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Mustang Speaker

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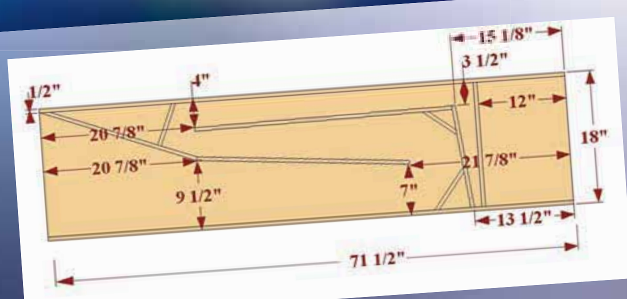
A LARGE AND LOW HORN SPEAKER

ACTIVE CROSSOVER AND SUB FOR AUDIO SYSTEM

A SUB MAKEOVER



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 By Claude Goeuriot

The Mustang Speaker

You can use this low-cost, quality speaker for computer audio or to replace the speakers usually bundled with compact hi-fi systems.

Mustang, a small, agile loudspeaker.

For my choice of a woofer, I started by searching for a wide band driver such as the Tang-band W3-871 or Fostex FE-103, but then abandoned these options for the following reasons:

- The W3-871's cutoff frequency was too high for my tastes (~130Hz) for such a small speaker.
- I questioned the bandwidth quality of the Fostex FE-103 for high-frequency response (check out the speakers designed by Troels Gravesen that use a FE-126).

I went back to a 10-13cm woofer plus tweeter solution. The website Zaph Audio (http://www.zaphaudio.com/) provides a lot of information on drivers, their performance, and their application, in addition to their remark-

PHOTO 1: SEAS ER15RLY woofer.

PHOTO 2: SEAS 22 TAF/G tweeter.

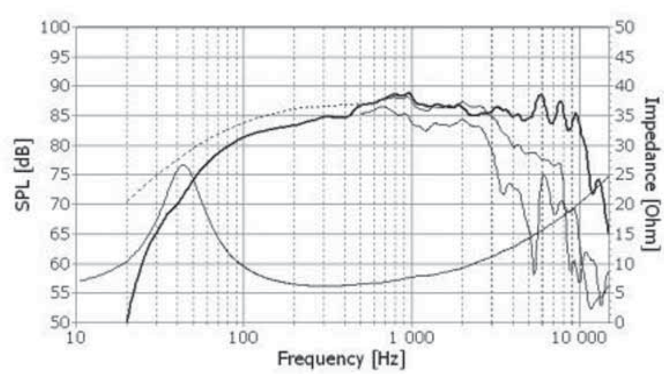

FIGURE 1: SEAS ER15RLY woofer.

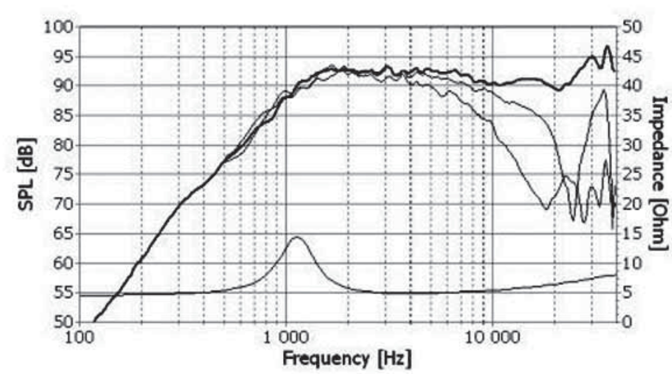

FIGURE 2: SEAS 22 TAF/G tweeter.

able DIY designs. If I set aside, for economic reasons, high-end drivers such as Scan-Speak, SEAS Excel, or Audio Technology, my choice would be SEAS Prestige drivers, which seem to be a good value (**Figs. 1-2** and **Photos 1-2**).

The bandwidth of the two drivers must be compatible with a crossover frequency between 2 and 4kHz, and permit simple filtering because the bandwidth overlap of the drivers is quite large. You will see further on, however, that the diffraction due to the baffle heavily disturbs the 1 to 3kHz zone. Because of its characteristics, the woofer (QTS = 0.324) is intended to be used in a bass-reflex design.

