologies such as Nelson Pass is using.

JD: You mentioned tubes; have you done any tube work in audio?

EB: Oh, yes, I started out in tubes in Norway. It's a funny thing with tube power amps. In my opinion—and, of course, this is a *very* general statement—they have less resolution and transparency than a good FET amplifier. Tube amps generally do color the reproduction, and that can actually be quite pleasant, but it isn't the original sound. Yet, I very much enjoy listening to my favorite music through a tube amp. I loved those E130s! One of those audio mysteries that we'll probably never solve!

JD: Would you use negative feedback in those tube amps?

EB: Well, for power amps in general, I am convinced that negative feedback (nfb) is a worthwhile tool for the audio designer. I don't believe that nfb "kills" the sound or that it comes too late to do

any useful correction because it can only correct after the fact. Delays in audio equipment are too short for that; I mean we have phase shift in the forward path, but the effect of that is to limit the positive effects of nfb as frequency goes up. On the other hand, nfb should not be the only tool used. But with tube power amps in particular, I think you should use some feedback to bring down the output impedance. An open loop tube power amp without nfb has appreciable output resistance and it will be very difficult to keep the speaker under control.

JD: You're not afraid of TIM and related overload issues in high feedback amplifiers?

EB: Otała showed us the importance of adequate slew rate to avoid internal overload and a few other things. TIM in itself was not new but it did re-focus us on some important issues and as such contributed to better design practices, but it was only one of the important issues. I think Matti at a later stage real-

ized that also. Walt Jung's series on slew rate distortion was another important contribution. My preference is to have a reasonably linear forward path in an amp, and then to add some feedback to it for two main reasons: first, to stabilize the DC and AC conditions, and, second, to have a reasonable damping factor to keep the speaker under control. This is an important point. You shouldn't use nfb to "fix" a bad design, but to clean up and stabilize the operating parameters of an already pretty good circuit. I can get good sound with or without feedback, but with some feedback it becomes better—repeatable and stable. I am always measuring my designs for open loop linearity. I have a bank of capacitors that I use to break the feedback loop for AC, but leave the DC loop intact to avoid excessive offset and drift. Every time I work from these principles, I get good-sounding amplifiers, and my customers seem to agree!

JD: So, you were back in the States

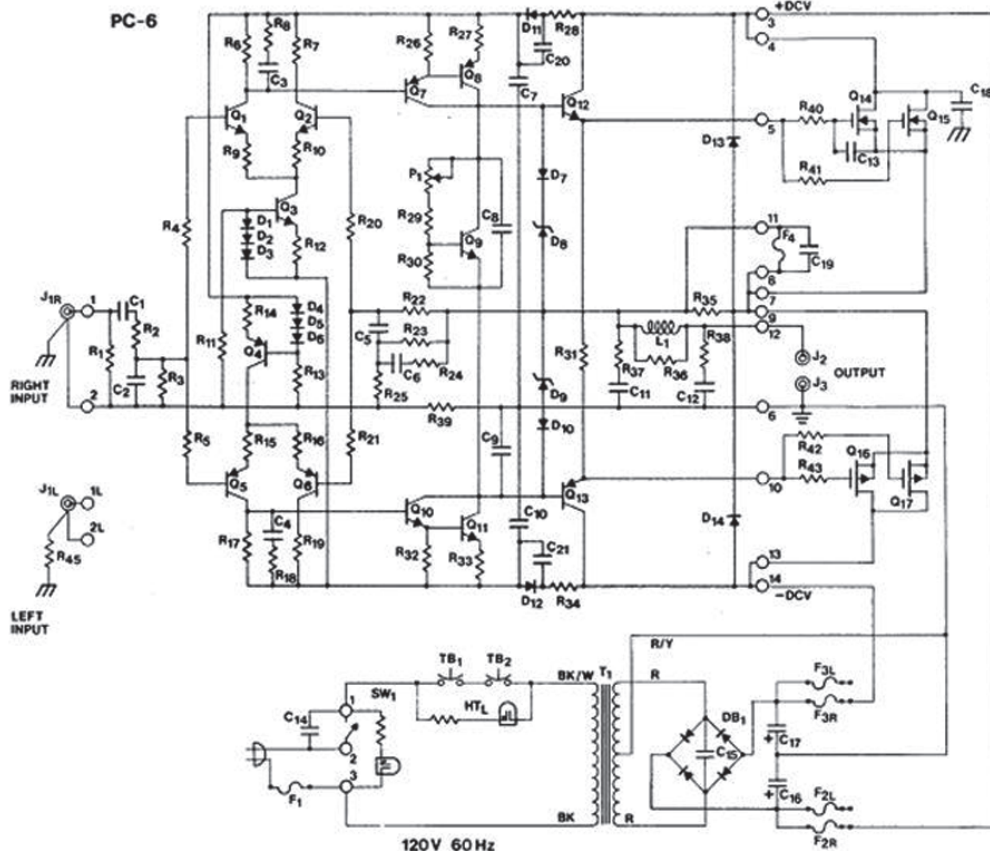


FIGURE 1: The circuit diagram of the Hafler DH200; you can download the full construction manual from www.hafler.com/techsupport/pdf/DH-200_amp_man.pdf.

working for Dave.

EB: Not for very long, alas. We did buy a house in New Jersey, and Irene spent a year fixing it up. Unfortunately, I had some disagreement with Dave about shares in the company, and decided to leave after just over a year. This was a big mistake, as the David Hafler Co. did very well for another ten years and made a lot of money on the sale! And, of course, Irene never forgave me for leaving the nice house she built up. We moved to Germany and I worked at National Semiconductor as European Training Manager. I started publishing my articles in *The Audio Amateur* and the Danish *High Fidelity* journal in the beginning of the 80s. We started Borbely Audio with Irene in 1984, initially to supply difficult-to-find components for my designs in (then) *The Audio Amateur*. Later on we started to sell some of the kits that I published. Irene was running the show, as I was busy at National, and could only contribute during the weekends.

JD: If memory serves me well, all your

Audio Amateur (currently *audioXpress*) designs were strictly FET-based, including the power supplies.

EB: Not really. The first power amp I published in *Audio Amateur* 2/82 was all bipolar, except, of course, the MOSFET output. The first JFET input phonos and line amps were also using bipolars in some stages—see, for example, “The Borbely Preamp” in *AA* 4/85. But, I like FET devices; I can design circuits with them that are not very complicated, that are well reproducible without tricky adjustments and compensations, and, most important of all, they really sound good. Also, their wide bandwidth is a big plus. There is a difference between sounding very good and absolute measured linearity; I don’t have to convince you of that. As I said earlier, tube amps don’t generally measure well but can sound good. But for me, FET-based amplifiers overall sound best in most situations, speakers, and music.

JD: Ironically, we now see that many of the JFETs are becoming obsolete, such as the Toshiba J74, following the fate of

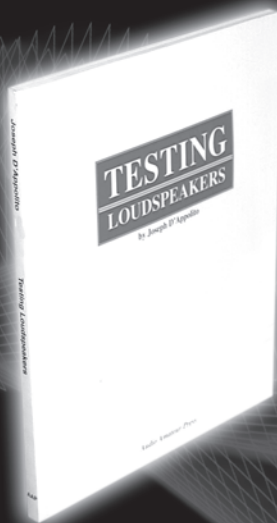
the obsolete dual K389 and J109.

EB: That is right. I think it was John Curl who pioneered the E110, E212, and E175 in the JC-2, and started using the Toshiba in the Dennesen JC-80. I am still using them in my own designs. There are companies that market replacement devices, but in my experience they are not as good as the original Toshiba ones. In fact, there have never been good replacements for those big chip Toshiba with 35Ω noise resistance, which were already obsolete in the 90s.

JD: How about power supplies? Many of your designs use regulated supplies, with FETs, of course!

EB: Yes, call me Mr. FET (smiling). There are two types of supplies I use. One is an LC input supply followed by a very low noise, very stable series regulator, with discrete FETs. The other is a very low noise shunt regulator. It’s often a matter of personal preference. My friend Are Waagbø used a combination of a series current-source feeding into a parallel shunt, and that is probably the very best you can do for regu-

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lated supplies. He described the design in his February 2008 *audioXpress* article ("Shunt or Not," p. 30).

What I find quite important in supplies is to use reservoir caps with the lowest ESR practicable, and use lots of it. I believe in very stiff raw supplies. That makes sure that the raw voltage going into the regulator is as smooth as possible. And, by the way, Walt and you have also done a lot of good work on regulators!

My preference for power transformers are R-cores. They have the best isolation from the mains noise and interference, and are dead silent mechanically. If you use toroids, you should use potted types to make sure there is no mechanical noise.

JD: I agree. Toroids are too wide-band to use for low-level circuitry. I used to have transformers wound for me in Germany. For, say, an 800VA transformer I would tell them to wind it on a 1200VA core. Those transformers would be absolutely silent, mechanically, without potting. Of course, you pay the 1200VA price. Another question: I don't remember seeing any protection system on your power amps?

EB: Oh, I have an output protection circuit for those customers who wish to use one. I never used one with my own amplifiers. Those output FETs are so rugged, I never had any problems with that. I noted on your website that you favor that special-purpose output relay from Amplimo, and one of my friends told me I should look into it; I may do that.

JD: When did you decide to retire from National?

EB: That was in '97 when National closed their European Training Center. I continued and expanded designing and selling high-performance kits worldwide. We have experienced some problems lately with the €/ \$ exchange rate (1:1.55 at the moment) and for our most important market, the US, pricing is difficult. We also face significant problems with the Postal service. Besides the fact that Germany has closed most of the normal post offices in our neighborhood, there is no basic insurance on packages to the US, Canada, Australia;

starting July 1, 2008, the German Post service will no longer accept any insured packages to the US, which forces us to use more expensive options.

Besides the kits, I have been designing amps for OEM customers. My most important OEM customer was PBN Audio in California. Peter Noerbaek of PBN is a very famous speaker designer and also built some of the best amps on the market. I was hoping that Aural Audio, which was established by Larry Black, would become the next high-end supplier, and indeed Larry got some very good reviews for his first products. (www.6moons.com/audioreviews/aural/audition.html) Unfortunately, he passed away at a very early age of 47. One of his dealers is considering resurrecting Aural Audio; we'll see what comes out of it.

JD: I noticed that most of your designs are fully published with schematic diagrams and even board layout, either at your website or in *audioXpress* articles. You make a point of noting that those designs are your intellectual property and that commercial users should get a license. Are your designs being copied without your approval?

EB: Unfortunately, yes, there are unscrupulous individuals and companies that try to make money from my work. Not just the single-person Far East outfits, but also established European and US-based companies. I'm sure my fellow designers also suffer from it. It is plain and simple theft, and difficult to prevent. I provide support and assistance to my clients when necessary, which those copycats don't, of course, and that is also worth something. (Says Mrs. Borbely, half jokingly: Erno is too nice. He can easily spend an hour on the phone to explain to a customer the ins and outs of one of his kits. He likes that kind of interaction, to be a teacher.)

JD: You've seen audio develop to a very high standard the last 40 years. Where will it go the next decades?

EB: I'm not sure. On the one hand, I fear that the "iPod generation" no longer cares about high-quality audio, but more about having their favorite sound always and everywhere. Supported by producers who seem to be mainly concerned with getting the highest loudness levels.

The situation is neatly described in an article in *Rolling Stone* magazine (www.rollingstone.com/news/story/17777619/the_death_of_high_fidelity). On the other hand, I see still a lot of young people going into audio DIY for serious listening, besides using the iPod or MP3 player when they're on the road. But it must work both ways. It's up to the record producers to make sure we have well-produced and recorded music that makes our equipment worthwhile!

In a sense, the possibilities for DIY audio are greater now than 40 years ago. You don't need to do an extensive study but you can still get the DIY-audio reward from building relatively simple tube amps or gain clones. So, maybe the future of quality sound reproduction is not that bleak after all. And an added advantage of tubes, for DIY, is that they're not converting to SMD-tubes in the foreseeable future!

JD: Do you participate in any of those online audio forums?

EB: Not really. I sometimes scan the discussion at diyaudio.com, for example. But the tone of some of the comments is often very hostile and negative. I understand that people can have very strong opinions on audio issues, but it should at least be possible to discuss in a civilized way, without using swear words or insults. I think that's why Walt Jung left that board, and look at some of the attacks John Curl has to endure. I think the main problem is to separate the chaff from the corn; there are some very good discussions there.

JD: I have been participating at [diyaudio](http://diyaudio.com) for about five years now and I know what you mean. I have also written some posts that in hindsight I regret. Part of the difficulty is the mix of, on the one hand, accomplished designers such as John, Nelson Pass, Charles Hansen, and so on, and, on the other hand, young, inexperienced DIYers. There are those who recognize the great possibility to learn, and others who want to show up the old guys. It's a very direct and immediate medium and that appears to increase the emotional content. The discussions in the *AES Journal* and, in earlier times, in *Wireless World* were also often very tough and potentially damag-

ing, but in a printed medium it seems less direct, and, of course, you can't react immediately so you mull it over and that softens the reaction.

EB: Yes. It is a worthwhile and very efficient way to exchange information and opinions, but it should be properly moderated. I think it would be a good move by *audioXpress* to sponsor an audio forum related to its articles and publications.

JD: Do you use simulation in your design work?

EB: Not as much as I've done in the past. I remember getting some ADI SPICE models from Walt Jung in earlier times. Seems to me, unless you are sure of very accurate models, you can have a design that seems fine and sounds bad or vice versa. I don't say a simulator is not valuable for some work; it is a great tool for "proof-of-concept" type of problems. I have less need for it nowadays.

I have most of the test equipment I need in my lab (very well equipped lab!—**JD**). I have a vintage HP distortion analyzer, but I use Bob Cordell's audio generator from his 1981 *Audio* article (www.cordellaudio.com/instrumentation/thd_analyzer.pdf) because it is better than the HP built-in oscillator. I can measure noise to 100 μ V full scale. I'm also thinking of getting some of Bill Waslo's stuff from Liberty Instruments, for the Juli@ soundcard. They're very flexible and high quality.

[At that point Mrs. Borbely (call me Irene) enters the lab. It is immediately clear that she knows the audio scene very well. They run Borbely Audio together, with Irene handling all the business issues, paperwork, money, shipping, and so on. As Erno mentioned before, it wouldn't have worked without the strong support of his wife. And her Hungarian goulash is the best I ever tasted, not to mention her beautiful paintings that grace the walls of the Borbely residence.]

JD: What's the latest project you have added to your lineup?

EB: That would be the DAC (www.borbelyaudio.com/pics/HB_DAC1704_V1_01b.pdf). Together with Miklós Kiss

of Whiteful Audio (www.whiteful.ini.hu) and Johannes Frederiks we've done a lot of research on that—jitter, oversampling, filtering. We get very low jitter by using a Sample Rate Converter, and then re-synchronizing the signal again after the SRC. We don't use oversampling filters because their pre- and post ringing causes the sound to be a bit "slushy" by want of a better word. We do the upsampling by a very precise interpolation in the SRC, and no down sampling filtering is needed when using the direct downsampling feature of the new high-end chip (SRC4392) from Texas Instruments. [To verify the difference between an SRC as "oversampling filter" and a direct audio chain without any digital filtering, the design includes a switchable non-oversampling mode—**JD**] Miklós designed several test boards, and finally we selected the best R/2R DAC currently available, the PCM1704 in a fully balanced configuration (using two 1704s per channel). Johannes designed the final HB-DAC1704 board. I'd been trying many different DAC chips before

deciding on the 1704, which for me gave the best overall sound. And the best I/V solution then is an active I/V; preferably with many-MHz bandwidth.

But I have also done a lot of research into power amplifiers since I developed that Hafler DH200, which has culminated in my Millennium and Ribbon power amps. They are similar in design, with the Millennium able to provide somewhat more output power (**Fig. 2**). With a maximum supply voltage of ± 50 V (± 55 V unregulated) and a EB-117/400 output stage (**Fig. 3**), the Millennium can deliver up to 100W into 8 Ω . If the power supply and the output stage/heatsink are chosen correctly, 75-80W of this can be delivered in pure Class-A. They both are fully complementary designs, using only FETs as active elements. The input stage is a complementary differential cascode FET circuit, using dual monolithic JFETs as input devices and MOSFETs as cascodes. The second stage consists of two JFETs in parallel and a MOSFET connected in cascode. The JFETs are operated at 8mA each, providing a total of

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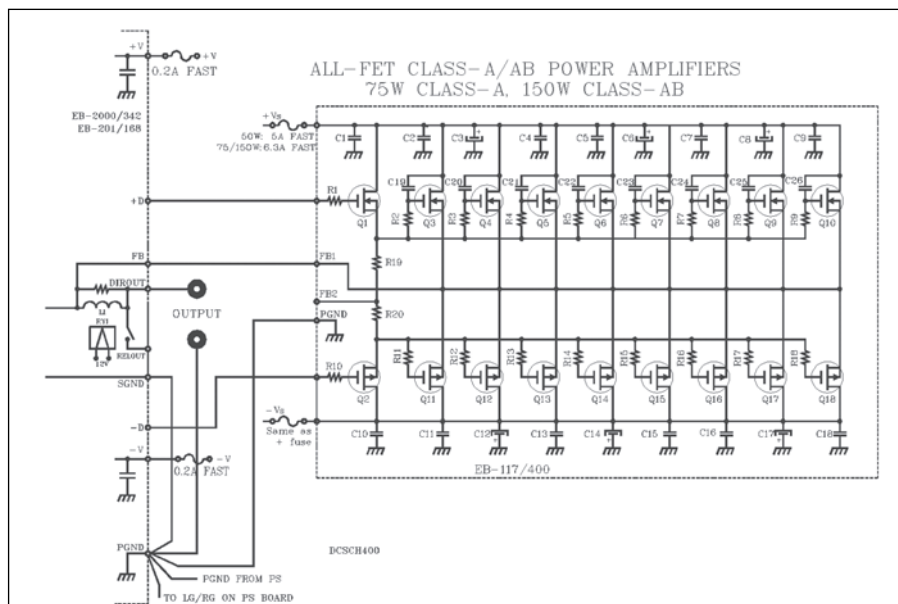
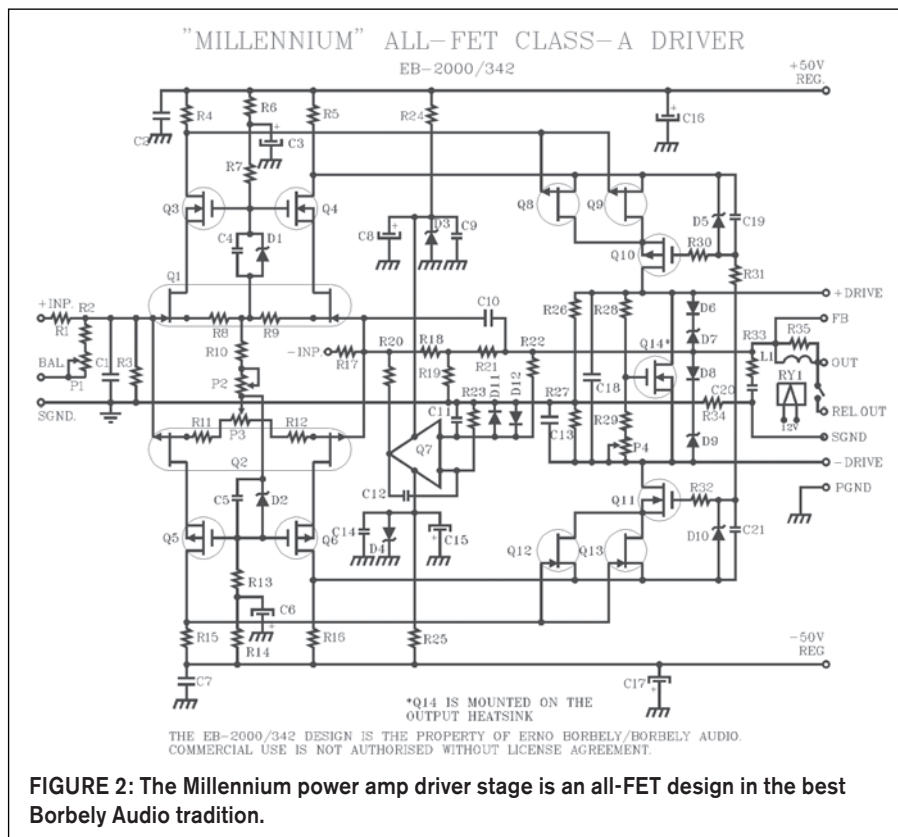
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16mA drive for the output stage.