CABINET SIZING

Several freeware packages are available on the Internet to help designers choose the volume of the enclosure dedicated to the woofer, including:

- WinISD (http://www.linearteam.dk/)
- Unibox (http://audio.claub.net/software/kougaard/index.html)

With Unibox (**Fig. 3**) you get a 7 ltr volume and a cutoff frequency of about 58Hz with a port of 5cm diameter and a length of 22cm. This is in bass-reflex for a Butterworth response type without a peak in the low frequencies.

The results are confirmed by SoundEasy (**Fig. 4**), which is the software I generally use (and that I heartily recommend despite its high cost when compared to other products such as "Speaker Workshop") for measurement and filter design. Small variant: a slightly larger volume (9 ltr) and 17cm reduced length for the port,

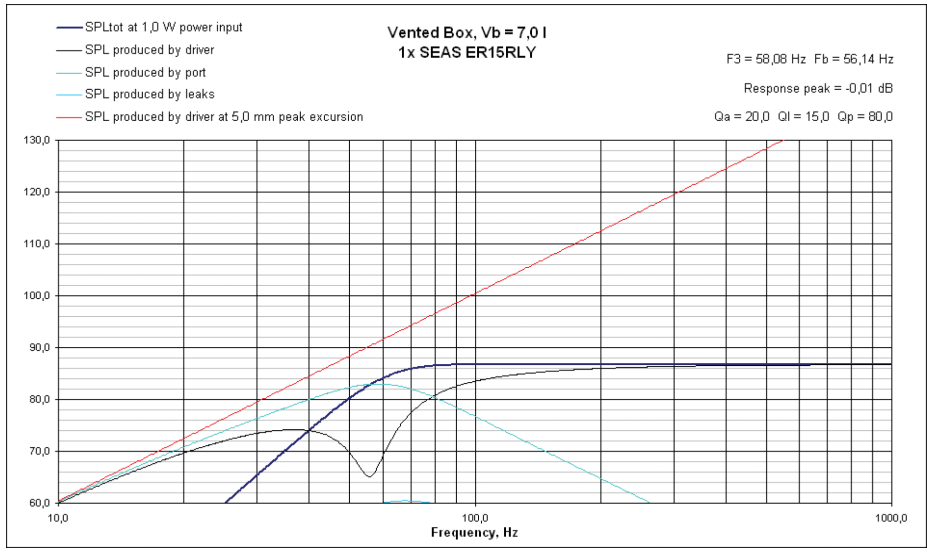

FIGURE 3: Low-frequency domain calculated with Unibox.

FIGURE 4: Low-frequency domain calculated with SoundEasy.

which is easier to install in a small cabinet, but at the expense of a slightly higher F3 frequency (60Hz). To conclude, I aim for a global volume of about 10 ltr, taking into account the additional space needed for the speakers, filter, and port.

BOX DRAWINGS

Because of its small size, the cabinet is a simple box to build, using six pieces of 19mm (3-4″) MDF, without any internal brace. The prototype I built differs from the plan because it has a removable front to allow easy testing. It is screwed to a front panel of identical size (**Photo 3**).

Due to its size, the port installs vertically and exits under the cabinet, which rests on 4cm rubber feet. The tweeter is offset to minimize the diffraction due to the baffle at high frequencies. These schematics (**Fig. 5**) represent the right-hand speaker, with the left-hand one being symmetrical (tweeter on the right-hand side).

The filter mounts on the back of the cabinet (not shown on the plan). At the beginning of the project, I wanted to use veneer as the finish for the cabinet, so none of the corners are rounded. Aesthetics came first, at the expense of a less linear response curve of the drivers in the box (see measurements).

CONSTRUCTION

The sides and rear of the enclosure are simply attached with wood glue and held in place with clamps while the glue cures. Before gluing, you need to prepare the rear of the box, drilling holes to allow the filter, which is wired on a printed circuit of 100 × 160mm, as well as the self-adhesive bituminous panels that damp the MDF panel. These bituminous panels are used to damp all sides of the box, except the front face.

Photo 4 shows the enclosure during construction with the backplate fitted to carry the front panel. The cabinet is then veneered. To do this, I chose poplar burr, instead of the more expensive walnut burr. Compared to traditional veneering, this requires a lot of coating and sanding after gluing. I then sealed the cabinets with several thin coats of

varnish and a final polish to achieve a semi-gloss finish.

The most difficult part of the woodworking is the front panel, which you must carve to fit the drivers. You need a router as well as a compass at hand (such as the Jasper Jig, www.jaspertools.com) to design rabbets. Note that the rear cutting for the woofers is also scalloped to give the most space possible to ease airflow behind the cone.

The sides of the speaker, with the exception of the front and the rear, are covered with an absorbing tissue (Resobson®) that is found in car accessories shops or online stores. It comes in the form of 1m2 sheets. The port is wrapped with insulating wool that is easily wound around it (**Photos 5-7**). The cabinet mounts on 40mm rubber feet to give sufficient room for the port to breathe (**Photo 8**).

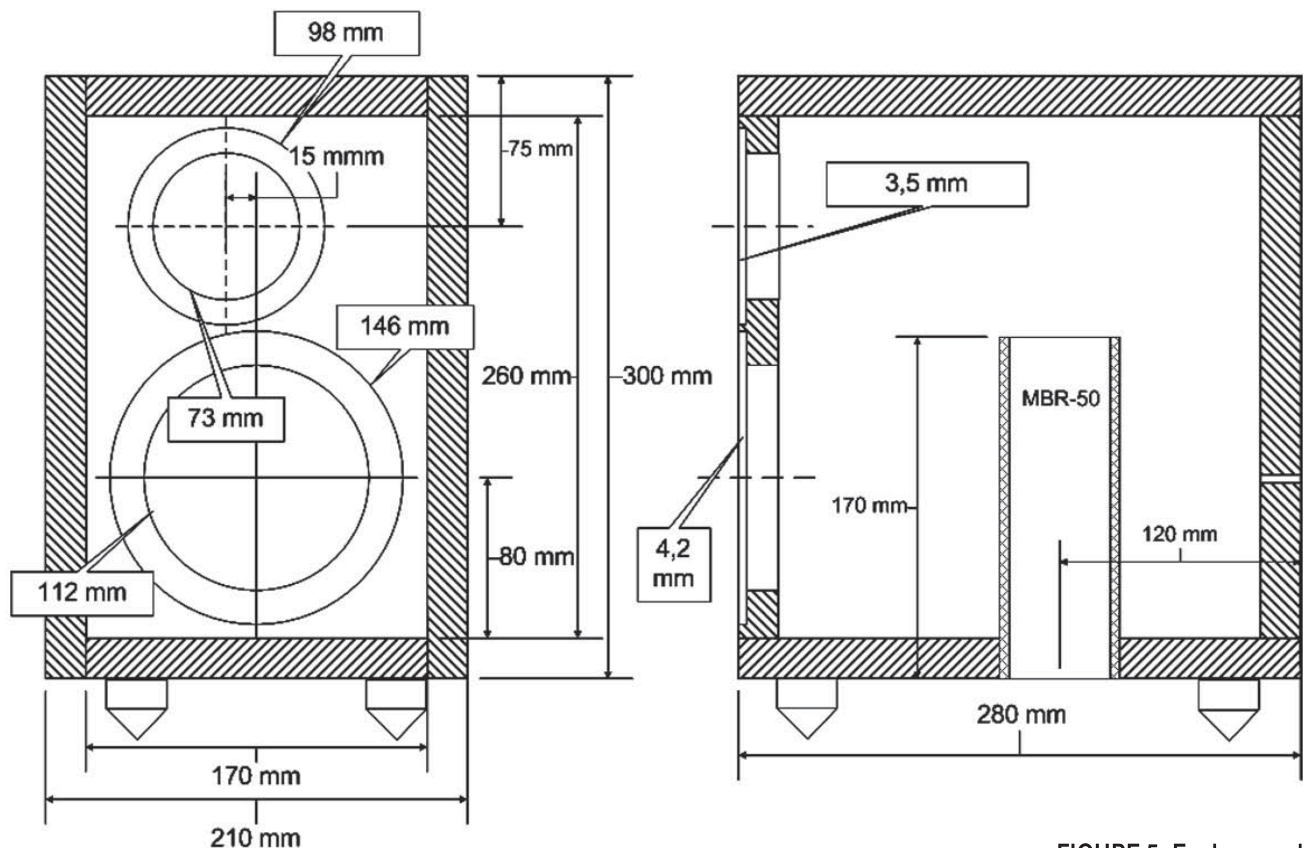

FIGURE 5: Enclosure plans.