The bias circuit is a V_{GS} multiplier, using a MOSFET. The MOSFET itself, a TO-220 device, is selected according to the MOSFETs used in the output stage and is mounted on the output heatsink, if necessary. This ensures proper bias tracking with the different output MOSFETs.

The output offset is tracked and controlled by Q7, which is a JFET-input op amp. Two shunt regulators provide $\pm 10V$ supply voltage for the op amp, while the driver itself is also fed from fully regulated, all-FET power supplies.

These power amps are all-round top performers, and really shine with ribbon



or electrostatic speakers that benefit from their transparency and high resolution. And again, my customers seem to agree!

JD: I see some pretty exotic parts here.

EB: Not for exotic's sake, but I do find that parts are important for good sound. I use Dale resistors, with Caddocks as an upgrade option. When I need small capacitors, I have used silver-mica, but after a lot of feedback from my customers as well as my own evaluations, I am now favoring polystyrenes. The problem is that not many manufacturers produce them, and if they do, only a few different values. So it's a continuing search for good parts these days.

Another related concern I have is the current switch to lead-free solder and lead-free plating on PCBs. We need higher temperatures to solder them, which is harder on the parts. I realize the environmental issues, but we don't know how good, or bad, those solder junctions will be 5, 10, or 20 years from now.

JD: Do you have any distributors or partners?

EB: Yes, I have two partners at the moment: Les Bordelon of LBAudio in California and Tai-Shen Lee in Taiwan. Les is doing custom assembly for North American customers. Les' amps can compete with any commercial product on the market, both in sound quality and in professional packaging (**Photo 4**). Tai-Shen is making phono preamps for Taiwan, which is one of the biggest vinyl markets in the world.

JD: Well, Erno, we sure covered a lot of ground in this talk, and I am very grateful for your willingness to share your history, your experiences, and your opinions about audio issues. Any last things you wish to mention?

EB: I must tell you that after 40+ years, I still very much enjoy trying to push audio performance. As you realize, I am now into my seventh decade, and most people would have retired already many years ago. For now, both Irene and I still like running Borbely Audio, but soon I will need to make that retirement decision. If I have a regret, it would be that I never got back into the pro audio field. I have designed some excellent mike preamps and sold a few kits, which custom-

ers claimed were much better than the most expensive commercial ones. However, to succeed in this field you need to offer complete units and a product line, and that wasn't an option at the time. But, as you've heard, I like to talk about audio in all its aspects, and I thank you and Ed Dell for the interest in my work.

audioXpress is a great magazine, and Ed has done a great deal to further DIY audio. We should recognize that as well.

[JD note: Most of Erno's designs are documented in articles written for *Audio Amateur*/*audioXpress* magazine, and are reprinted on www.borbelyaudio.com.] **aX**

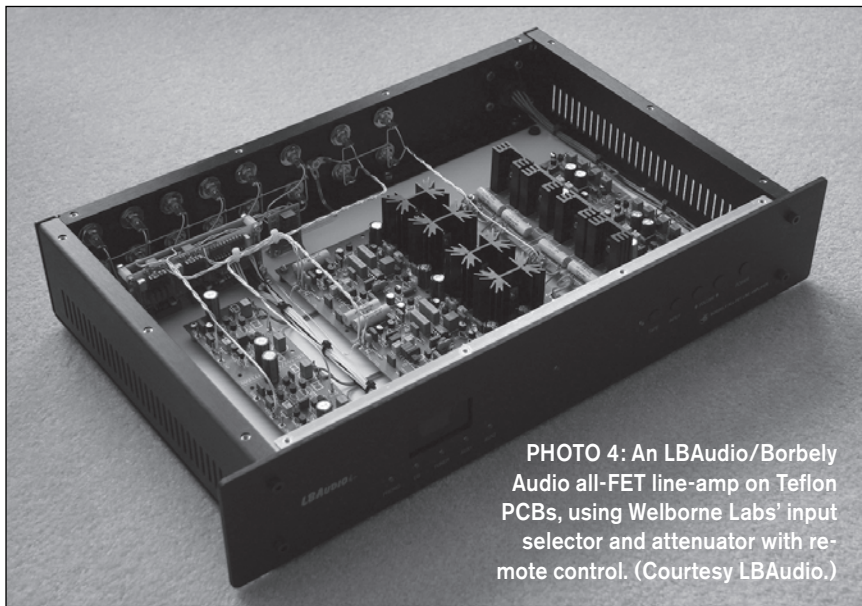


PHOTO 4: An LBAudio/Borbely Audio all-FET line-amp on Teflon PCBs, using Welborne Labs' input selector and attenuator with remote control. (Courtesy LBAudio.)

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Designing a Midrange Horn

This cone speaker covers 200Hz and higher frequencies, up to three octaves of the audio band.

Before we get to the project part, I'll describe some design characteristics of midrange horns. Across the mid-frequency band, midrange frequencies have a ray characteristic and act less like waves. As rays, the frequencies tend to not bend around corner reflectors. Thus, the horn path is typically straight and not folded when it comes to midrange.

The only exception is using a circular duct with circular corners, as musical instruments—trumpets, trombones, and the like—are built. The off-axis frequency will vary depending on the horn expansion type. High-frequency beaming is another term used to describe the off-axis variance in frequency.

There is less of a need to use mutual coupling¹—for example, a corner or double cabinets—to extend the lower midrange frequencies as with bass horns. The practical midrange horn mouth size is within reach of the builder compared to low-frequency sizes. I will show that a quarter wavelength horn is sufficient for midrange bandwidth. Quarter wavelength corresponds to the corner position or a 1/8 space horn expansion. Reactance annulling² extends the frequency response at or near the cutoff (in this case 200Hz) allowing a flat response right down to 200Hz. The same effect used to extend bass horns response is applicable to midrange horns.

A circular horn mouth provides identical off-axis response coverage in the horizontal and vertical plane. This is not always desired. An excellent compromise is to make the expansion in the horizontal plane tractrix curved and the expansion in the vertical plane radial (straight angled) expansion. This will minimize beaming in the horizontal plane and help remove or reduce floor reflections due to the beaming in the vertical plane.

THE INITIAL PROJECT

I started this project by building a mid-

range/midbass horn for a friend who wanted to use a 12" driver—the JBL 2020H—to cover the midrange/midbass

frequency band. Throughout this article, I use my custom horn calculator software tool, Hyperhorn, for horn design.

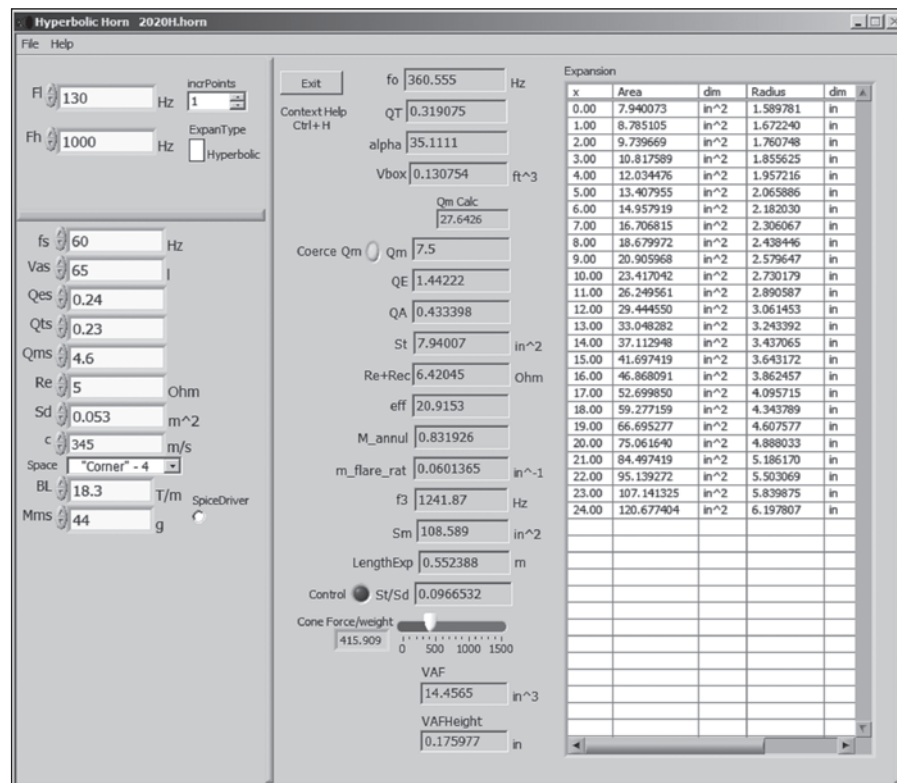


FIGURE 1: Hyperbolic horn design for the JBL 2020H driver. The chosen low and high cutoff frequency is 130Hz and 1kHz. The horn length requires a length of 60cm (24”).

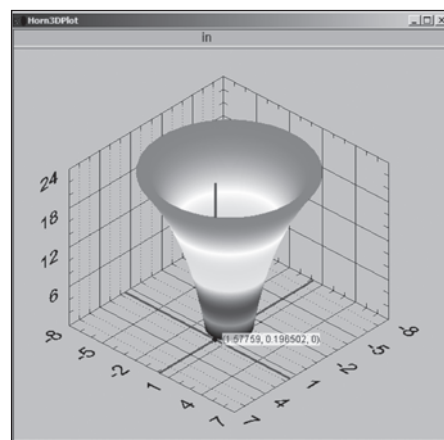


FIGURE 2: Three-dimensional plot of horn design in *Fig. 1*.

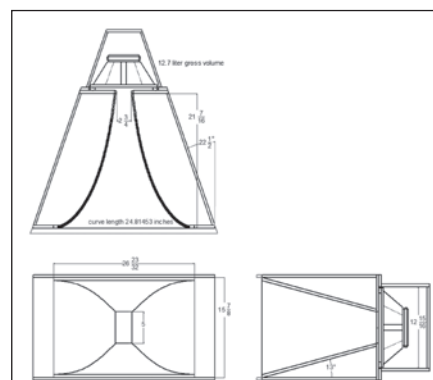


FIGURE 3: CAD drawing of horn designed in Fig. 1. A wood brace (not shown) near the mouth is glued on the internal side to stiffen the tractrix curve.

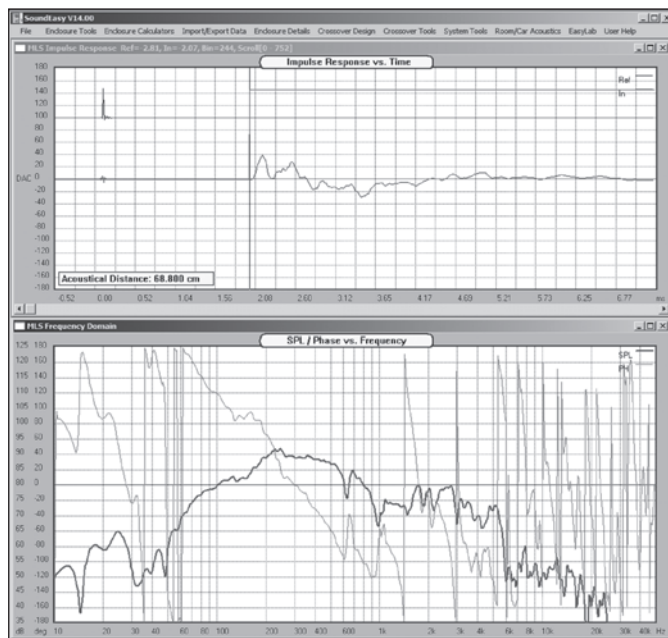


FIGURE 4: SoundEasy measurement data for the horn shown in Fig. 3 CAD drawing.

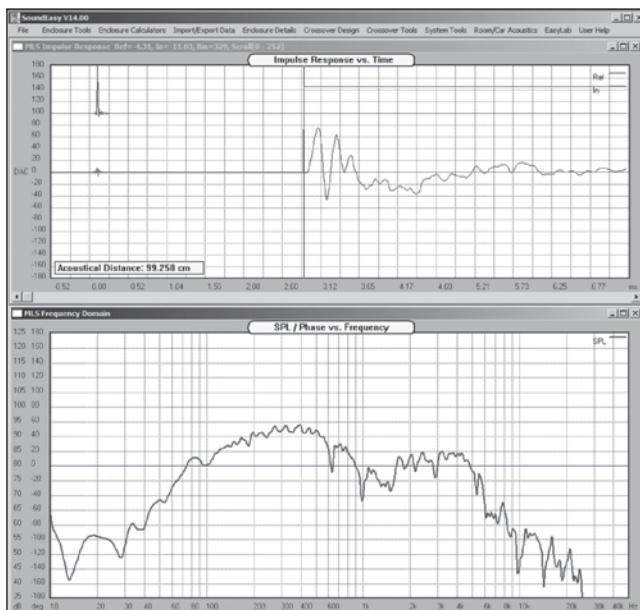


FIGURE 5: Frequency response for a double stack JBL 2020H tractrix horn of Fig. 3. The frequencies below 200Hz are raised where there was a dip shown for a single cabinet.

Enter the Thiele/Small data of the woofer and the horn calculator does the number crunching (Figs. 1 and 2). This driver appeared well suited because it was designed specifically for midbass/midrange. **Figure 3** shows the dimensions of the horn via a CAD drawing, while **Fig. 4** depicts the frequency response of this horn cabinet in free air, i.e., mounted on a speaker stand.

At this point let's access the JBL 2020H tractrix horn design and note where it does not meet the design requirement. Rolloff begins at or near 700Hz and there are two nulls positioned in the band of interest. One null occurs just above 600Hz, and the second is at about 1kHz. Studying my CAD drawing, I was scratching my head over the reason for the nulls. In the CAD drawing (**Fig. 3**), I could not find a length corresponding to a half-wavelength at the null frequencies shown in **Fig. 4**.

Because this horn covered midbass frequencies, my plan was to use double stack cabinets to cover below 200Hz, and the midbass will mutual couple once the horns are stacked. **Figure 5** shows the double stack horn frequency response. Considering the bass begins somewhere below 200Hz, the horns are part midbass and part midrange. Two disappointments turned out to be that the upper rolloff starts too low, not meeting the 1kHz requirement, and the nulls occurring at 600Hz and 1kHz, which I mentioned.

The problem, clearly, is the horn design

does not couple the frequencies above 700Hz to allow horn loading. Above 700Hz the response behaves like a direct radiator but has increased efficiency due to directivity gain. Notice in my design calculator (**Fig. 1**) the throat (St) require-

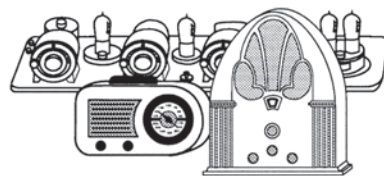
ment is 7.94 in² (51.2cm²). The ratio of throat area to the woofer diaphragm, St/Sd, is 0.0966, which is too small because it's impractical to construct and position over the woofer's curved cone surface. My compromise shown in the CAD drawing

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(Fig. 3) is a throat size of 5×2.75 , i.e., 13.75 in^2 (88.71 cm^2), a St/Sd ratio of 0.1673. I'll explain later how I arrived at this value.

The throat size has a relation to the horn's bandwidth. Wide-frequency or high-frequency bands require a small throat size. Manufacturers of metal or hard plastic midrange horns sometimes use a phase plug to couple high frequencies into the horn's throat entry. The phase plug mounts a few mm from the driver cone and is meant to allow, ideally, an identical path length to the throat at all points from the cone. Inadvertently, my design includes a volume of air just in front of the cone, essentially a 700Hz low-pass filter.

The front volume between the woofer cone and throat is determined by two issues. The cone surface is not flat but curved, and I include a $15/32$ " lip; i.e., the thickness of the wood material, around the woofer frame where it mounts to accommodate the cone's excursion of 5mm. Study Fig. 3 where the woofer bolts to the horn to visualize the issue regarding the front volume.

As a side note to the speaker system, the double stacked horns are crossed over electronically to mate with a 2" commercial compression horn with a sensitivity rating of 110dB for 1W input. The double stack 12" horn SPL required attenuation to match the 110dB, 1W sensitivity!

I should mention that the horn expansion

in the CAD design (Fig. 3) is tractrix³ and my calculator refers to hyperbolic. At a mouth size of $1/8$ wavelength the tractrix expansion reaches the $1/8$ space mouth size at a linear distance of 6" (Fig. 6). Due to the tractrix equation, the horn mouth is calculated at $x = 0$. Therefore, Fig. 6 plot shows the horn flipped on its Z-axis compared to the Fig. 2 plot mentioned above.

Figure 7 shows the tractrix horn taken out to full size. Notice the linear length is 34" and the corresponding tractrix curve length is not shown but hidden (horizontal scroll) inside the expansion table. I arbitrarily chose a throat size of 13.75 in^2 (88.7 cm^2) and recalculated the tractrix curve to get a linear distance of 21.4375" at a curve length meeting 24". Essentially, I take the quarter wavelength hyperbolic linear length (24") and draw the tractrix curve until its curve length reaches 24. My chosen throat area is close enough to 7.93 in^2 (St) throat calculation.

To draw the tractrix points including all my drawings, I use DeltaCAD drawing software to plot the tractrix lines (Fig. 3). For a small fee, DeltaCAD is available from their website, www.deltacad.com. DeltaCAD has a BASIC language script to automate some of its drawing parameters. I set the script to draw the tractrix radius points using circles the thickness of the wood material. When I build the horn, the tractrix curve is printed out on paper at the real size for tracing.

The tractrix curve length taken out to

the hyperbolic length and using double stack cabinets for midbass appears to load the 12" driver appropriately from the impedance plot of two stacked horns (Fig. 8). The impedance phase huddles around the 0° line above roughly 130Hz out to 1kHz.

THE MIDRANGE HORN PROJECT

I went back to my horn calculator and SoundEasy (Bodzio Software) to redesign a midrange horn to meet the 200Hz to 1kHz frequency band. The first step is to find a driver that has a lower cone mass than the JBL 2020H. The driver should also match or exceed the magnetic field strength of the JBL 2020H.

I compared two candidates against the 2020H using the magnetic field strength (BL) in units of T/m and the mechanical mass of the diaphragm (Mms) in gram units. Divide these two numbers to get the force/weight unit. A comparison of the three drivers is shown in Table 1.

I chose the BC 8PE21 for the project and thought that finally a manufacturer had produced a driver specifically for midrange horns. The BC magnetic field strength falls between the JBL 12" and 10" driver. Its cone mass, though, is 18 grams, which gives it a force/weight ratio of 922! That value is almost twice the values of the JBL drivers. The BC 8PE21 driver also has a high power rating of 200W RMS, which is similar to the JBL's ratings.