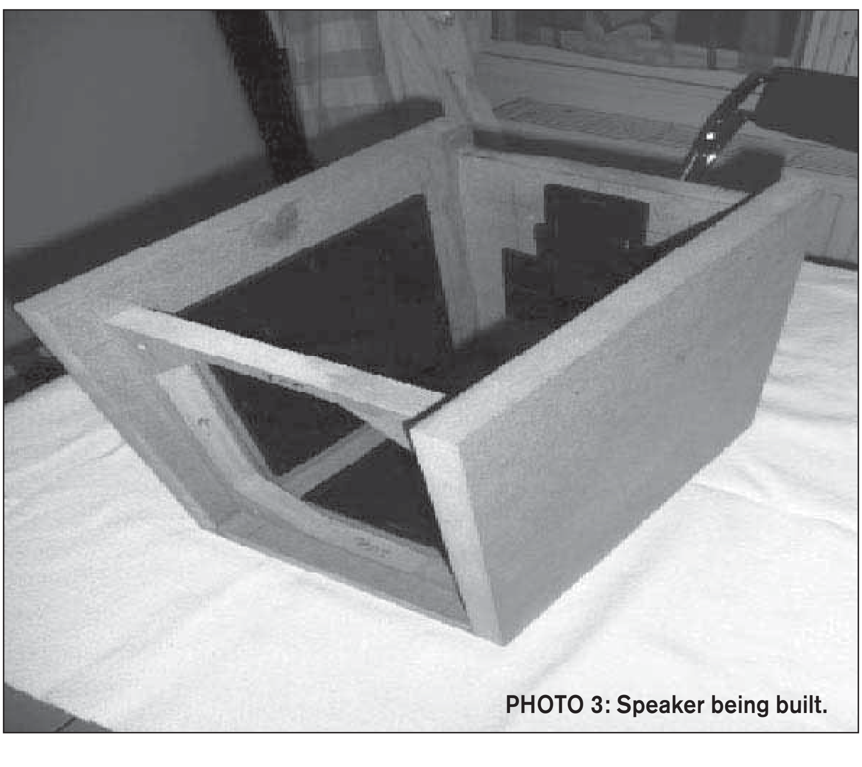

PHOTO 3: Speaker being built.

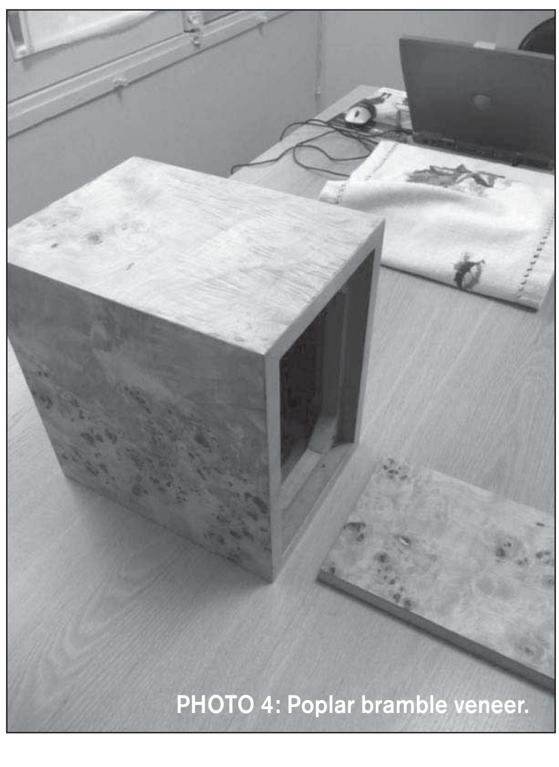

PHOTO 4: Poplar bramble veneer.

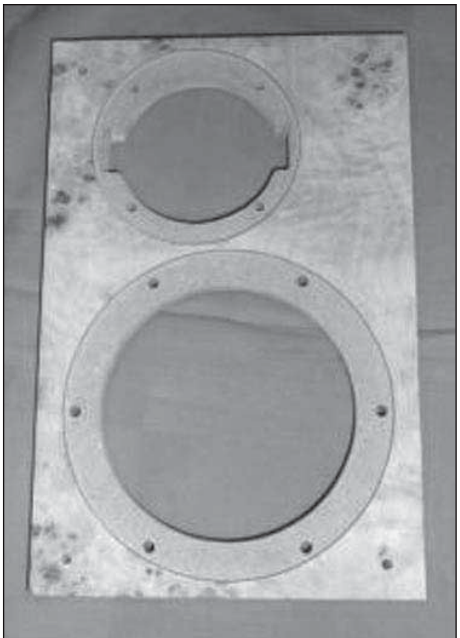

PHOTO 5: Front.

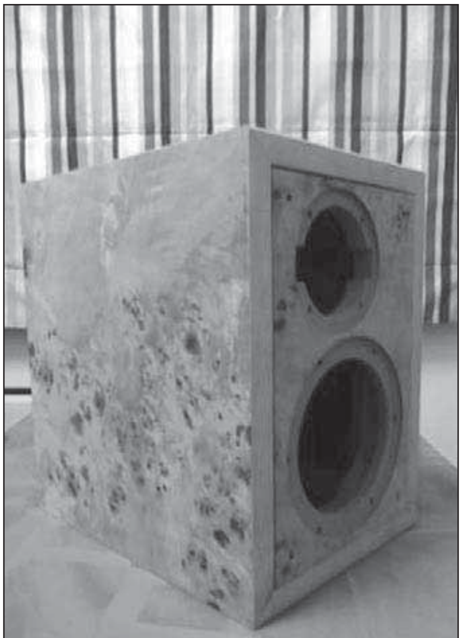

PHOTO 7: Woodwork finished.

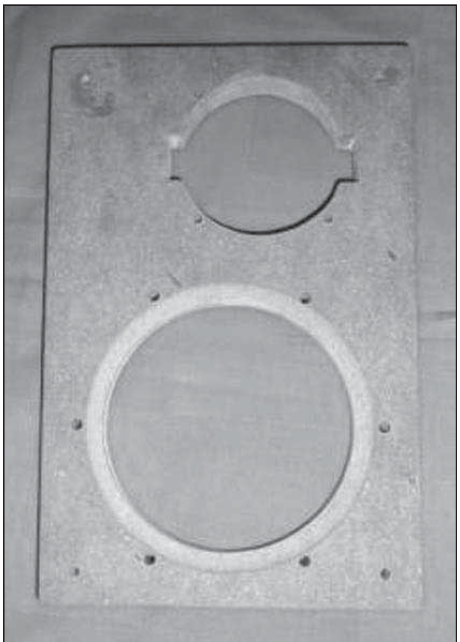

PHOTO 6: Back.

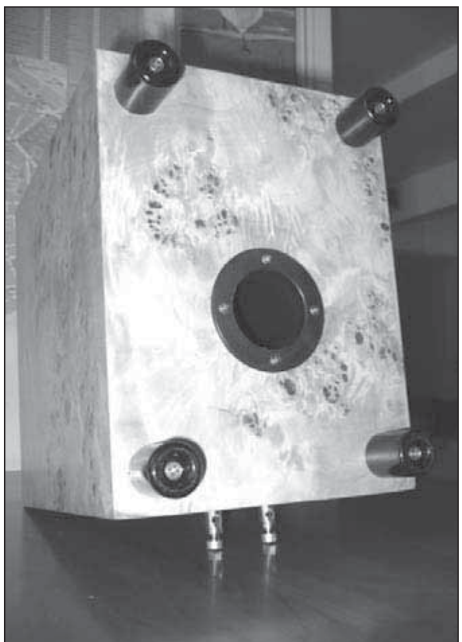

PHOTO 8: Underside.