For this 8" horn design, I flush-mount-

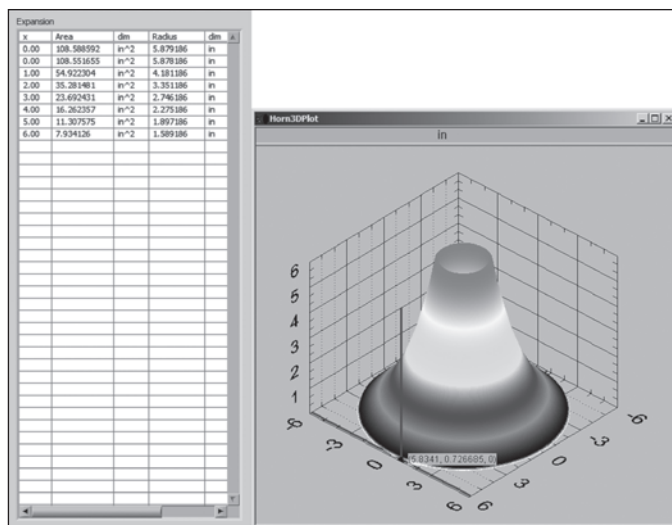


FIGURE 6: The $1/8$ space tractrix expansion for an equivalent mouth size of the hyperbolic expansion. The horn length is only 6" and does not meet the 24" for the hyperbolic design.

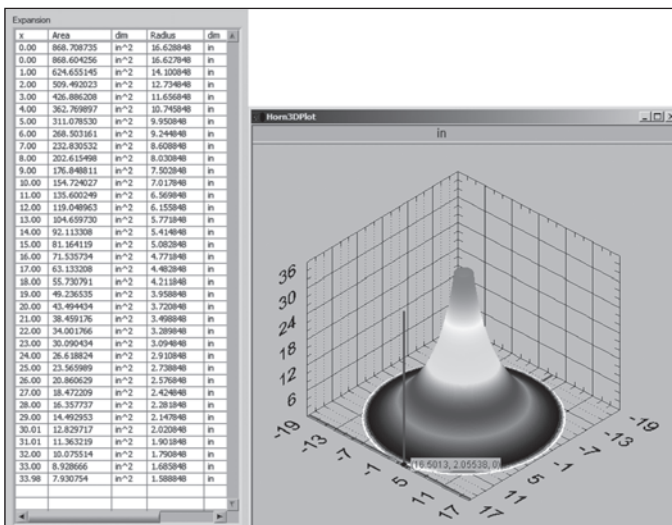


FIGURE 7: The full space tractrix horn expansion for a mouth size that is equivalent or exceeds the hyperbolic horn length. The tractrix length is 34", so I need only 24" to meet the hyperbolic length.

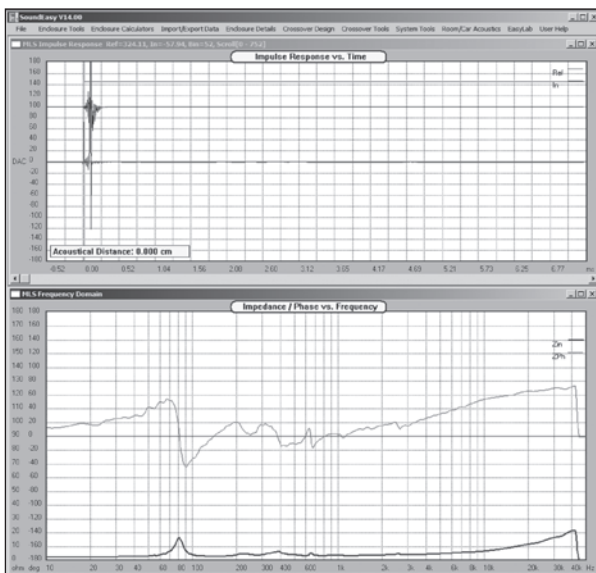


FIGURE 8: Impedance plot of double stack tractrix horn. Lower line is impedance magnitude. The upper line is impedance phase.

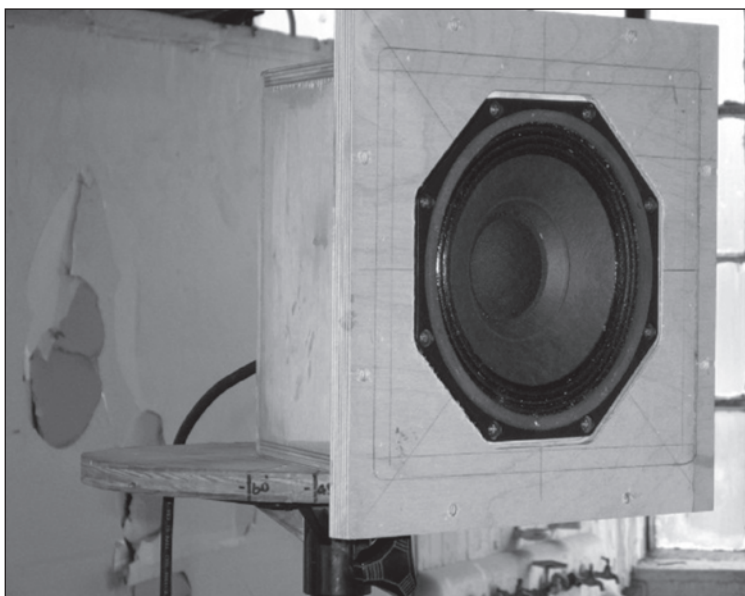


PHOTO 1: Sealed rear chamber and BC 8PE21 200mm (8") driver of the horn shown in the CAD drawing of Fig. 11.

ed the driver to the rear chamber to eliminate any extra front volume acting as a low-pass filter. This is relevant because the design is intended for midrange and not midbass; besides, its maximum excursion is only 1mm. Because this driver is small—200mm (8")—the front volume will turn out quite low.

I started the design by entering the Thiele/Small data into my horn calculator—I chose a bandwidth starting from 170Hz to 1kHz (Fig. 9). I then took the calculated data and entered it into SoundEasy to plot the frequency response and impedance response simulation (Fig. 10).

Armed with this data, I went forward with the design and drew my CAD drawing of the BC 8PE21 midrange horn. In Fig. 11, you see the design is similar, and I've flush-mounted the driver to the rear sealed chamber. Photo 1 shows the finished rear chamber with the driver mounted as in the drawing. The rear chamber with the driver mounts to the horn throat.

Now I'll take some time to go over some points of the horn construction. I first read about bending wood from the personal website of Mark Winquist long before this writing. To bend wood, cut a side of wood grain with slits spaced incrementally apart. The depth of the cut should leave the last ply of wood grain in place, typically 1/8" thickness. Most important, when you cut the slits the wood grain should align parallel to the path of

the horn; if not, the bending will cause the wood to break instead of bend. Photo 2 shows the horn assembly clamped with threaded rod on a jig to hold everything in place. Photo 3 shows the horn removed from the jig before assembling any additional wood pieces.

The top and bottom pieces are cut using a band or jig saw at a 90° angle to the wood. I then used an oscillating orbital spindle tabletop sander with a tilting table to sand out the tractrix curve at an angle of 14½° approximating the CAD tractrix curve by hand. I then tacked the

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top and bottom pieces to an angled piece of wood and set the assembly in the jig for clamping using threaded rod.

MEASUREMENTS

After assembly, I measured the finished horn to access my design. The frequency response is measured at a distance of 91.375cm (36") from the cone. The horn occupies almost 22"; thus the microphone is just over 14" from the mouth of the horn. **Figure 12** shows the frequency response results. The bandwidth appears to meet the 200Hz to 1kHz target bandwidth in the design. The null centered at 800Hz needed investigation, and I was

also concerned about the second null located at 2466Hz.

I decided to find out whether the null at 800Hz is due to either the horn path or the horn's rear sealed chamber. Because I could just detach the horn's rear chamber with the driver mounted to the rear, I took the frequency response of the BC 8PE21 driver in its rear chamber and compared the response. In **Fig. 13**, I compare the frequency response of the sealed box by itself versus its horn. I also took a close mike measurement and plotted this shifted -30dB in order to show smoothness in the measurement.

From **Fig. 13** it's clear the null at 800Hz

is due to the rear sealed chamber! The drawing of the horn (**Fig. 11**) shows a rear chamber with a constant height of 9" in the vertical plane. If I take the speed of sound as 13560 inches/sec, the half-wavelength corresponding to 800Hz frequency is 8.475", i.e., just under the chamber's 9" height, alas! I can apply the same principle to the JBL 2020H horn I previously built. The JBL tractrix has a constant rear chamber height of 12.9375". A 1kHz null frequency wavelength is 13.56", just over the rear chamber's 12.9375" height.

I'm not an acoustician, but this tells me the nulls are due to the box modal resonance of the six sides of the rear sealed

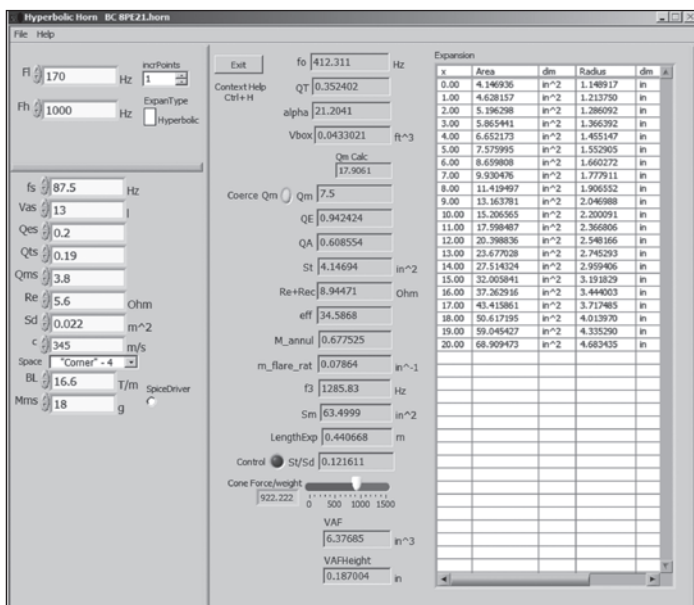


FIGURE 9: Thiele/Small data and Leach design parameters for the BC 8PE21 200mm (8") driver. The chosen bandwidth is 170Hz to 1kHz. The required horn length is 50.8cm (20").

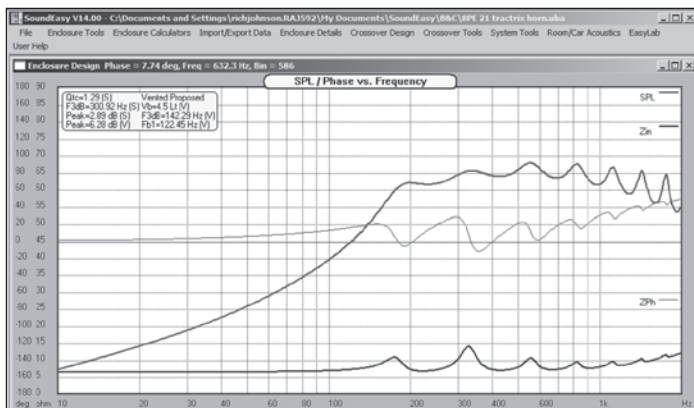


FIGURE 10: SoundEasy plots the horn data from the horn parameters in **Fig. 9**. At the 20Hz vertical line, the bottom plot is impedance magnitude, the top line is impedance phase, and the middle line is the frequency response.

TABLE 1 A comparison of magnetic strength divided by cone mass for three candidate horn drivers.

	JBL 2020H	JBL 1212H	BC 8PE21
Cone force/weight	416	517	922
BL	18.3T/m	15.5T/m	16.6T/m
Mms	44g	30g	18g

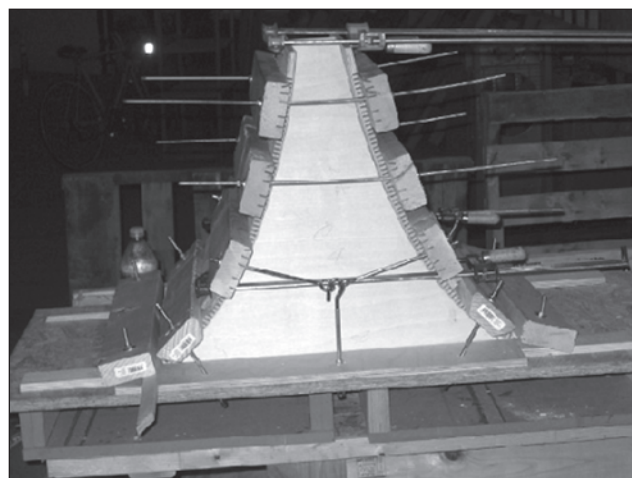


PHOTO 2: The BC 8PE21 200mm (8") tractrix horn clamped to the threaded rod jig.

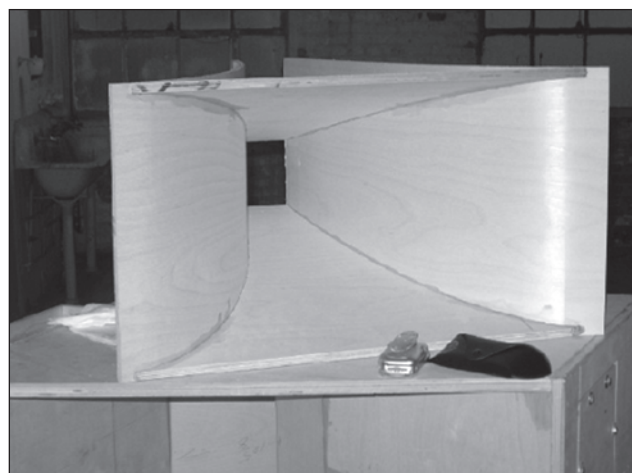


PHOTO 3: Horn removed from the jig prior to any wood assembly.

chamber! Because the rear chamber top and bottom pieces are parallel, the standing wave pattern dominates the enclosure modes⁴. Hmm, maybe I should have stuck with one of the golden ratios of 1.59, 1.26, and 1 or similar for the rear chamber.

Next, the second null, located at 2466Hz, comes from the horn path because the null does not appear in the frequency plot of the rear sealed chamber. I traced my steps back to when I smoothed the tractrix horn curve by hand for the top and bottom piece of the horn duct using an oscillating orbital sander. It's very difficult to trace the curve accurately by hand. Most will not get the curve exactly right doing it by hand.

Study both CAD drawings of Figs. 3 and 11, and you'll notice the

tractrix angle adjacent to the horn wall is small near the throat then becomes 90° at the mouth. If I were to do this again, I would use a straight edge near the throat for tracing instead of the tractrix line to avoid parallel surfaces during construction. If you're equipped to trace the tractrix line perfectly, then by all means do so.

I decided to resolve the 1kHz null fre-

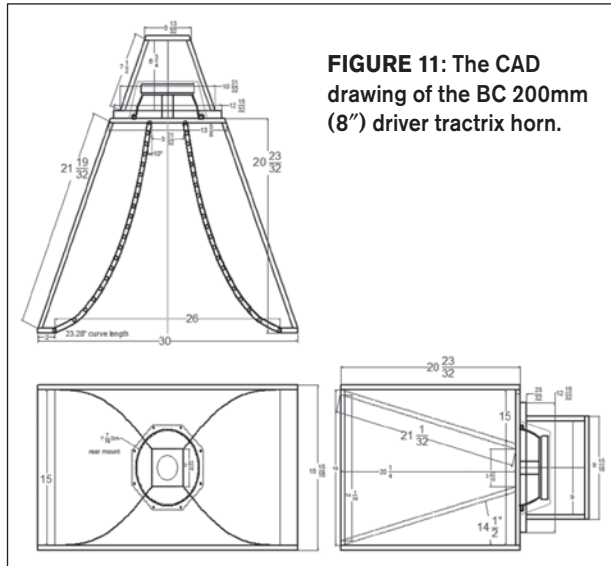


FIGURE 11: The CAD drawing of the BC 200mm (8") driver tractrix horn.

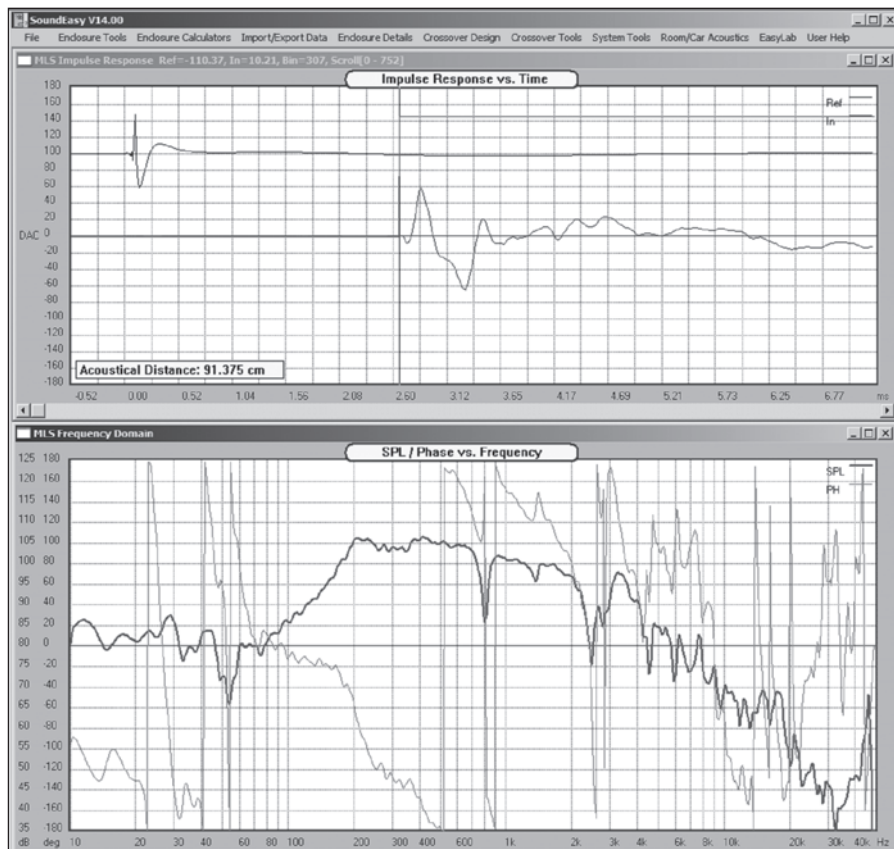


FIGURE 12: Frequency response of BC 200mm (8") tractrix horn shown in the CAD drawing in Fig. 11. At 10Hz line, the top line is frequency and the bottom line is phase.

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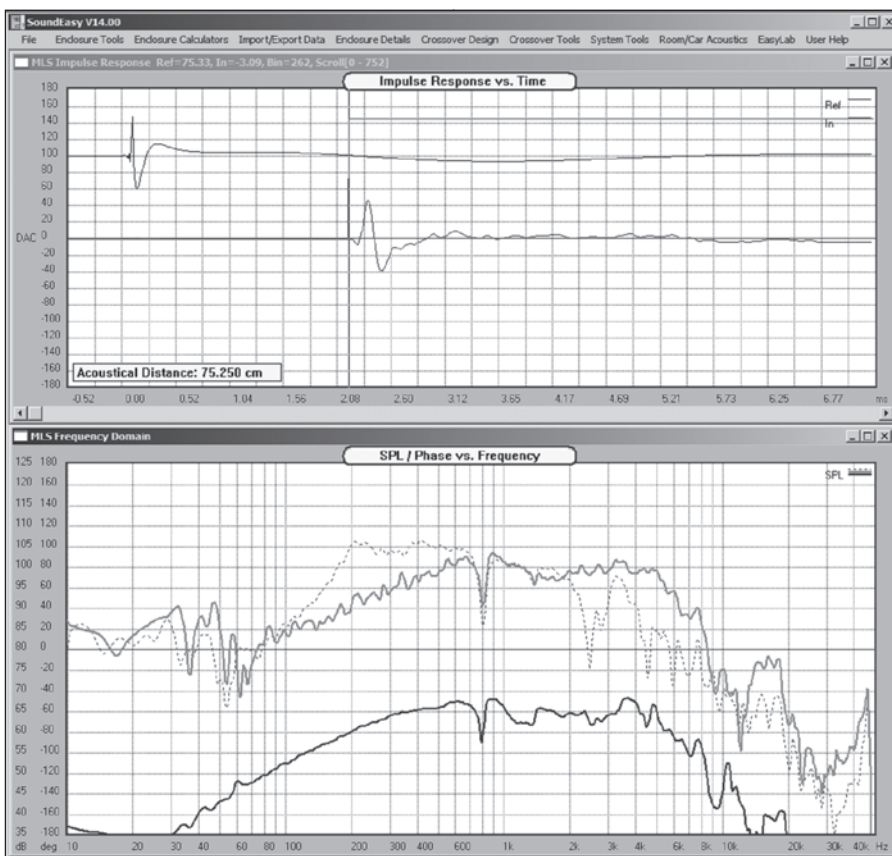


FIGURE 13: The frequency response of the BC 200mm (8") driver in its sealed back chamber without the horn duct attached. The original horn response plot is the dotted line. At 200Hz the top line is the horn's response; the bottom line is the close mike response of the driver in its sealed box. The middle line is the frequency response of the rear sealed chamber at 75.250cm (29.6").