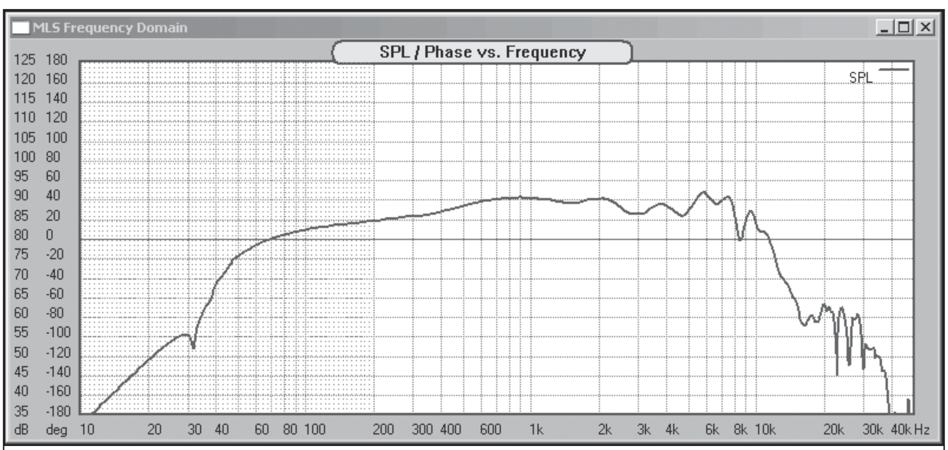

FIGURE 6: ER15RLY loudspeaker domain.

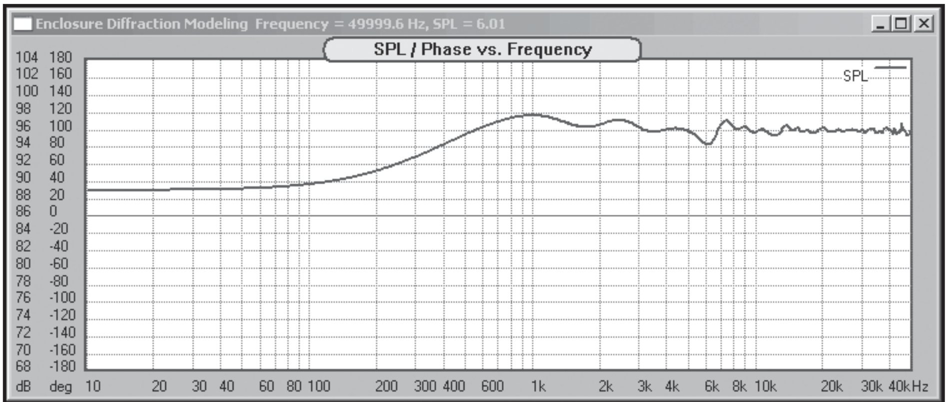

FIGURE 7: Influence of the baffle on the frequency response of the woofer.

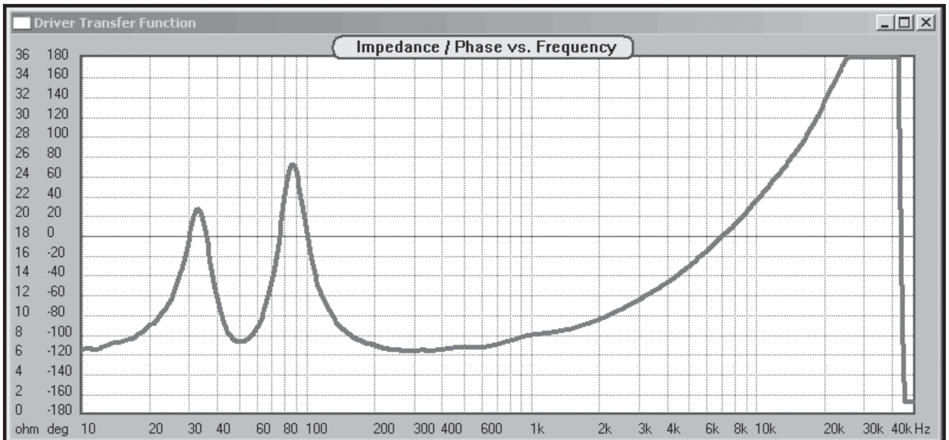

FIGURE 8: Impedance of the ER15RLY loudspeaker.

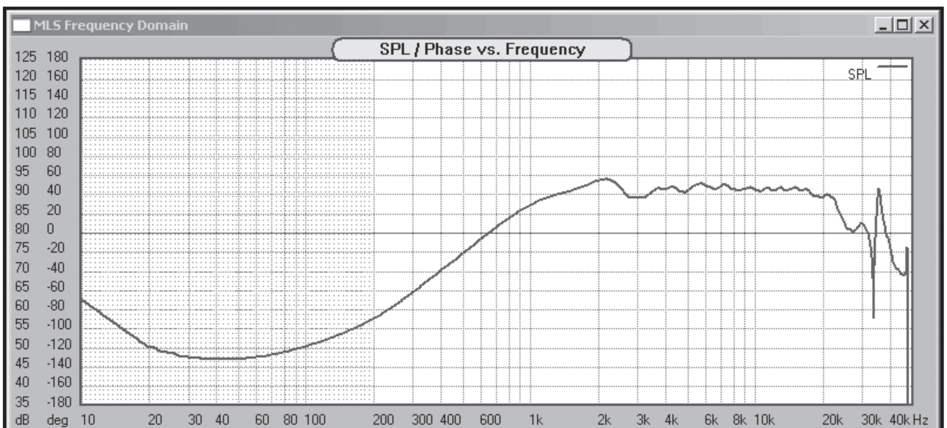

FIGURE 9: Frequency domain of the 22TAF/G tweeter.

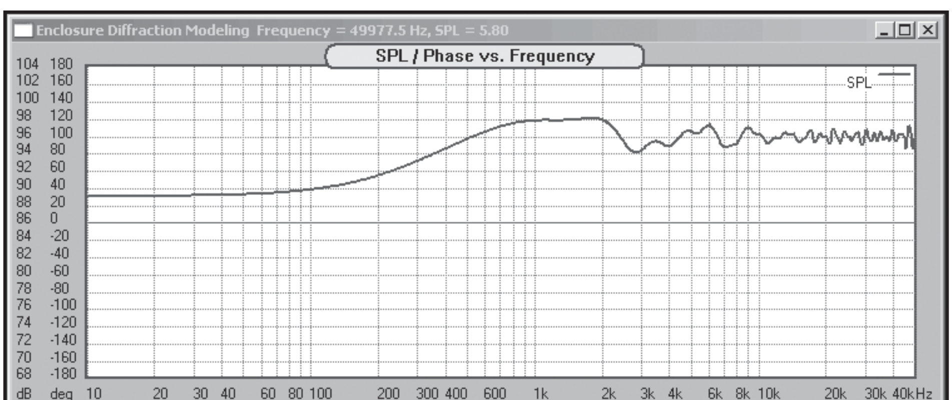

FIGURE 10: Influence of the baffle on the frequency domain of the tweeter.

MEASUREMENTS

I performed measurements using SoundEasy software, along with an EMU 0404-USB sound card, a calibrated Behringer ECM8000 microphone, and a DENON PM520 amplifier.

ER15RLY. The domain diagram (**Fig. 6**) is the combination of the far-field measurement and of the near-field measurement (speaker and port), taking into account the baffle diffraction, which shifts the radiation angle of the speaker from 4 Pi radians to 2 Pi radians between 200Hz and 1kHz.

The influence of the baffle on the frequency response is illustrated in **Fig. 7**: You can observe a 6dB gain between 100Hz and 1kHz, followed by ripples up to 8kHz. The diffraction diagram is drawn using SoundEasy software based on the location of the loudspeaker on the front face and of the dimensions and shape of the baffle.

The impedance response is usual for a bass-reflex speaker, without any remarkable anomaly that would point to a defect in the design. The drop at 50Hz is as expected (**Fig. 8**).

22TAF/G. Next is the tweeter. You can see the influence of the baffle on the frequency domain (**Fig. 9**) by noticing a bump in the response at 2.2kHz. This is consistent with the simulation of distortion due to the baffle, followed by small ripples. Disregard the measurements above 20kHz, because the microphone I used (Behringer ECM 8000) cuts off quickly above these frequencies.

As a reminder, the far-field measurement takes into account the diffraction due to the baffle. You can see the peak between 500Hz and 2kHz due to 4 Pi to 2 Pi radians radiation angle, followed by scrambled signals (**Figs. 10** and **11**).

The maximum impedance is 9Ω at 1.3kHz. This differs in width with SEAS data, but remains coherent with Zaph Audio findings. With all this data, you can now go on to the next step: filter design.

FILTER DESIGN

I generally aim to realize acoustic filters of the 4th-order Linkwitz-Riley type. The combination of a lesser order LC filter and the natural response of the loudspeakers generally enables you to achieve the desired result without complications (**Fig. 12**).

The choice of the cutoff frequency also depends on the distortion that extends beyond 4kHz and below 2kHz for the tweeter. For the loudspeakers, the Zaph Audio website is very useful to pinpoint the truly useful bandwidth for them.