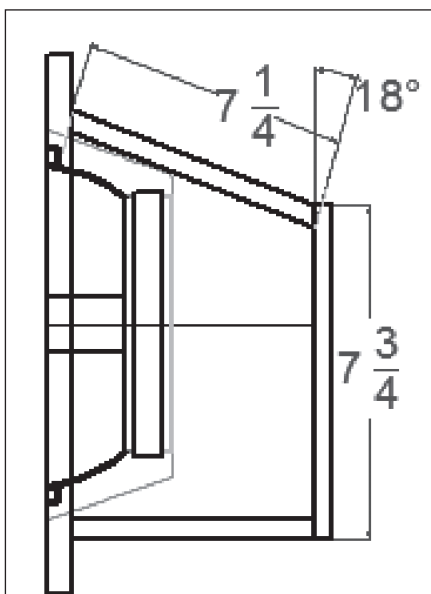


FIGURE 14: Side view of the new rear sealed chamber to replace the one in the drawing of Fig. 11. The top view is unchanged. The 18° cut is really a compound angle of 18° in one plane and 6° in the other.

quency by making a CAD drawing change and rebuilding the rear sealed chamber (Fig. 14). I modified the rear chamber so that the top piece now angles at 18°. This should reduce the 800Hz null in the original frequency response mentioned above, although I have no data to prove this yet. The reduction does not violate the rear back volume because the original calculation is 0.0433ft³, or 75 in³ (1.23 liters). The volume I built originally is 435.735 in³ (7.14 liters). The volume sizes I mention don't account for the driver's volume occupied in the rear chamber. The occupied volume will take the rear volume lower, approaching the required 1.23L. Envision a liter of carbonated soda—that's my required volume.

I don't want to reduce the depth of the rear box because the driver is vented and air needs to circulate behind the driver's vent hole, which is located on the woofer's magnet at the center. Leaving space behind the magnet vent is critical in bass horns because the cone movement is greater compared to midrange horns. It could mean the difference of a burned-out coil should your woofer need to handle high power. Nevertheless, leave space for the magnet to cool when building the rear chamber.

From studying Fig. 13, it's clear the

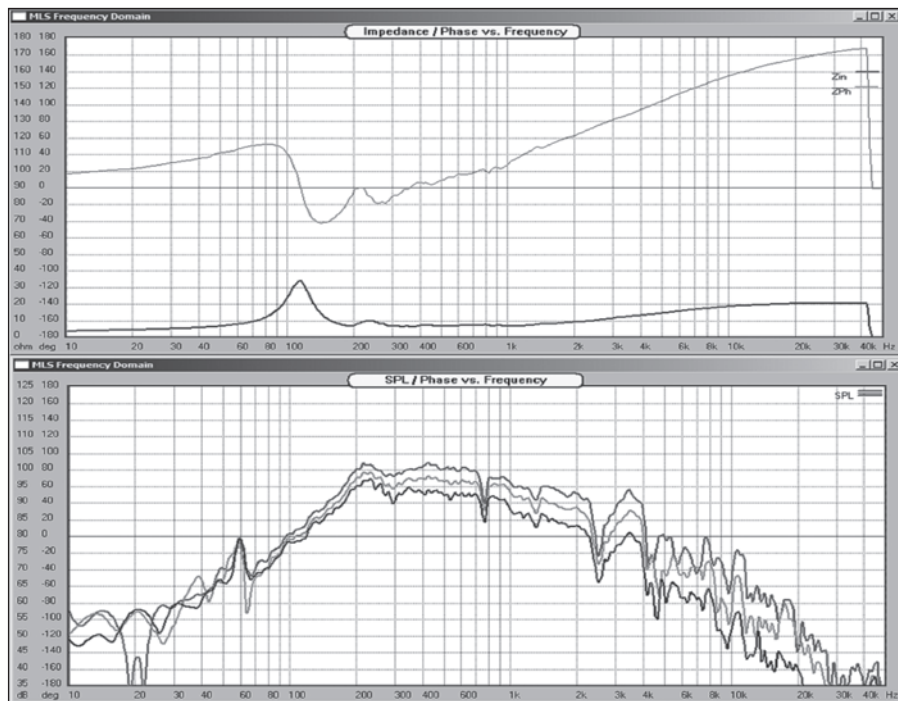


FIGURE 15: Impedance and frequency response of the BC 200mm (8") tractrix horn. Top plot is impedance, the lower line is impedance magnitude, the top line is phase. The bottom plot is frequency response at 0, 30, and 45° horizontal.

horn loading occurs from 200 to almost 1kHz. Beyond 1kHz the frequencies are not raised in SPL as the 200-1kHz horn passband shows. The driver provides some directivity increase above 1kHz, which is a good trade-off because it comes close to matching the SPL of the horn's passband. **Figure 15** shows the impedance response and frequency response at 1m distance for three angles.

Notice the impedance reactance is cancelled by the rear chamber because the impedance phase at 200Hz is near the 0° line. The line clearly shows a phase change in slope at 200Hz as a result of the sealed back chamber. The inductive rise with increasing frequency in the impedance phase plot is due to the driver voice coil inductance.

The frequency response at three angles confirms I have very good dispersion up

to 1kHz. Ignore the 60Hz spike, which is due to AC power hum from the use of an external soundcard. The persistent hump just above 200Hz of the three angles is probably due to the proximity effect of the test microphone located too close to the horn cabinet edge. In the far field the hump just above 200Hz should not persist.

I intended to mate the tractrix horn with a 2" compression horn driver that is crossed over using an electronic filter near 1kHz. Thus, a LPF will attenuate the BC 8PE21 tractrix horn beyond 1kHz. With this in mind, I decided there's no need to correct the 2466Hz null, leaving the horn as it is.

All that was left was constructing a new rear sealed chamber. I discovered, to my surprise, the new 18° angle of the top piece required a compound angle cut for

the top to properly fit the angled and tapered sides. My CAD drawing is 2D. A 3D CAD tool is needed to discern the angle. Luckily, after some trial and error, that angle turned out to be 6° tilt. **Photos 4** and **5** show the pieces just prior to construction and after I made the cut.

Before the glue had time to dry, I set the horn on a speaker stand to take measurements. **Photo 6** is the pre-finished new rear sealed chamber attached to the horn.

Figure 16 shows that there is some reduction in the null at 800Hz, but it's not completely gone. The dotted line is the original frequency response shown for comparison. Although five sides of the rear chamber were now unparallel, the null was not completely gone!

SoundEasy can

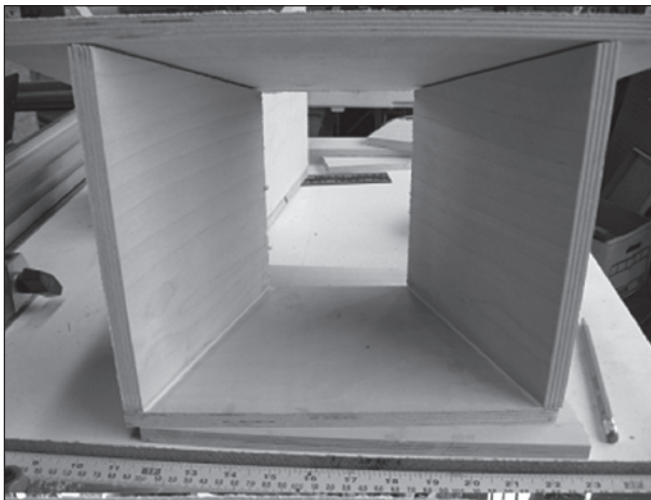


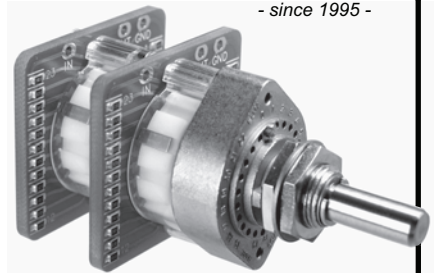
PHOTO 4: New chamber construction before attaching the top and before compound angle (6°) cut.



PHOTO 5: After the cut, the top piece now fits tight to the side pieces.

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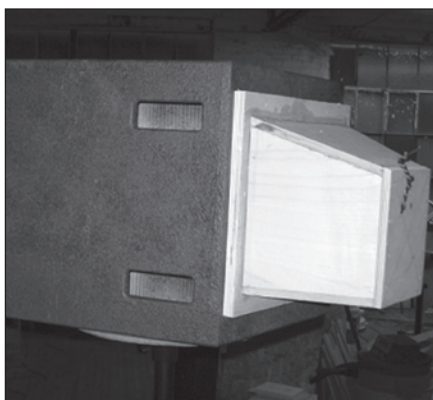


PHOTO 6: The new rear sealed chamber is stapled to the horn throat for initial frequency response test.

model an enclosure's modes using Finite Element Method (FEM), but at this stage I never simulated any sealed chamber boxes using SoundEasy FEM analysis. I don't think acoustic stuffing would remedy this because the volume is quite small. If I were to do this again, seeking perfection, I would test the rear chamber for response irregularities before moving on to a final design.

Thus far, all frequency response measurements posted are performed at a 1m distance or less. The horn occupies much of this measurement distance. My point is that a 1m distance is an unrealistic listening position and what's needed is a far field measurement. In a multi-way speaker or horn system, the drivers combine in some coherent manner at the listening position.

I repositioned the horn and test microphone to an over-the-head vertical position (~9') in a huge room to minimize floor reflections. I moved the microphone distance to 8'. **Figure 17** shows the far field frequency response including three horizontal angles—0, 30, and 45. Once again, the spike at 60Hz hum is due to AC hum, so you should ignore it.

Now, after the results of **Fig. 17**, I can say the operating frequency range of the BC 8PE21 tractrix horn is 200Hz to 2k! If the horn's tractrix path is drawn with greater accuracy, as I explained in the middle of this article, a 3k upper limit is surely within reach.

To sum things up, I have shown that a wideband cone midrange requires a low cone mass driver. The horn driver, without use of a phase plug, requires coupling to the horn throat for horn loading; other-

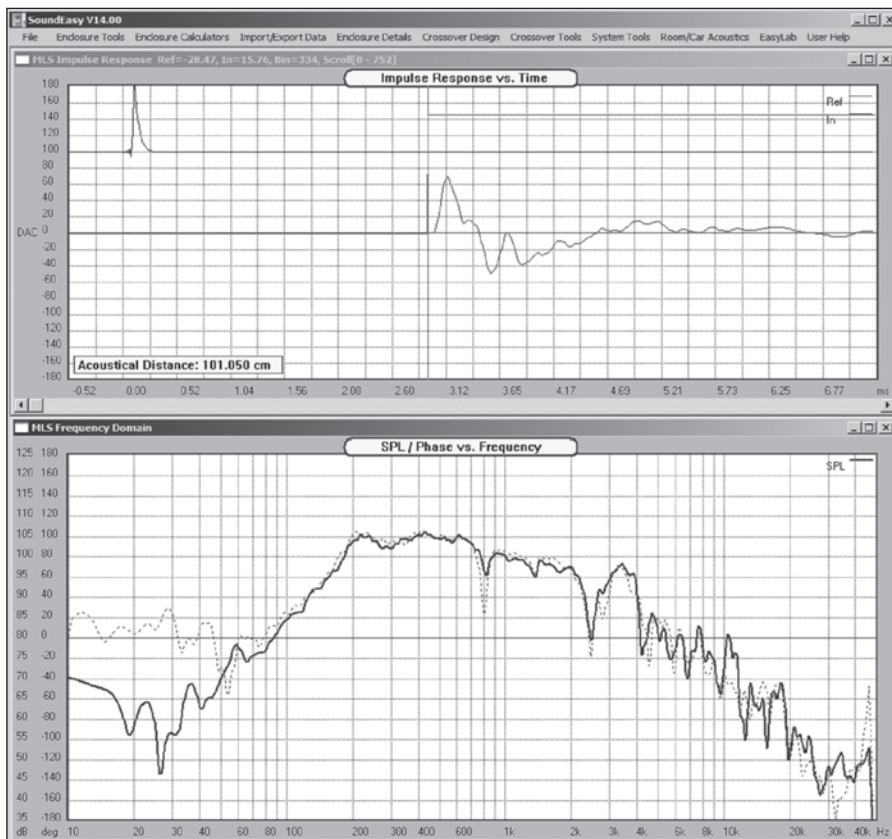


FIGURE 16: Frequency response of the new rear sealed chamber attached to the horn duct as in *Photo 6*. Solid line is frequency of new sealed chamber; dotted line is original's frequency response. The 800Hz null is suppressed.

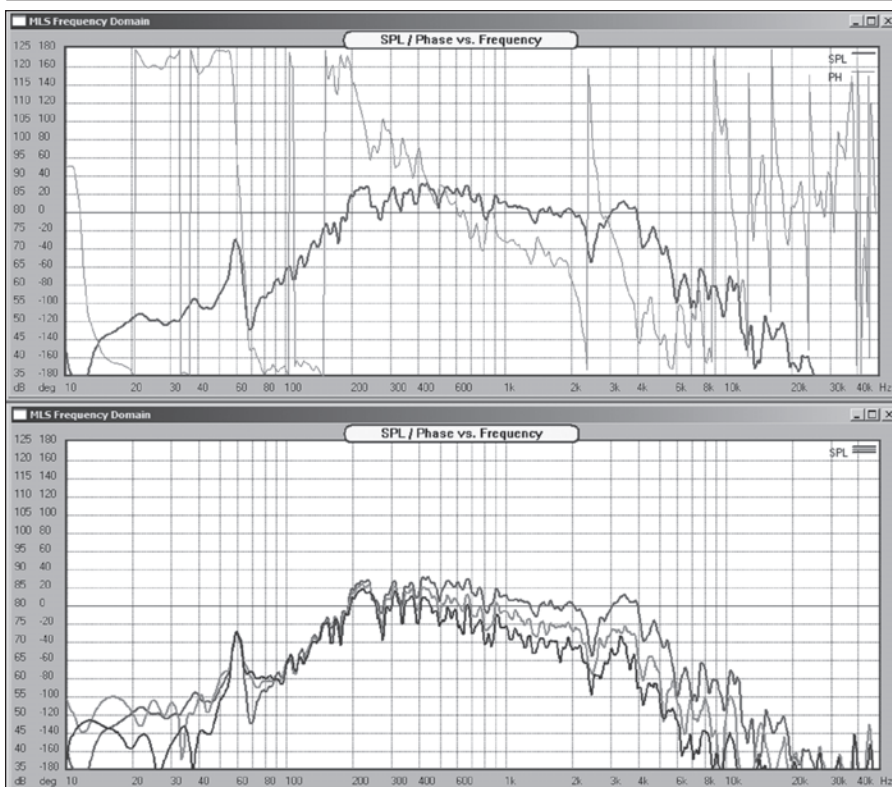


FIGURE 17: Far-field frequency response of the BC 8PE21 tractrix horn. The top plot is response measured at 8'. Between 200Hz to 2kHz the frequency magnitude hovers near the 0dB centerline. The bottom plot is the 8' frequency response at three angles—0, 30, and 45°. A 60Hz AC hum artifact exists, so you should ignore the spike at 60Hz.

wise, a low-pass filter of the front volume may be in effect. You may use a tractrix curve length of the same linear length as hyperbolic for improved off-axis response.

When possible, simulate or measure your driver in its rear chamber before selecting the rear chamber shape and size. For the tractrix curve to be effective, its mouth should extend the 90° bend a few inches before the horn mouth ends. You must avoid parallel surfaces in the horn duct, or else corresponding nulls will appear in the frequency band. Include far-field measurements when accessing mid-range horn frequency response.

POSTSCRIPT

I mentioned above that SoundEasy is capable of simulating the rear chamber box response using FEM analysis. Up to now, I never used the Delauney FEM analysis in SoundEasy, so I would now learn. If the analysis can predict the nulls in the mid-range band, I could change the rear chamber design prior to any construction!

Keep in mind I'm a novice at the FEM analysis, and to really use the feature the SoundEasy manual should be referenced. In a nutshell, here's my SoundEasy (SE), version 14, Delauney FEM procedure. One plane of the rear chamber's internal dimension is laid out on an XY grid in cm units. The total volume is set via the grid size of X, Y, and Z parameters.

The software calculates a mesh of nodes using nine layers in the Z-axis. All nodes are now labelled with mesh node numbers. The position of the driver is set using mesh node numbers. Either four or nine nodes are used to set the driver position; i.e., points on the driver frame where it bolts to the enclosure.

SE calculates the frequency response using Delauney FEM analysis. The process is CPU intensive, so a single frequency response run could take five minutes or more. Calculating just the enclosure nodes, though, is quick.

Figure 18 is the simulated frequency response of SoundEasy Delauney FEM analysis using the original rear chamber of the BC 8PE21 200mm (8") tractrix horn. **Figure 19** shows parameters used to generate the Delauney FEM frequency response. The analysis shows the infamous 800Hz null that appeared in my finished horn measurement! Wow, I could have prevented the null if armed with this in-

formation ahead of time.

To summarize, a midrange horn's rear chamber usually requires a small volume where the internal dimension could result in nulls located in the horn's passband. It is a wise step to first simulate before construction and test for nulls in the passband. I assume that using golden ratios for the internal dimensions of rear chamber is critical for smooth mid-range. The issue also exists when designing basshorn, but more often, the null(s) fall outside the passband of the

bass frequencies.

With that said, I can now explain the cause for nulls outside the passband in my first article, "An Excellent DIY BassHorn," December 2007 *audioXpress*. In the article the "huge null located between 500Hz–600Hz" is a result of the basshorn's rear chamber mode behavior as I have dealt with this issue in this article for mid-range horns. I hope DIYers will use this information to build better cone mid-range horns and avoid the design issues described in this writing. *aX*

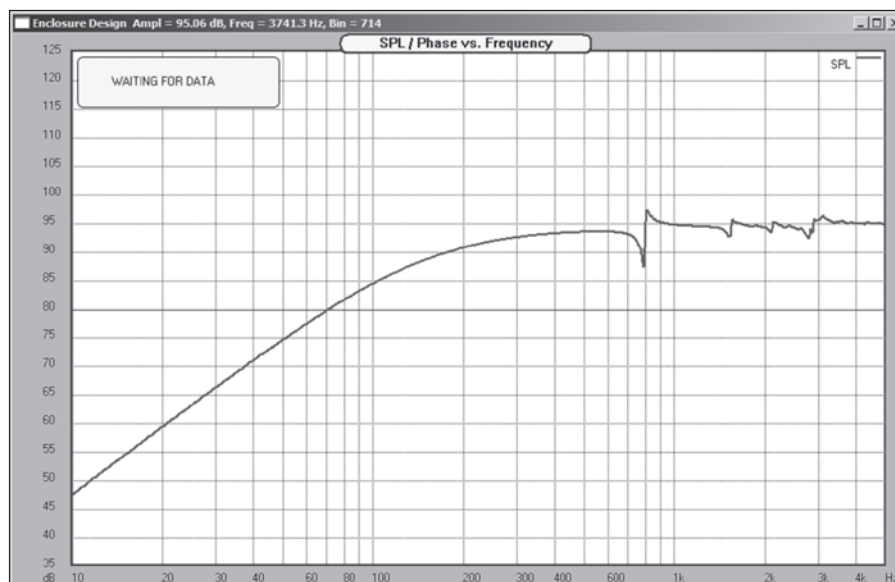


FIGURE 18: SoundEasy Delauney FEM analysis frequency response of the original rear chamber of the BC 8PE21 tractrix horn.

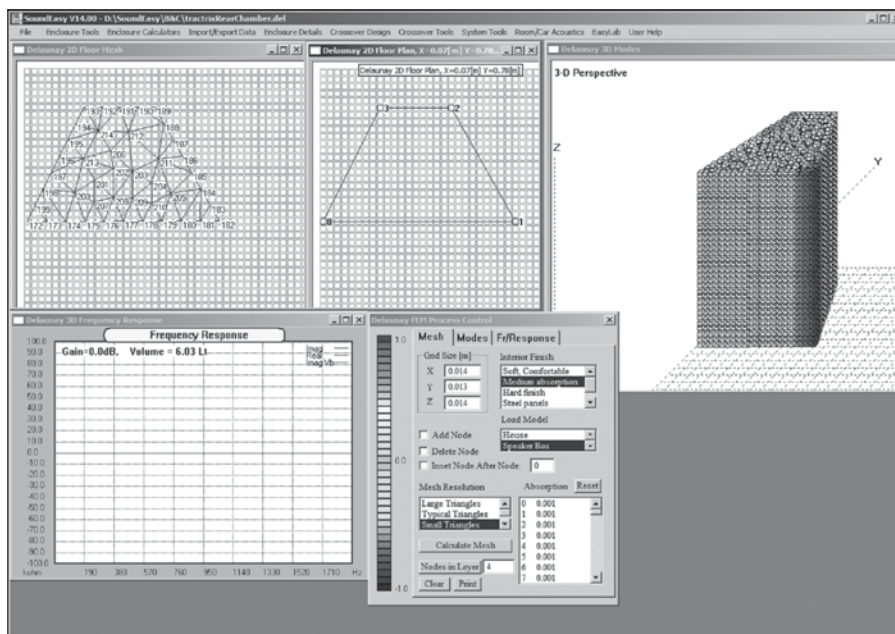


FIGURE 19: SoundEasy Delauney FEM window. The top view of the rear chamber is drawn using node number counterclockwise. X, Y and Z units set the internal volume.

Building the Occam Audio MTM 4 Kit

By Bill Fitzmaurice



PHOTO 1: The Occam Audio MTM 4.

The Occam Audio MTM 4 kit uses two CSS WR 125ST 16Ω shielded 4" mid-basses and a Wavecor TW030WA01 8Ω 30mm soft dome tweeter in a 10 liter aperiodic vented enclosure (Photo 1). Versions are available configured for bookshelf, center channel, and wall mount/surround. I built a pair of the bookshelf models.

ASSEMBLY

The kit, which includes an instructional DVD, is shipped with the parts nested inside the plastic wrapped enclosure. The magnetically attached grille hides four screws that hold the cabinet together. Removing these screws with the enclosed Allen wrench reveals polyester batting and a box of parts (Photo 2). The 1" thick CNC machined MDF enclosure consists

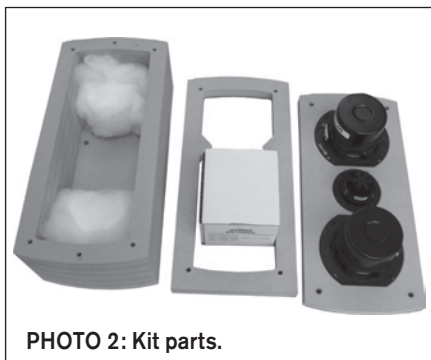


PHOTO 2: Kit parts.

of a rear shell, a spacer that fits between the shell and the baffle, and the baffle.

A variety of finish options range from veneer to paint. The DVD offers tips on finishing, as well as a step-by-step video on the assembly process. Using the DVD instructions, I applied a paper-backed red oak veneer, finished with sprayed-on shellac base coats and clear lacquer top coats. The result is a very attractive speaker with a high WAF. That was very important, because their destination was her den.

Inside the parts box are screws and other hardware, a crossover board, crossover components, lead wires with spade lugs on one end, an input cup, and the aperiodic vent. You can either attach the crossover parts to the board with the supplied plastic ties, or glue them in place with a high temperature hot-melt glue or RTV adhesive. Because the board is not a PC, you directly connect and solder the component leads. When you complete the crossover, mount it via screws and spacers to pre-drilled pilot holes inside the shell (Photo 3).

The crossover input leads attach to the input cup, which you install in a pre-drilled hole on the cabinet back. The cup has push terminals that you can use with bare wire or with a banana plug. The aperiodic vent, consisting of a fiberglass filter and plastic escutcheon, screws

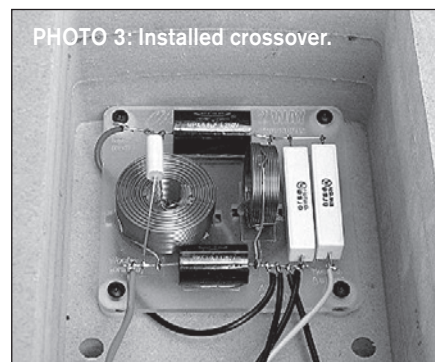


PHOTO 3: Installed crossover.

in place over another pre-drilled hole (Photo 4). You attach the drivers to the baffle, connect the wires, and screw the baffle, spacer, and cabinet shell together again. While purists may leave the grille off, I like the look and the driver protection it offers (Photo 5). Assembly time,



PHOTO 4: Installed input cup and aperiodic vent.



PHOTO 5: Completed cab with grille.

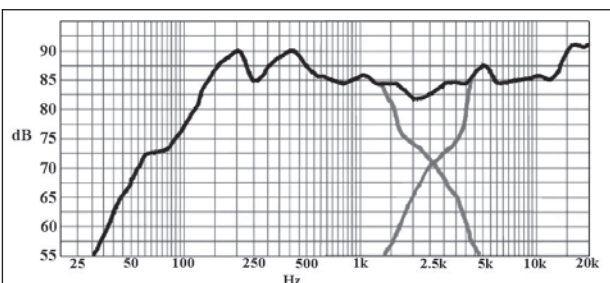


FIGURE 1: SPL 1m/1W.

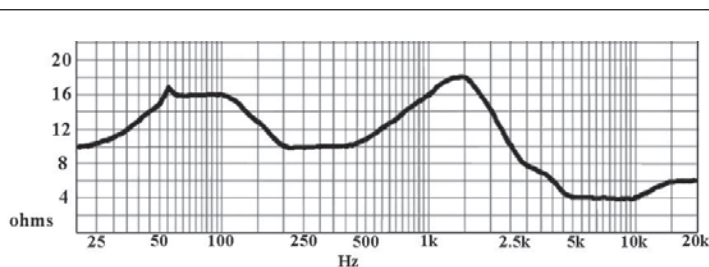


FIGURE 2: Impedance.

exclusive of applying the finish, is less than an hour.

This is an ideal kit for the first-time builder or for those with no access to a shop, because there is no woodworking required. The only reservation you might have is that there is no provision for sealing the mid-bass drivers to the baffle, or sealing the joints between the cabinet parts. Of course, that's easily addressed with some sealing caulk should you choose to do so. Because the aperiodic alignment acts as a lossy sealed box anyway, I doubt that sealing everything airtight would significantly affect performance.

PERFORMANCE REVIEW

The 16 Ω CSS WR 125ST manufacturer's recommendation is a 7 liter bass reflex alignment, with a nominal f_3 of 90Hz. A pair of them in the 10 liter Occam has an f_3 closer to 140Hz (Fig. 1). The impedance of the aperiodic alignment is unusual (Fig. 2), lacking a low frequency peak. It's possible that the generous amount of cabinet stuffing is responsible for this.

The lack of any mid-bass boom indicates a moderate Q , also the product of a well-damped enclosure. As a result of the rather high f_3 , this isn't a cab that you'd want to use for high output HT mains, but with the limited output capabilities of a pair of 4" drivers that's not a use you should be considering anyway.

Where the fours really shine is in off-axis response (Fig. 3). Measured at 30° off-axis, their response is essentially the same as on-axis to 900Hz, and never drops by more than 2dB above that. The neo magnet Wavecor tweeter does very well up to 10kHz, above that dropping to a still respectable -6dB off-axis. The Wavecor has an on-axis response bump above 12.5kHz, resulting in a nice off-axis response, though it only benefits those of us still young enough to

hear it. The superior off-axis response makes this box ideally suited for HT surround duty.

The close center-to-center spacing of the midbasses allows it reasonable horizontal dispersion as a horizontally placed center channel. With music input, they deliver a very smooth response, with pinpoint imaging. The sound quality was equal to that of the MTM section of my reference David floor-stander, albeit about 10dB lower in sensitivity. Those interested in an easy listening bookshelf style speaker for stereo will find these a serious upgrade to the typical offerings of the big-boxes.

The crossover is a 2nd-order/2nd-order, with a nominal 4kHz corner frequency (Fig. 4). A pair of resistors pad the tweeter down to match the sensitivity of the midbasses. Typically, MTMs use 8 Ω midbasses wired parallel, to overcome the baffle step and match

the sensitivity of the tweeter. The resulting 4 Ω nominal load can be problematic, especially for inexpensive HT receivers that are often rated for 6 Ω operation. The 8 Ω load of the Occam is very friendly for those receivers, whose owners would seem to be the perfect target demographic for this cab.

MEASUREMENT PROTOCOL

I took measurements outdoors, at 1m, unweighted, with a 2.83V swept sine input. I averaged ground plane and cabinet center axis measurements to approximate half space anechoic conditions. I made measurements with the grille in place. *ax*

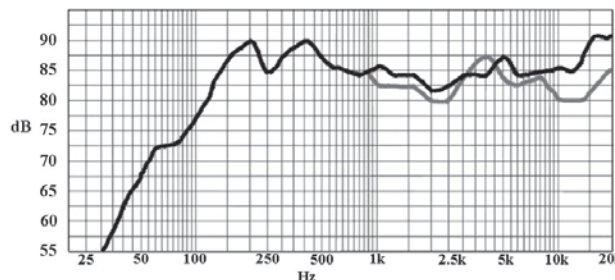


FIGURE 3: SPL, on-axis and 30° off-axis.

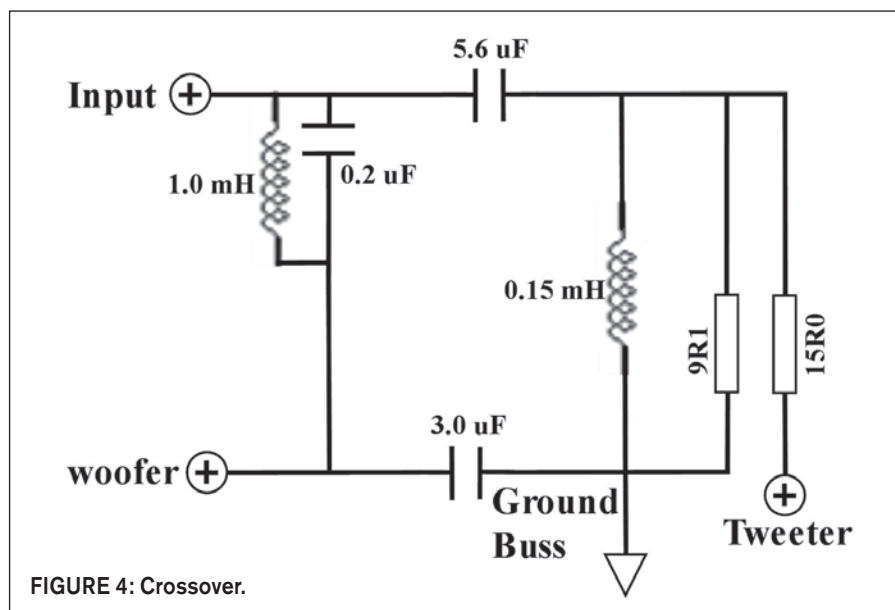


FIGURE 4: Crossover.

Vg1 Correlations

Next time you design a power amp, try this mathematical model to predict the required bias voltage for your pentode or tetrode output tubes.

Words have different colorations for different people. We often associate a word with a remembrance from our lives—be it a person, a place, an emotion, or a piece of music. Experience is what gives a word personal meaning.

It's difficult to imagine something other than rationality when you think about the word *correlation*. From a mathematical perspective, it means the degree of relative correspondence between two or more sets of variables. But it also reminds me of one of the courses I enjoyed the most while I was studying finance at the University.

To put this into context, I was taking my final course of finance, the most advanced and difficult one, and half the session was dedicated to a very singular task: To find a correlation that would describe the evolution of the stock market.

I will always remember how Professor Farrell answered the legitimate question a student asked: "Sir, how are we going to find such a correlation?" He drew a huge spiral on the blackboard, starting from the outside and terminating in the center. Nobody understood what he was doing until he looked at us and said, "Search." That was his last word on the matter until we revealed the outcome of our work in front of the class at the end of the session. That was it, no other guidelines.

The metaphor of the spiral speaks for itself. When you start searching for something, you don't know in which direction to go. But the more you search, the more you learn and the more you

understand your subject. From one idea to another, you get closer and closer to the objective and, sometimes, as in the case in this article, you are fortunate enough to reach the center of the spiral.

THE PROBLEM

Have you ever found yourself in a situation where you wanted to design a power amp with a pentode/tetrode tube, operating in the triode mode or the ultralinear mode, but did not know the voltage bias required beforehand? The reason for this is quite simple: Most manufacturers publish the plate characteristics for only the pentode/tetrode mode.

This situation bothered me because it deprived me of the possibility of designing and optimizing a driver stage on paper before I committed to building the amplifier. I wondered whether it would be possible to predict the value of bias voltage in the triode and the ultralinear mode, from the pentode/tetrode mode plate characteristics. In other words, is there a correlation between Vg1 in the pentode/tetrode mode, and the triode and ultralinear modes?

SEARCHING FOR A CORRELATION

My search for a model was rewarded by the discovery of two very significant correlations that combine in a mathematical equation. The correlations apply to the various power tubes I studied, namely, the EL34, KT66, KT77, KT88, and the 6550A. Odds are good that the correlations also work for other power tubes such as the EL84, the 6550, and the 5V6550C.

FIRST CORRELATION

The first correlation concerns the value of Vg1 in the triode mode when the plate voltage Va is set to 425V. For each type of tube I analyzed, I found that Vg1 varied by approximately 10% of its value at Va = 425V, for each plate voltage variation of ±25V around the median of 425V. **Table 1** shows the results from the datasheets of the power tubes I studied. Note that I limited the scope of the study to high tensions ranging between 350V and 500V, which covers the vast majority of the design operating points for these tubes.

Mathematically, the first correlation may be expressed in the following form:

$$Vg1_T = Vg1_{T425V} \left[1 + \left(\frac{Va - 425}{250} \right) \right] \quad (1)$$

where : **Vg1_T** is the value of Vg1 in the triode mode for a specific plate voltage Va; **Vg1_{T425V}** is the value of Vg1 in the triode mode when Va=425V; **Va** is the value of the plate voltage.

** Note: Only the numeric value of Va is retained in this equation.*

If you have a datasheet that publishes the plate characteristics in the triode mode, then you can predict quite accurately any values of Vg1_T using equation 1. Note that this correlation also applies to the ultralinear mode because Va and Vg2 are at the same potential under DC conditions, just as in the triode mode.

All this was very interesting but not

very useful, because there is no need to calculate the value of V_{g1T} if the information is readily available in the datasheet. What I needed was a way to predict a value of V_{g1T} from the pentode/tetrode mode because this information is always present in all datasheets. The second correlation solved this issue with surprising accuracy.

SECOND CORRELATION

Building on the revelation of the first correlation, my objective was now to find an equation that would make it possible to determine the value of $V_{g1T425V}$ from the pentode/tetrode plate characteristics.

Close inspection of various datasheets revealed general relations between the triode mode and the pentode/tetrode mode at the operating point of $V_a = 425V$. After developing and testing numerous models, I finally reached the center of the spiral with the following equation:

$$V_{g1T425V} = V_{g1P425V} \left(\frac{4 - V_{g2}}{100} \right) \frac{5\sqrt{P}}{19} \quad (2)$$

where : $V_{g1P425V}$ is the value of V_{g1} in the pentode/tetrode mode for $V_a = 425V$; V_{g2} is the value of V_{g2} in the pentode/tetrode mode as indicated in the datasheet; P is the maximum plate dissipation of the tube as indicated in the datasheet.

** Note: Again, only the numeric values of V_{g2} and P are retained in this equation.*

It is important to mention that you must use plate characteristics exhibiting a value of $V_{g2} = 250V$ or $V_{g2} = 300V$. Other values of V_{g2} simply don't work with this model.

Substituting $V_{g1T425V}$ of equation 1 with equation 2 leads to this final mathematical model:

$$V_{g1TU} = \left[V_{g1P425V} \left(\frac{4 - V_{g2}}{100} \right) \frac{5\sqrt{P}}{19} \right] \left[1 + \left(\frac{V_a - 425}{250} \right) \right] \quad (3)$$

Continued on page 38

		Plate Voltages (Va)						
Tube Type		350	375	400	425	450	475	500
1- KT88 Genalex 1959	Vg1	-33	-38	-43	-47	-52	-57	-61
	Variation (V)	-14	-9	-4		5	10	14
	Variation (%)	30%	19%	9%		-11%	-21%	-30%
2- KT88 MOV 1974	Vg1	-32	-36	-41	-46	-50	-55	-58
	Variation (V)	-14	-10	-5		4	9	12
	Variation (%)	30%	22%	11%		-9%	-20%	-26%
3- KT88 Reflector	Vg1	-33	-38	-42	-47	-52	-56	-61
	Variation (V)	-14	-9	-5		5	9	14
	Variation (%)	30%	19%	11%		-11%	-19%	-30%
4- 6550A GE 1972	Vg1	-30	-35	-38	-43	-47	-52	-56
	Variation (V)	-13	-8	-5		4	9	13
	Variation (%)	30%	19%	12%		-9%	-21%	-30%
5- KT77 MOV	Vg1	-20	-22	-25	-28	-31	-34	-37
	Variation (V)	-8	-6	-3		3	6	9
	Variation (%)	29%	21%	11%		-11%	-21%	-32%
6- KT66 GEC	Vg1	-28	-32	-36	-39	-43	-47	-52
	Variation (V)	-11	-7	-3		4	8	13
	Variation (%)	28%	18%	8%		-10%	-21%	-33%
7- KT66 Marconi 1956	Vg1	-28	-32	-36	-39	-43	-47	-51
	Variation (V)	-11	-7	-3		4	8	12
	Variation (%)	28%	18%	8%		-10%	-21%	-31%
8- EL34 Mullard 1964	Vg1	-25	-28	-31	-33	-37	-40	-43
	Variation (V)	-8	-5	-2		4	7	10
	Variation (%)	24%	15%	6%		-12%	-21%	-30%
Average Variation (%)		29%	19%	9%		-10%	-21%	-30%

TABLE 1: First correlation: When V_a varies by increments of $\pm 25V$ around the median of $425V$, V_{g1} varies approximately by 10% per increment.

		Input Parameters			Plate Voltages V_a						
Tube Type		$V_{g1P425V}$	P	V_{g2}	350	375	400	425	450	475	500
1- KT88 Genalex 1959	V_{g1} (data sheet)				-33.0	-38.0	-43.0	-47.0	-52.0	-57.0	-61.0
	$V_{g1_{TW}}$ (model)	-30	35	300	-32.7	-37.4	-42.0	-46.7	-51.4	-56.0	-60.7
	Deviation (V)				-0.3	-0.6	-1.0	-0.3	-0.6	-1.0	-0.3
	Deviation (%)				0.9%	1.7%	2.2%	0.6%	1.2%	1.7%	0.5%
2- KT88 MOV 1974	V_{g1} (data sheet)				-32.0	-36.0	-41.0	-46.0	-50.0	-55.0	-58.0
	$V_{g1_{TW}}$ (model)	-27	42	300	-32.2	-36.8	-41.4	-46.0	-50.7	-55.3	-59.9
	Deviation (V)				0.2	0.8	0.4	0.0	0.7	0.3	1.9
	Deviation (%)				-0.7%	-2.3%	-1.1%	-0.1%	-1.3%	-0.5%	-3.2%
3- KT88 Reflector	V_{g1} (data sheet)				-33.0	-38.0	-42.0	-47.0	-52.0	-56.0	-61.0
	$V_{g1_{TW}}$ (model)	-27	42	300	-32.2	-36.8	-41.4	-46.0	-50.7	-55.3	-59.9
	Deviation (V)				-0.8	-1.2	-0.6	-1.0	-1.3	-0.7	-1.1
	Deviation (%)				2.3%	3.1%	1.3%	2.0%	2.6%	1.3%	1.9%
4- 6550A GE 1972	V_{g1} (data sheet)				-30.0	-35.0	-38.0	-43.0	-47.0	-52.0	-56.0
	$V_{g1_{TW}}$ (model)	-25	42	300	-29.8	-34.1	-38.4	-42.6	-46.9	-51.2	-55.4
	Deviation (V)				-0.2	-0.9	0.4	-0.4	-0.1	-0.8	-0.6
	Deviation (%)				0.5%	2.5%	-1.0%	0.8%	0.2%	1.6%	1.0%
5- KT77 MOV	V_{g1} (data sheet)				-20.0	-22.0	-25.0	-28.0	-31.0	-34.0	-37.0
	$V_{g1_{TW}}$ (model)	-19	32	300	-19.8	-22.6	-25.5	-28.3	-31.1	-33.9	-36.8
	Deviation (V)				-0.2	0.6	0.5	0.3	0.1	-0.1	-0.2
	Deviation (%)				1.0%	-2.9%	-1.8%	-1.0%	-0.4%	0.2%	0.6%
6- KT66 GEC	V_{g1} (data sheet)				-28.0	-32.0	-36.0	-39.0	-43.0	-47.0	-52.0
	$V_{g1_{TW}}$ (model)	-18	30	250	-27.2	-31.1	-35.0	-38.9	-42.8	-46.7	-50.6
	Deviation (V)				-0.8	-0.9	-1.0	-0.1	-0.2	-0.3	-1.4
	Deviation (%)				2.7%	2.7%	2.7%	0.2%	0.4%	0.6%	2.7%
7- KT66 Marconi 1956	V_{g1} (data sheet)				-28.0	-32.0	-36.0	-39.0	-43.0	-47.0	-51.0
	$V_{g1_{TW}}$ (model)	-27	30	300	-27.2	-31.1	-35.0	-38.9	-42.8	-46.7	-50.6
	Deviation (V)				-0.8	-0.9	-1.0	-0.1	-0.2	-0.3	-0.4
	Deviation (%)				2.7%	2.7%	2.7%	0.2%	0.4%	0.6%	0.8%
8- EL34Mullard 1964	V_{g1} (data sheet)				-25.0	-28.0	-31.0	-33.0	-37.0	-40.0	-43.0
	$V_{g1_{TW}}$ (model)	-17	25	250	-23.5	-26.8	-30.2	-33.6	-36.9	-40.3	-43.6
	Deviation (V)				-1.5	-1.2	-0.8	0.6	-0.1	0.3	0.6
	Deviation (%)				6.1%	4.1%	2.6%	-1.7%	0.2%	-0.7%	-1.4%

TABLE 2: Comparison of different values of V_{g1} extracted from the datasheets to the predicted values of V_{g1TU} . The overall standard deviation σ is 1.6%.

i2i Transmitter/Receiver

Reviewed by Chuck Hansen

Aerielle, Inc.

625 Ellis St., Suite 206
Mountain View, CA 94043
650-967-4047

www.aerielle.com

List Price One Pair \$120 US
(Street price \$90)

Dimensions: 1.25" W × 5/8"
D × 2.5" H for each module

Warranty: 1-year limited warranty
parts and labor on the i2i Stream.

The i2i® Stream is a wireless digital audio device that allows you to broadcast your favorite music to friends who also have i2i Streams. i2i Streams can send or receive "CD quality" sound up to 30' wirelessly. You can plug any MP3 player into one end of the i2i Stream and your favorite earphones into the other end and choose one of seven RF channels to start broadcasting. The i2i Stream is said to be universally compatible with iPod®, MP3 and media players, and 32Ω or higher impedance earphones, headphones, and portable speakers.

OPERATION

The Aerielle i2i Stream consists of two or more identical battery-powered modules that can function either as a transmitter or receiver (**Photo 1**). You recharge the modules by plugging them into an active USB port, or by means of an optional AC powered adapter. Charging is indicated by a flashing red LED. When the LED stops blinking and stays on, the battery is charged. Initial charge time for me was about five hours, and the units are said to operate 5-7 hours per charge.

The unit on the right in **Photo 2** shows the on-off switch/volume control

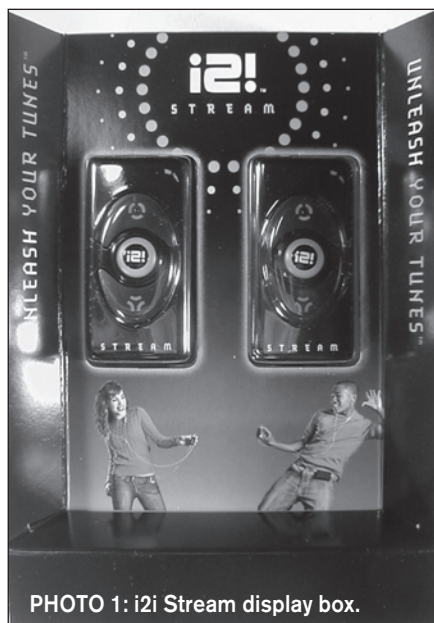


PHOTO 1: i2i Stream display box.

on the top and the mini-USB input port for the battery charger. The unit comes with two pairs of mini-phone jack I/O cords (12" and 28") and two USB-mini USB battery charge cords. I would invest in the optional AC powered battery charger if you do not want to tie up your computer to charge the batteries. The computer USB ports on my desktop are only powered after



PHOTO 2: i2i Stream modules.

Windows XP starts.

I found the i2i units quite easy to operate. You insert the smaller plug of the audio I/O cable into the small jack at the bottom of the unit you select to be the master transmitter. Then you plug the larger plug of the audio I/O cable into the output jack of an iPod or any other analog audio source. The i2i passes the original audio source through to its own headphone output jack whether the i2i is powered or not, making it easy to set the transmitter input level.

You insert another stereo headphone plug into the larger jack at the top of the i2i unit you want to be the receiver. You can add as many i2i receivers as you want. To power up each unit you press the square nub on the volume control inward and hold it until the large center "i2i" Channel Select button lights. A small battery indicator LED near the volume control will flash one to four times to indicate the relative battery charge power level.

There are seven RF channels available. Set each i2i unit to the same channel color by pressing the large center channel button until they all match. I used red for my testing with no sign of interference from any other nearby i2i units. (I cycled through the other six colors to check for any external interference and found none.)

Finally, you must select the transmitter mode by pressing the Send button above the Channel Select button, which will flash to indicate it is an active transmitter. You select the receive mode by pressing the Receive button below the Channel Select button, which remains lighted. I found this phase of the operation a bit quirky because the button you must activate is on the opposite end from the audio jack you use in that mode.

With the audio source to the transmitter active, everyone with an i2i receiver should hear the same music source and can set their headphone level independently. They can switch sources by having another person assume the “transmitter” role. You can hang the i2i from either the supplied neck lanyards or pocket clips.

AUDITION

I auditioned the i2i Stream units with an iPod Nano and a Creative Nomad II MP3 player. The Nomad is limited to 128k sampled CD tracks, while the iPod played a mix of 128k through lossless AIFF/WAV file CD tracks.

The iPod Nano itself is a pretty good audio source with lossless WAV or AIFF music. With its 128k sampled audio, the older Nomad is less so. The i2i was equal to the Nomad in sound quality, while it couldn't quite compare to the iPod at its higher sampling rates. Remember, the i2i needs analog input signals, so it must re-process the converted analog audio signals of the other players, which is never as good as doing the original D-A conversion.

I found that the range of the 2.4GHz transmitter with fully charged batteries (four LED flashes) was 36'. Beyond that distance the number and duration of dropouts increases.

The i2i is not aimed at high-end audio enthusiasts, so for its target market (who probably use the lowest bit rate in order to store the most songs) it will do quite well. The midrange was nearly equal to the sources, with a bit edgier treble and more noticeable and less controlled bass. The bass seemed louder than the iPod source, so some added LF emphasis may be in play at lower volume levels.

MEASUREMENTS

Measurement data depended on the battery conditions of both transmitter and receiver. I maintained the battery charge between three LED flashes and fully charged four LED flashes, recharging when necessary. As a result, the data repeatability is somewhat variable.

The i2i Stream inverts polarity somewhere between the transmitter input and receiver output, because

the passed-through signal has the same polarity. The input impedance with the transmitter power off measured 935 Ω at 1kHz, 2k7 at 20Hz, and 866 Ω at 20kHz. With the transmitter powered, these numbers dropped to 674 Ω at 1kHz, 2k5 at 20Hz, and 648 Ω at 20kHz. Your audio source will need to be no less than 65 Ω to avoid altering the frequency response. The iPod Nano output is rated for 32 Ω .

When starting the receiving unit, there is a beep whose loudness represents the volume setting. The unit powers up at the same volume as that which was set at shutdown. When the receiver is shut down, there is another beep. There is no indication in the receiver that the transmitter is shut down, just loss of audio. There is a very low level of hiss in the receiver unit phones at maximum volume with the transmitter off.

I recorded the response for loads of 47k, 60 Ω , and 32 Ω for Fig. 1. The capacitive coupled output rolls the low frequency off -3dB at $f = 1/2\pi RC$. At 47k, this -3dB point is 3Hz, but it increases to 21Hz with a 32 Ω load. The high-frequency response was flat out to 22kHz, where it dropped rapidly due to the steep digital LP filter. A-weighted hum and noise (maximum receiver volume, transmitter active with its input shorted) measured -76dB relative to the maximum undistorted 1kHz output level of 384mV RMS. DC offset at the receiver output was less than 1 μ V.

Figure 2 shows the maximum volume receiver output voltage versus transmitter input voltage. Gain at 1kHz is about 22dB up to the point where compression is inserted to limit the output voltage. There is more compression imposed at 20Hz than at the higher frequencies.

The i2i Stream THD versus frequency is shown in Fig. 3. I maintained the transmitter input at the point just below where output compression occurs and set the receiver volume to what I considered to be the maximum comfortable music listening level of 100mV pp (313mW into 32 Ω). I used the distortion test set steep 22kHz LP filter to remove out-of-band noise and digital artifacts. The distortion rises rapidly below 50Hz, and there is a bit of a notch at 60Hz indicating some pickup

of the 60Hz line frequency somewhere in the transmit-receive audio chain.

Figure 4 shows the THD+N versus transmitter input voltage with the receiver volume at maximum. Output level compression causes the steep increase in 1kHz THD at about 1.26V in RMS. The 20Hz trace shows this increase occurs at lower input levels,

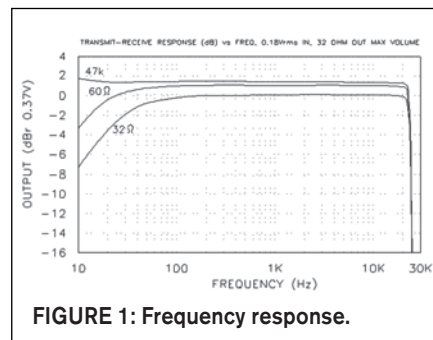


FIGURE 1: Frequency response.

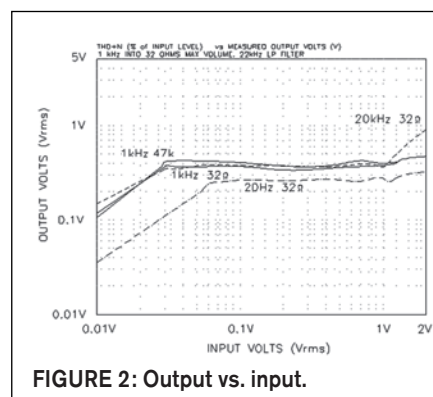


FIGURE 2: Output vs. input.

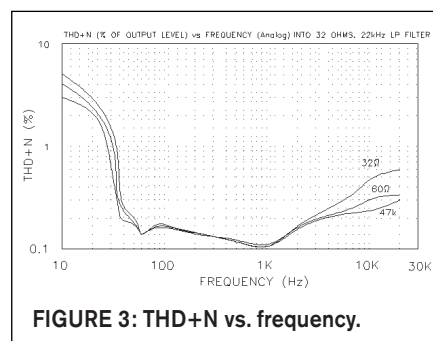


FIGURE 3: THD+N vs. frequency.

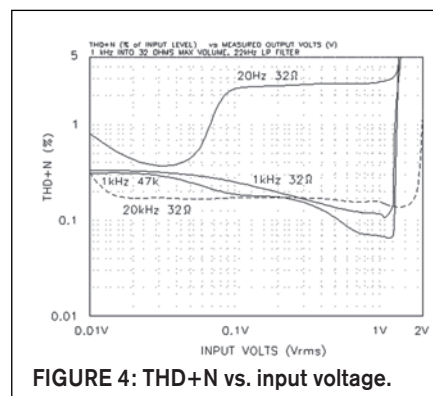


FIGURE 4: THD+N vs. input voltage.

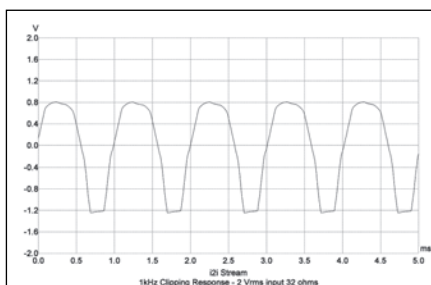


FIGURE 5: Clipping response.

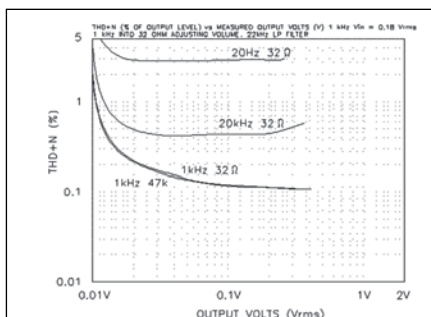


FIGURE 6: THD+N vs. output voltage.

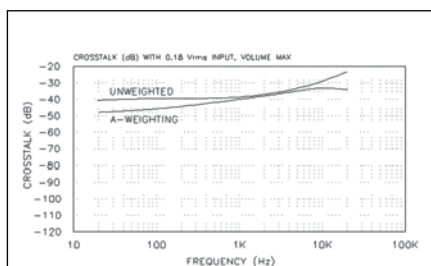


FIGURE 7: Crosstalk.

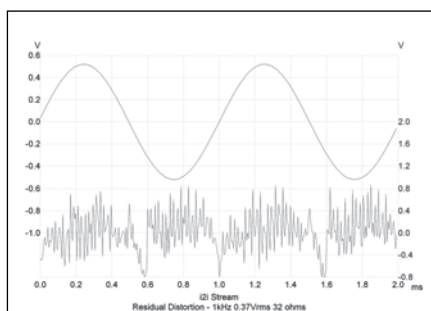


FIGURE 8: Residual distortion 1kHz.

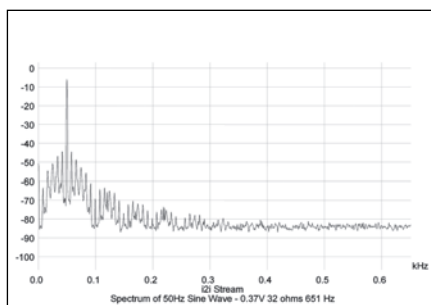


FIGURE 9: Spectrum of 50Hz.

and the i2i Stream compression seems to relax the compression somewhat at 20kHz allowing for more undistorted input voltage. You can also see this in **Fig. 2** with 2V RMS into the transmitter. The upper peaks of the waveform are softly compressed with increasing voltage while the lower peaks become hard clipped as shown in **Fig. 5**.

For the graph of THD+N versus output voltage in **Fig. 6**, I held the transmitter input to 180mV RMS and adjusted the receiver volume control to increase the output. This produces the more usual distortion characteristics with distortion decreasing with output. The low 180mV input signal does not allow output clipping at any of the test frequencies.

Crosstalk performance is shown in **Fig. 7**. There is quite a bit of coupling between the two channels, whose test setup includes the supplied 12" input cables and simulated headphone wires.

The distortion waveform for 370mV RMS into 32Ω at 1kHz is shown in **Fig. 8**. The upper waveform is the receiver output signal, and the lower waveform is the monitor output (after the THD test set notch filter), not to scale. This distortion residual signal shows mainly the 2nd harmonic, with quite a bit of high-frequency noise and output stage crossover distortion. I don't know whether the headphone amplifier is a linear amp or a battery-saving Class-D switching amp. The THD+N at this point is the lowest of the testing, at 0.108%.

The spectrum of a 50Hz sine wave at 370mV RMS into 32Ω is shown in **Fig. 9**, from zero to 651Hz. The THD+N measures 0.185%. 60Hz line frequency pickup has caused intermodulation of the 50Hz fundamental waveform so that the spectrum is filled with peaks every 10Hz, which is the 60Hz-50Hz IMD product. Note the minimums that occur at the location of each harmonic of 50Hz (100, 150, 200, and so on). Without a chassis ground on the i2i module, I could not attempt to eliminate the 60Hz pickup with the usual manipulation of ground connections.

Increasing the test signal to 1kHz produces the spectrum in **Fig. 10**, which shows mainly the 2nd and 4th harmonics projecting above the rather high

noise floor.

Figure 11 shows the i2i Stream output spectrum reproducing a combined 10kHz + 11kHz intermodulation distortion (IMD) signal at -12dB below peak output into 32Ω. There are significant products at 1kHz, 2kHz, 8kHz, 9kHz, 12kHz, and 13kHz. The peaks around 20kHz and the more prominent peak at 21kHz are probably artifacts of the DAC/digital filter conversion.

SQUARE WAVE TESTS

I viewed the response of the i2i Stream to three square wave test frequencies on an analog scope at 1.5Vpp using a 32Ω load. I recorded the responses with my 16-bit digital scope. The response at 40Hz (**Fig. 12**) showed a large amount of tilt due to the limited LF response and phase shift.

The 1kHz square wave (**Fig. 13**) exhibits the Gibbs phenomenon ringing associated with the steep digital filters used in the i2i Stream. No clipping is evident on the highest peaks of the ringing, but there is still a noticeable tilt to the horizontal portions of the square wave.

Finally, the 10kHz square wave appears as a sine wave in my scope capture of **Fig. 14**.

CONCLUSION

I can't really compare my measured data to the manufacturer's data because there wasn't any provided. The audition was hampered somewhat by having to reconvert the analog output of a digital audio device. Its low input impedance would not lend itself to transmitting line-level analog signals. While the measured results do not exhibit high-end performance, the i2i Stream works quite well at its intended function, especially with the lossless iPod audio files.

Manufacturer's Response:

The i2i Stream wireless music streaming device digitally transports and reconstructs an audio stream that effortlessly exceeds the quality of 128Kbps MP3 tracks. In fact, its 16-bit, stereo, 48kHz uncompressed digital medium rivals the quality of similarly sampled lossless recordings.

While it's true that some audio quality is lost in digitizing and then recon-

structing the analog audio signal, this is an inevitable condition of any analog-interfaced wireless audio device, and certainly any wireless device which can mate with iPods. Direct digital streaming would not only require integration within the player, but raises issues of copyright infringement with the direct distribution of digital content.

If the goal of the high-end audio enthusiast is to realize digital content with the highest possible fidelity, they will embrace wired speakers or wired headphones. But for the convenience and freedom of a wireless device, our measurements show that the i2i Stream provides excellent reconstruction of the source material within the restrictions of going through an interim analog medium. For the benefit of going wireless, it may still be enjoyed by the enthusiast with this understanding.

Note that the i2i Stream does not compress the instantaneous dynamic range of the audio. Rather, it has an Automatic Level Control circuit (ALC) which acts over longer periods to improve the overall experience in a one-to-many broadcast scenario. When multiple listeners or several simultaneous streaming destinations are involved (such as multi-room audio), all recipients need to be able to customize their listening levels, and keep those levels consistent across changes in source material and manipulation of source devices by the person who is transmitting.

For this reason, we use ALC to normalize the input level. Almost all cases of actual source material have mixtures of frequencies, so that the ALC attenuation engages and maintains an average level of attenuation, and avoids the distortion artifacts possible with certain pure tones especially at low frequencies.

The i2i Stream is intended primarily for use with music players with headphone outputs. We recommend setting the player volume at halfway, which in most players is enough to engage the ALC attenuation at an average level while staying clear of saturation. Our audio tests are performed under these actual operating input conditions, and perform quite well under audio analysis.

We have found that under real-world conditions, the user will adjust the input volume of a complex source to optimize

subjective clarity and minimize distortion to which the ALC may contribute under sub-optimal input conditions.

The i2i Stream system maintains a consistent signal polarity from the source, to the transmitter pass-through, to the receiver output. No inversion of signal occurs. There is a small amount of fixed digital-channel latency which may appear as an inversion with pure sine waves at certain fixed frequencies.

Notwithstanding the differences in impedance, our tests show flat frequency response (within 3dB) from 20Hz to 20kHz with an unloaded iPod source, with an iPod source loaded with earbud headphones, and with a CD player line-output source. Adaptation to the line output is possible with a passive attenuator accessory in a home-audio accessory kit.

The i2i Stream has excellent Signal-to-Noise Ratio, better than 75dB below the reference signal. This spec is borne out by 3rd-party testing as well as our own.

We performed Stereo Separation (Crosstalk) tests with standard i2i cables and standard iPod headphones, with nominal input levels at mid-ALC engagement as previously stated.

In the unloaded-transmitter test, which is equivalent to the simple remote listening scenario, our result was -75dB at 1kHz, and it was no worse than -50dB all across the band. In the test where the transmitter is loaded with the iPod headphones, equivalent to the music-sharing scenario, the result was a solid -45dB across the band.

If the Stereo Separation tests are performed at the lower threshold of ALC operation, or up near the saturation of the codec, it could have unpredictable results. Significantly poorer results could also be due to defective cables or crosstalk in nonstandard cabling in the test apparatus.

Thank you for your interest and for the review. In certain cases where your findings and the test conditions deviate from ours, we are pleased to have had the opportunity to add our comments.

John R. Haggis
VP Engineering
Aerielle, Inc.
aX

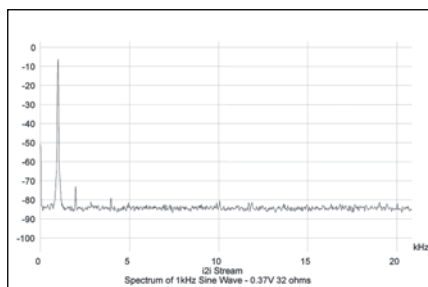


FIGURE 10: Spectrum of 1kHz.

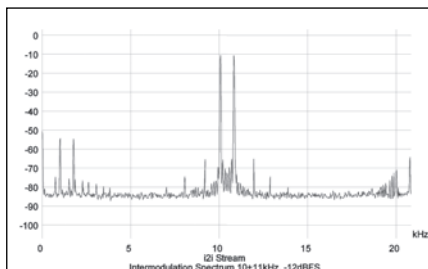


FIGURE 11: Intermodulation spectrum 10kHz + 11kHz.

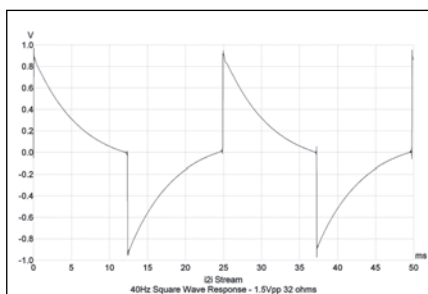


FIGURE 12: 40Hz square wave.

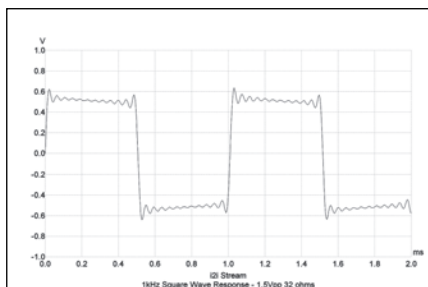


FIGURE 13: 1kHz square wave.

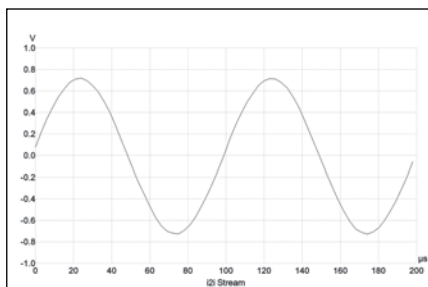


FIGURE 14: 20kHz square wave.

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Continued from page 33

where : $Vg1_{T/U}$ is the value of $Vg1$ in the triode or ultralinear mode for a specific plate voltage V_a .

This formula enables the prediction of any value of $Vg1_{T/U}$ for high tensions ranging between 350V and 500V. As you can see in **Table 2**, this relation seems to be highly significant because it exhibits a standard deviation σ of only 1.6%. Based on the data I collected, the statistical meaning of this is that 95% of the time, the real value of $Vg1$ will be within the calculated value of $Vg1_{T/U} \pm 2\sigma$ or $Vg1_{T/U} \pm 3.2\%$. As a general rule of thumb, I can more conservatively state that this model will predict any value of $Vg1_{T/U}$ with an accuracy $\pm 5\%$, which will be true 99% of the time.

FURTHER THOUGHTS

The development of this mathematical model was made strictly from published datasheets. Though my personal measurements tend to confirm the correlations, exhaustive evaluation with empirical data would be required to further validate and perhaps refine the model. Nevertheless, with an accuracy of $\pm 5\%$, it fully answers my needs and exceeds my expectations.

The model was designed on the basis of maximum bias current, but behaves equally well for lower values. In fact, the aforementioned levels of accuracy remain the same for bias currents set anywhere between 80% and 100% of the maximum current bias admissible. You only need to carefully determine the value of $Vg1_{P425V}$ with the desired current bias and apply this value in equation 3.

CONCLUSION

Tube technology is certainly one that is very mature. Still, there seem to remain little secrets that are there for us to discover. This is at least what these correlations suggest. With this mathematical model in my toolbox, I can now design and optimize driver stages well before I start building my amplifiers.

For your convenience, I have prepared an Excel application that enables fast calculation of $Vg1_{T/U}$. You can download this file from the *audioXpress* website. I wish this model will be as useful for you as it is for me.

aX

CONTRIBUTORS

George Danavaras ("The Reflection Loudspeaker," p. 5) graduated from National Technical University of Athens, Greece in 1986 with a degree in Electronic Engineering. He currently works in the R & D division for a Greek Telecommunication company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

Jan Didden ("The Borbely Annals," p. 12) built his first OTL amp with 807 tubes 35 years ago. He has built speakers, preamps, and tape recorders, but is most interested in power amps, especially using error correction as discussed by Hawksford. Many of his projects have been published by *audioXpress*. Now retired from a career with the Netherlands Airforce and NATO, he tries to complete all those half-finished projects accumulated for lack of time. He now also has the time to travel to interesting audio events and interview audio luminaries. His projects are documented on his Linear Audio website.

Rich Johnson ("Designing a Midrange Horn," p. 20) resides in Newark, N.J.

Bill Fitzmaurice ("Building the Occam Audio MTM 4 Kit," p. 30) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long. Vice president of DeltaSounds Loudspeakers Inc., Fitzmaurice is the author of *Speaker Builder's Loudspeakers for Musicians* and over 30 magazine articles dealing with speakers and electric instruments, as well as the action-adventure novel *Operation: Sergeant York*. Bill and his wife of over 30 years reside in Laconia, N.H.

Rudy Godmaire ("Vg1 Correlations," p. 32) works as a sales consultant for Bell Canada. DIY audio has become a passion ever since his first attempt in 1998. He deeply enjoys sharing his passion with other DIYers and especially with friends François and Pierre. Above all, listening to music and attending classical concerts and opera with his beloved wife Elaine remains among the most fulfilling activities he cherishes. Readers may visit his personal website at www3.sympatico.ca/r.godmaire.

Chuck Hansen ("Review: i2i Transmitter/Receiver," p. 34) is an electrical engineer and holds five patents in his field of engineering. He began building vacuum-tube audio equipment in college. He plays jazz guitar and enjoys modifying guitar amplifiers and effects to reduce noise and distortion, as well as building and restoring audio test equipment. He enjoys sailing and has over 150 magazine articles to his credit.

John Shand ("Jazz Track," p. 46) is a CD reviewer for *Australian HiFi and Home Theatre Technology*.

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CORRECTION

Cynthia Wenslow was the photographer of Photo 1 in our Burning Amp show report (Jan. '09, p. 39). We regret the omission and apologize for any inconvenience.—Eds.

COMPUTERS AND ANALOG AUDIO

Since digital audio—and converting analog audio to digital—is such a hot topic lately, I just wanted to share a few observations and suggestions on the subject.

1. If you are planning to digitize your record collection, give serious thought to what you use for the conversion. The sound cards built into many computers (especially the on-the-board ones) are sometimes *really* bad—worse than you can imagine—hissy, noisy, high distortion. If your recordings all seem noisy, or have little twittery noises, don't waste time trying to fix the unfixable—you can get a pretty nice sound card for a desktop machine for about \$30. Try the Sound Blaster Audigy SE, which has an odd quirk of using the same plug for line and mike in and switching them via software, but is relatively quiet and clean, sounds quite nice, and costs less than \$30 at Wal-Mart or via mail-order.

2. Desktop recording software, especially the commercial applications such as Adobe Audition (a *very* nice program), has many features and requires a lot of computing power. If you're using an older machine that isn't all that fast, you may not be able to record audio with some programs (editing, because it is not done “real time,” will just take longer on a slow machine—recording, however, will skip and hiccup and do all sorts of evil things). The answer is to use *less* powerful software when recording. While I was trying to do some recordings for a friend on an old 866MHz Dell machine—with the Audigy SE card, Adobe Audition couldn't record at all without continual glitches and hiccups. The solution was to use the simple little recording utility that came with the sound card to record—which worked

perfectly every time—then load the files into the good editing program to edit them afterwards.

3. Generally, real-time (as in audio recording) is easier on a fast machine, and, with slower machines, you must be careful to not “distract the machine” while actually recording.

You can use a separate drive partition for recording because the C drive *always* becomes fragmented. An external USB drive may be a good answer, *but* make sure that your machine has the newer USB 2.0 high-speed ports. Older (pre-2001) machines often have the slower USB 1.0 ports, which are supposedly fast enough for audio, but are really borderline. Adding a second internal drive would then be a better bet, or you can update the USB ports by adding a USB 2.0 card for about \$15.

Of course, try and ensure that other programs aren't running, *especially* antivirus and security programs. If they are, be sure to switch off the “live scan” function while recording, which slows the machine down a *lot* even when it's not scanning the drive. (You can disconnect the machine from the Internet while recording if this makes you uncomfortable.)

If you have an old machine that's not being used, it might make sense to dedicate it to audio recording. Partition the hard drive (or add a second one, which doesn't need to be big by today's standards), leave off all extra software (such as antivirus programs), and install a decent sound card. I have an old Dell 866MHz box set up that way with the Audigy sound card, a 20GB hard drive divided in half—so half is a clean 10GB “music” partition—the Audigy (free) recorder to record, Adobe Audition to edit, and nothing else running on it. It works *very* well.

4. Sometimes it's difficult to figure out what you're looking for, and what it's called. You may find that a computer store, an audio store, and an audio “studio equipment” store each sells something that *does* the same thing, but they each call it something different, and the

prices are wildy different as well. This is especially true for computer/audio gear.

For example, you may want to purchase a good-quality analog output from your laptop—the one with the not-so-good audio card. You can buy a “USB Headphone Amplifier” on eBay for \$85; a line-out equivalent for a similar price; a “USB Sound Card” from a computer outlet for anywhere from \$6.95 up (many of which also take analog IN and convert it to digital); or a “USB 2-input/2-output Digital Audio Interface” by Behringer from a studio-audio outlet for \$29 that does both input and output *and* provides a digital output for connecting your own outboard DAC (a feature that generally costs about \$100 from a “home audio” device).

Behringer also makes a “headphone amplifier” (AMP800-MINIAMP), which has two inputs and four headphone outputs—each with a separate level control. Real handy if you connect it to your computer and connect your powered speakers to one “headphone out” and your headphones to the other. It even includes another input (with level control) for your iPod, which you can send to any of the four outputs individually. (Yes, it might be more fulfilling to build a better one, but, for \$39, you can't even buy the case and connectors, and, if you need it today, you're sunk.)

For that matter, people spend serious money for “iPod speakers” when iPod speakers, “self-powered studio monitors,” and “computer speakers” are all the same thing—speakers that take line-level inputs. Cheap “computer speakers” make great iPod speakers, and cheap “powered monitors” make even better computer speakers.

5. Grounding. Computers still need to be grounded. Even a laptop, running on batteries, may be susceptible to noise pickup. If so, finding a piece of actual metal on it and connecting it to an earth ground—like the ground pin on the outlet strip—will help (sometimes the little screw fittings that you use to secure the VGA connector are actually connected

to the metal chassis when nothing else seems to be.

6. Digital recorders. The latest of a new generation of portable recording devices, the Zoom H2, can record stereo or surround sound digitally to a standard camera-type SD memory card (which is now incredibly cheap). It has four *built-in* microphones. Because the recordings are already digital, you don't need to convert them, and the quality is very good. It also has microphone and *line* inputs, so you can use it to digitize your line level inputs—such as from your turntable.

It can record low-quality MP3s to four-channel 48kHz/24 bit files, has several limiters and filters which can (optionally) be invoked, and runs for about four hours on a pair of AA alkaline batteries. You can also connect it to your computer and use it as a digital sound card and/or digital microphone, and it has threads to fit a standard camera tripod mount. You can also take its microphone output from the line out and use it as a microphone with built-in preamp. And it's about the size of an electric razor and costs about \$200. There are several competing products which claim to be even better (for a bit more).

7. Last, digital sound in general clips *hard* when it clips. When you record digitally, you can usually afford to record at lower levels than you might be used to, because most converters are very quiet. This is good, because, unlike analog systems, digital systems clip *absolutely* at 0dB (or whatever their "0dB" level is). There is no graceful rounding of waves, or gradually increasing distortion.

Most digital devices are perfectly clean at -0.1dB and cut everything off flat at zero. So, when using a sound card or digital recorder, you want the levels to be good, but you *never* want the peak indicators to flash—not even occasionally. Reduce the level until they don't if you want to stay clean. Luckily, most digital recorders have peak indicators that are *really* instantaneous, responding to a single sample that clips.

Also, if you're new to digital recording and editing, be *sure* to read the book for your editor—there are all sorts of cool new features that simply didn't exist with analog (check out "normalize" and "noise removal" and, in Adobe Audition, "edit in spectral view"). Before you buy

a cheap editor, or settle on a free one, download a demo copy of Adobe Audition and decide whether you can really live without the extras. Most programs even include the ability to make perfect test tones, filtered noise, and sweep tones—just by punching in what you want (such as "431Hz sine at -3.26dB for 13.4 seconds").

Keith Levkoff

kLevkoff@panix.com

SIMPLE PROTECTION FOR SPEAKERS

When I read about David Mayfield's speaker mishap ("Speaker Setback," Xpress Mail, Jan. '09, p. 42), I thought I'd pass along the following simple suggestion. By connecting a 0.5A fast-acting fuse in series with each of your speakers, you can protect your speakers from power company mishaps, like in David's case, and from solid-state amplifier failures which can result in destructive DC voltage at the speaker terminals.

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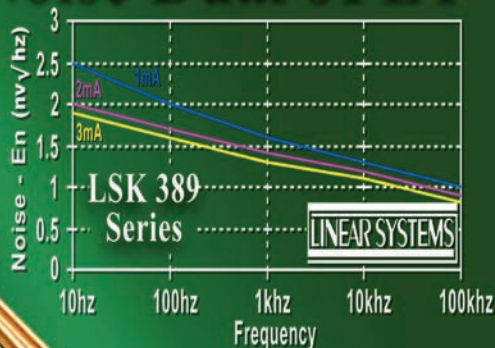
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Castro Valley, Calif.

HORN OPERATION

In response to Mr. Kolbrek's response to my letter ("How Horns Work Revisited," Dec. '08, p. 18):

I would tend to agree that my dismissal of the horn equation may be a bit too absolute for every case, but for what I do, it has not shown to be useful. In some circumstances I can see where its simplicity might be attractive.

Quite recently I had a thought: if we have an exact formulation of a "horn," namely the OS waveguide, for which solutions are available, then by simply plugging the OS contour into the horn equation a reasonable comparison of the accuracy of the two approaches can be made. I tried this and the results were quite interesting. The mathematical details of solving these equations will need to be forgone for brevity, but the conclusions are straightforward and simple.

The horn equation appears to break down right at cutoff, and it is not hard to see why. It is right at cutoff that the wavelengths begin to exceed the size of the device itself. This means that in Morse's original assumption limitations

$$\left| \frac{d}{dx} \sqrt{s_0 \cdot e^{m \cdot x}} \right| \ll 1.0$$

it appears that we should actually compare the rate of change to the wavelength. This makes perfect sense because it is actually the rate of change of the boundary with respect to the wavelength that determines how "fast" the boundary changes.

In the horn solution of the OS waveguide the answers are completely wrong right at cutoff, but become progressively better above cutoff. And yes, a clear cutoff is predicted in the horn solution that

does not exist in the exact solution. The "loading" of the OS goes to infinity at cutoff and becomes imaginary below this point, almost exactly like some other horn contours do.

Above cutoff it is found that the relationship θ/a , where θ is the wall asymptotic wall angle and a is the throat radius, plays an identical role as the flare rate m does in an exponential horn. In essence, as far as horn theory is concerned, the OS waveguide and the exponential are almost completely equivalent contours.

In this light it appears that the horn equation is not very good at predicting the loading of a device near its cutoff, but, of course, because all devices become the same at higher frequencies, it works fine in that region (as far as the impedance goes).

In regard to my claim that virtually all contours for practical devices will result in equivalent performance, I submit the following simulation (Fig. 1) of the same driver on three devices, conical, exponential and oblate spheroidal, with identical input and output areas and length.

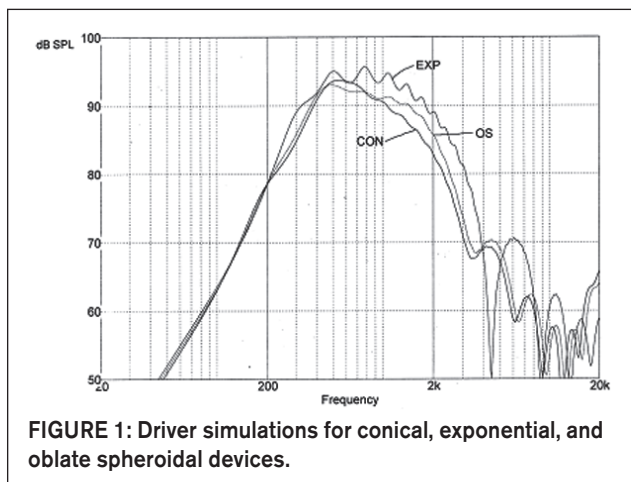


FIGURE 1: Driver simulations for conical, exponential, and oblate spheroidal devices.

This simulation uses a finite (conical) element simulation for the contours, *not* the horn equation or the wave equation. It is thus more accurate than the horn equation but less accurate than the exact wave equation (which only exists for two of these devices). It is the most accurate at the lower frequencies where, as I have discussed, the horn equation is losing accuracy.

You can see that there are differences, but these differences are not such that one device is clearly superior to another at the lower frequencies where the model is the most accurate. The ex-

ponential horn appears to have a slight gain advantage, but it also appears to have a highly resonant loading. The flare rate for the exponential in this case is not very great, which is why its efficiency appears higher, but it also beams more with substantial lobing, although at these higher frequencies the directivity model is becoming suspect. At the lowest frequencies and near cutoff is where they all tend to look about the same, and this is precisely where the horn equation says that they should be the most different. The horn equation just does not seem to give useful results at low frequencies because, as I have said, all contours appear to be converging on the same answer, and it fails at higher frequencies because its plane wave assumption is invalid.

In conclusion, let me just say that throat impedance and efficiency are not irrelevant in all situations, but they are not nearly as important or as different as the horn approach would lead us to believe. (Nonlinear distortion performance of horns and waveguides is not at

issue here because neither approach offers anything in this regard, so I am not sure what Mr. Kolbrek is referring to, but I would suggest a review of the several papers that my colleague and I have done on distortion perception which shows how the horn is very unlikely to be a significant factor in the "nonlinear" sense, although its HOM (high-order modes) performance appears to be critical.) The differences in various contours are predominately at the higher frequencies where

the horn equation offers no insight into performance.

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CLASS D AMPS

I just wanted to add a few more comments to Bruce W. Brown's excellent article about Class D Amplifiers in the November 2008 issue (p. 22).

First, I do want to dispute the oft-repeated claim that "the design of class A amplifiers produces the least distortion." From an engineering point of view this is a bit of a simplification, and really only

applies to relatively simple transistor and tube circuits. The premise carries over when feedback is added into the picture because “regular” negative feedback serves to reduce distortion—so, the less distortion you start with, the better off you are. Class B and higher types of circuits are prone to distortion because they “hand off” the signal between multiple devices, and because devices are rarely—if ever—identical, or identically opposite enough for distortion to cancel exactly; this hand-off process tends to cause distortion.

If you look at the individual parts in a class D amplifier, they each produce nearly 100% distortion, because, by design, they are not operated in a linear manner at all. If, however, you look at the overall circuit as a single “device,” then a good PWM amplifier could easily produce less distortion than a good class A amplifier. In theory, a good PWM amplifier could also be built without using feedback, although the sort of folks who design class D amplifiers generally aren’t superstitious about avoiding feedback. The main difference is that the transfer characteristics of tubes, transistors, and FETs are a property of the device it-

self, while those of a class D design are properties of the circuit and are quite divorced from the properties of the devices used in the circuit.

For those not interested in engineering technicalities, you should simply accept that class D amplifiers cannot be directly compared to linear amplifiers (the analog kind) at the circuit level—and trying to do so is pointless. For example, some class D designs are sensitive to power supply noise and fluctuation, while others are entirely immune to it.

One fair generalization is that class D amplifiers must be more complex in order to produce satisfactory audio performance—nobody has yet demonstrated one using four or five parts that sounds anything other than awful. But, if you count an integrated circuit with a bunch of parts inside it as “one part,” then there are many single chip class D amps that far outclass most class A SET triode amps.

There are many interesting class D amplifiers available—and some of them are amazing bargains. The Bel Canto model that Bruce is hoping to audition uses a chipset made by someone called TriPath (now maybe out of business, as

it turns out). There are several models using the same or similar chip sets which are available for much less money.

41Hz (www.41hz.com) offers several nice-looking class D amplifier kits (using TriPath chips) of various power levels and configurations—in both surface mount and throughhole versions (you can do digital with non-SMD components, it just takes more design effort). I have ordered two and they appear to be well designed—but I haven’t gotten around to building them yet.

Trends Audio (www.trendsaudio.com) sells a very small amplifier called a model 10.1, which puts out about 10W/channel (WPC) using a TriPath chip—for about \$150. It runs on a 12V 4A power supply (included), or can be run on a big 12V battery or car voltage if you like. It sounds remarkably good, absolutely quiet, and very clean. The build quality is also excellent.

I haven’t done any serious comparisons, but, when connected to a pair of vintage Advent Smaller in place of a 190 WPC Hafler power amp, the little Trends amp had cleaner midrange and the bass sounded clearer and obviously

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less muddy. It was also able to reach reasonable listening levels (although, obviously, the Hafler would play louder).

This is very impressive considering that the Advents have heavy woofers known to require solid damping and serious power to work well, and less-than-impressive midrange—and considering that the Trends amp is less than 1/10 the power of the Hafler and much less than 1/10 the weight or size. The improvement in sound was not at all subtle.

For the budget-minded, there was an amplifier based on the same or a similar chipset available until recently for \$29 (the Sonic Impact T-Amp). It is small, very plastic, has really cheap thin binding posts, and will produce about 2 WPC using eight AA batteries or about 10 WPC using an external—and not supplied—12V power supply (plug-in, battery, or car). The entire circuit inside is about the size of a fat thumb—including the sizable heatsinks that it doesn't have and doesn't need—it doesn't even become warm under normal use. It sounds remarkably good when running on external power (in my opinion, and those of many reviewers, better than the few single-ended-whatevers I've heard). It even sounds quite good at low listening levels running on AA batteries—a set of which will last eight or ten hours—which makes it a great laptop or iPod amp. The company has discontinued this model, but they now sell a more expensive model of basically the same amp, and the original occasionally turns up on eBay (now not cheap) or in clearance bins.

As with many modern products, the sound quality of class D amplifiers is market driven. There are some out there that sound very good, and others that do not—and, obviously, in many cases the size and weight are the major considerations in the design. The point is that they are quite capable of sounding very good if well designed and implemented. It would be a serious mistake to assume that a good class D amp may not equal or top a good amp of any other type.

Unfortunately for DIYers, class D designs tend to be quite complex, both because of the circuit complexity and because of the design and fabrication requirements due to the high frequencies involved. Even the chipsets tend to be quite fussy about circuit layout and

component specifics. Unless your high-frequency design skills are up to date, it's probably safer to stick with pre-fabricated PC boards if you set out to build one. (Check out the user forums on www.41hz.com for a good idea of the design considerations and pitfalls common to class D amps.) In general, many parts in certain circuit locations must be of specific types because of the high-frequency requirements, and so you should not substitute parts unless you are very familiar with the design requirements.

On the upside, most of the chip-based designs are quite insensitive to power supply variations and noise beyond having sufficient power of reasonable quality available, so power supply design is usually simplified. (Many of the 41Hz models include a simple but sufficient power supply on the amplifier board—of the “four diode bridge and a capacitor” variety.)

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CD PHENOMENA

In reference to Keith Levkoff's excellent, somewhat neophyte-oriented, treatise on CDs (“How CDs Work,” Xpress Mail, Dec. '08, p. 42) and proper lambasting of “magic” potions to improve audio playback quality of the same, I would like to add a few observations and experiences gathered over the last 35 years of audio engineering.

In the mid-70s, when CDs were relatively new, and at my first retail hi-end audio salon work experience, a sales rep paid us a visit, flogging a nifty little green magic marker (\$79 retail). One was to apply around the outer circumference of the disc, thereby removing reflections from the polished edge of the new CD and magically improving the sound. . . somehow. When he was unable to explain scientifically how this might be, he was sent on his way with much derisive laughter from me and my skeptical colleagues. We peeled the label off the sample and discovered a Schaeffer No. 8 green marker. We actually tried it, just to be fair, in simple blind tests and had trouble breathing from laughter, and washing the green ink from our hands.

On a more serious bent, many years later, as the head service engineer for the Canadian Linn distributor, I experienced

very strange and repeatable CD hardware phenomena, about which I am still uncertain. As readers may be aware, Linn makes some very respectable and spendy CD machines. While servicing a big-dollar CD12—arguably a very fine Aston Martin-like player—I came across an odd phenomenon during repair. The laser “sled” was obviously faulty, with no laser focus activity on power-up. I replaced it, having a dozen brand new ones in stock, and cleaned and lubed the mechanism. Note that Linn units have very few, if any, adjustments required or available, as, say, in adjustable trimmers, for focus offset, focus gain, and so on, unlike some other machines.

During my final listening test, Diana Krall's drummer's snare brush *only* sounded nasty. Like crap, actually—hard and clearly distorted in some way. *Every other instrument* on this track, including her voice and cymbals, sounded perfectly fine or as good as my musical memory said. This was a very clean shop test disc, used and heard daily. The shop listening equipment was very modest, but the bad sound was plainly obvious in spite of this, and also later, with a room full of experienced people. They were sent on their way.

As an experiment, I replaced the laser sled with a new one. Voilà! Glorious! No bad snare brush sound. I repeated this A-B-C several times with the same results and the same listeners. The “eye pattern” (the high-frequency RF signal from the working laser outputs) on the scope was perfect-looking *in every case and aspect*. This confused me, to no end.

Caveat: Since the 70s, I had probably serviced or replaced literally thousands of CD laser mechs of every brand on the planet, and as such, have a pragmatic disdain for audio “voodoo,” or attempts to legitimize marketing audio BS, in any way. That said, I encounter a head-scratcher once in while. If it's played back with all 16 or 24 bits intact and synchronized, it is what it is, or is it? Laser jitter? Any similar technical experiences out there?

The “eye-pattern” previously mentioned can be compared to the signal coming directly off a tape recorder head, if you like. It's measured at the source, or as close as is practically measureable. In case you weren't aware: When you burn one on a typical PC or Mac, the level *coming directly off the burned disc* is approximately 60%

that of a commercial stamped audio CD. If your machine's servos and so on have no problems tracking at this actual lower recorded level, it will sound normal.

Where this issue typically shows up is in your crappy car CD player where the focus error/tracking/offset correction servos are already working overtime anyway, from bumps and vibration, due to driving on the road. Don't necessarily accuse your wives or kids of throwing your jazz CDs on the car mats when your car copies start skipping. On the other hand, it's a great excuse to not be willingly subjected to that vile and pernicious "Mamma Mia" soundtrack.

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WT3 PRINT OUTPUT

In the "Dayton Audio WT3 Woofer Tester (Nov. '08, p. 32)," G.R. Koonce notes, "The once missing file option seems to be the ability to output the graphs to files as pictures. This would be useful when using the WT3 to prepare material for publication. . ." I would suggest trying the "PrintScreen" key as follows: hold down "Shift" + "PrintScreen"; or "Ctrl" + "PrintScreen," or "Alt" + "PrintScreen." Any of these actions will allow the screen picture to be "pasted" into a document, by holding down "Ctrl" + "V" or right-clicking the

mouse and selecting "Paste." You can thus copy the screen picture onto a WORD document.

Or, better yet, copy it into a PAINT document. Then place the mouse-cursor over the "Select" icon (which is directly below the "Edit" choice on the menu line). Position your mouse "Cross-Hairs" at the desired point on the picture to begin cropping the desired section of the picture. Just hold down the left-click button on the mouse and drag to the opposite end of the picture in order to crop the desired section. Let go of the left-click mouse button. Next, hold down the "Ctrl"

key + "X" in order to "cut" the cropped section. Then switch over to a WORD document and hold down the "Ctrl" key + "V" in order to "copy" the cropped section into the WORD document. Presto! Instant picture!

The original "screen dump" (Fig. 1) includes all the top and bottom menu items, which are extraneous and unwanted. Cropping could result in the text being cut off or stretched. Hope this is helpful.

Angel Rivera

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aX

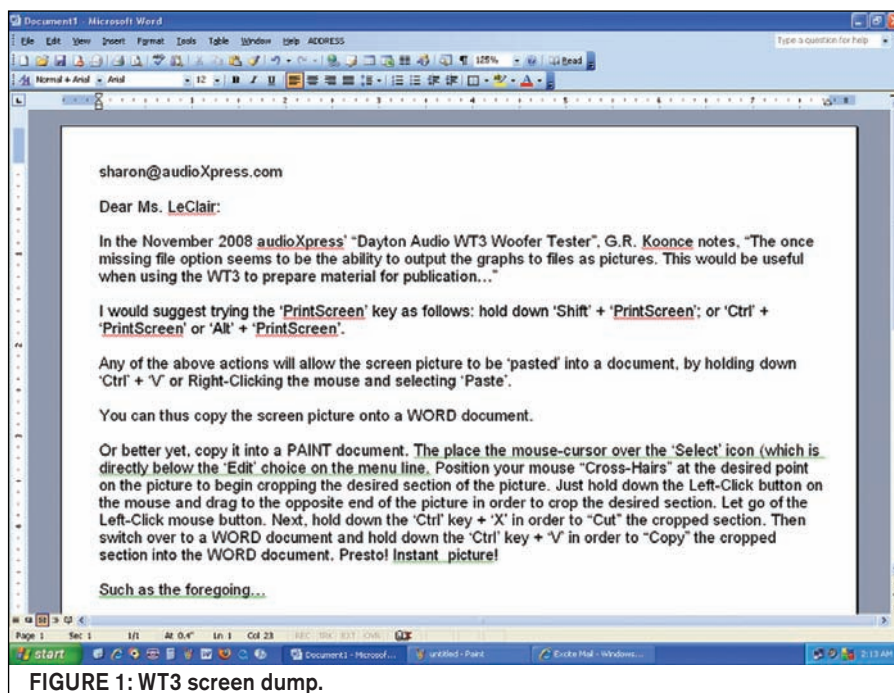


FIGURE 1: WT3 screen dump.

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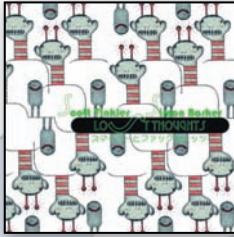
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Lost Thoughts|Scott Tinkler/Simon Barker|Kimnara NARA008

Before surrealism was usurped in the public imagination by hallucinatory painted images, the movement was primarily bent on directly accessing the subconscious via writing without technical, aesthetic, or moral filters. Such purity is approached by trumpeter Scott Tinkler and drummer Simon Barker in these improvised duets. Where much so-called free improvisation is intent on not being “polluted” by such mundanity as conventional melody, harmony or rhythm—and therefore is anything but free—these conversations between two long-term colleagues are allowed to follow the thread of invention unfettered. They range from a meditative temple of gongs and whispered horn to the thundering tom-toms of the apocalypse with napalm blasts of trumpet. Always the myriad sounds arrest and engage, so you never feel you’re eavesdropping on a private conversation.

Expensive Habit|Sean Wayland|SEED CD009|seanwayland.com

Sean Wayland writes wonderful melodies. The aptly titled *Beautiful* catches the ear the way a rose does the eye, and draws you to it for closer scrutiny of the tune’s contours, which are wrapped in an engaging breeziness. Then you want to hear it again, or whistle it. When you listen closely to Wayland’s piano (and synthesizer) solos, that same gift for melodies is hard at work, but is presented in such an unassuming fashion that it could pass you by, as your attention wanders to the brilliant rhythm section players who, like Wayland, are New York-based. German drummer Jochen Ruekert displays unflinching verve in negotiating the rapid twists of *Arc Etude*, which also features a spitfire solo from guitarist James Muller and the authoritative bass of Kiwi Matt Penman.



As Wide as the Sky|Tony Gorman/Bobby Singh|Birdland BL008

What a perfect circle: this music was recorded at Birdland Records, published by Birdland and then sold by Birdland. Filling the circle is Tony Gorman’s infinitely nuanced clarinet playing, perhaps less anguished than on his mesmerising *Songs of Hope* album, but no less moving. In fact, this pensive music is so devoid of affectation

it seems to bore its way directly through your sternum with no mental processing and with no concern for labels beyond an awareness of the purity of the improvising. Gorman begins by himself, gently accompanied by synthetic drones, before being joined by tabla wizard Bobby Singh on four duets. Miraculously, Singh seems to compound rather than dilute the sense of intimacy; the sense that the music is being whispered in your ear, like a confession from a best friend.



Early and Late|Steve Lacy, Rosewell Rudd Quartet|Cuneiform Rune 250/251 (2CDs)

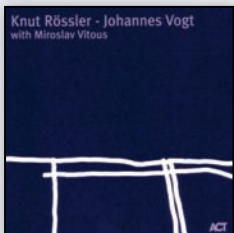
This double CD’s name refers to the fact that the quartet co-led by the late soprano saxophonist Steve Lacy and trombonist Rosewell Rudd is represented both at its beginning (1962) and end (1999–2002). From 1962 comes a session with the founding

father of free-jazz drumming, Dennis Charles, and bassist Bob Cunningham. This was only a few years after Lacy and Rudd had leapt from Dixieland to the outer reaches of modernity, and their band was dedicated to Thelonious Monk’s music, although one Cecil Taylor tune sticks its head up amid Charles’ loose-limbed, highly swinging propulsion. The later material, recorded live, is mostly Lacy’s. The more austere elements of his distinctively dark-toned soprano contrast sharply with Rudd’s explosive ebullience, unleashed over Jean-Jacques Avenel’s brilliant bass playing and John Betsch’s volatile drumming.

He Never Mentioned Love|Claire Martin|Linn/Rockian AKD295

Claire Martin is maturing nicely, slotting the final pieces into place to fulfil her promise as an exceptional jazz singer. This Shirley Horn tribute has none of the dodgy song choices or show-off tendencies which compromised her otherwise impressive 2006 Australian performances. She is mastering that tricky artistic conundrum of playing to her strengths while simultaneously expanding her horizons. She is also learning that less is more; that a suggestion of an emotion left hanging in the air can be more telling than pile-driving it home to the listener.

The *If You Go* is a highlight, Martin’s dark voice lit by an aura of quiet desperation as she pleads her case with her lover, shaded by Gerard Presencer’s lonely flugelhorn. Even when she slides toward cocktail jazz, she oozes quality and charm.



Between The Lines|Knut Rossler/Johannes Vogt/Miroslav Vitous|ACT/Planet 9463-2

Seventeenth-century French lute music seems an unlikely bedmate for jazzy improvisation, yet here it is, made up with what soprano saxophonist Knut Rossler and lutenist Johannes Vogt rather charmingly call “paraphrases” of early pieces by Ennemond Gaultier, Denis Gaultier, Pierre Dubut, and Jacques Gallot. To these are annexed improvisations which are rarely directly based on the compositions, but are opportunities to extend the works beyond their almost perfect elegance. The beautiful melodies are kept chaste by the lute’s clean formality, and the improvising has another jewel in the crown besides Rossler’s ravishing forays on soprano and alto flute: brilliant bassist Miroslav Vitous is present on most pieces, and has never been better recorded. His instrument will shake the foundations of your home, even as the music rattles the flimsy wall between genres.