Taking these criteria into account, I selected a 3.5kHz cutoff frequency. The SoundEasy software enables very rapid filter simulations and optimizations. After a few hours of work, you obtain the following filter (**Fig. 13**).

For the woofer, use a second-order cell which both realizes the cutoff around 3.5kHz and, thanks to the 2.2mH inductance, compensates for the loss of low-frequency efficiency due to the 4 Pi radians radiating (**Fig. 14**).

For the tweeter, you also come to a second-order high-pass C1-L2 cell followed by a R1 resistance that equalizes both loudspeakers. This inductance is shunted by C3 to compensate for the drop in low frequencies beyond 10kHz (**Fig. 15**).

The simulated response of the speaker is depicted in **Fig. 16**. By inverting the tweeter profile, you can observe a dip in the response around the cutoff frequency. This is typical of a Linkwitz-Riley filter.

Finally, the graph in **Fig. 17** traces the response in the vertical and horizontal axes at the cutoff frequency. The ideal listening position is slightly below the measurement axis which was positioned on a level with the tweeter. The dips at ±22.5° represent loudspeaker/microphone angles or distances that produce a 180° phasing null, which is also typical of LR4 filters.

FILTER

The filter is on a breadboard PCB (100 × 160mm), which you can then install on the rear panel of the speaker (**Photo 10**). The installation is very easy because there are very few components compared to the available

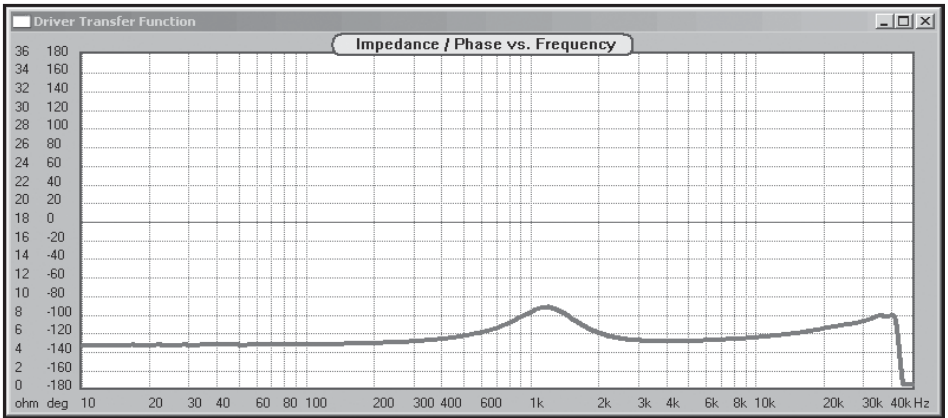

FIGURE 11: Impedance of the 22AF/G loudspeaker.

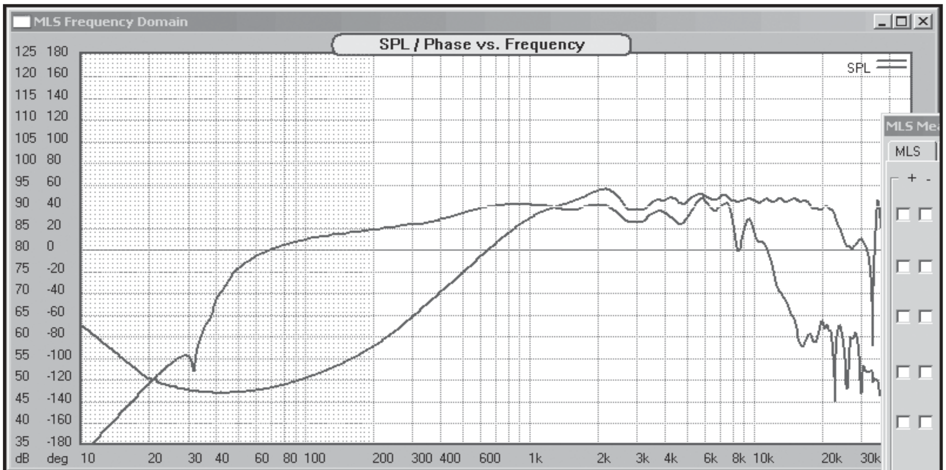

FIGURE 12: Overlapping responses of the two loudspeakers.

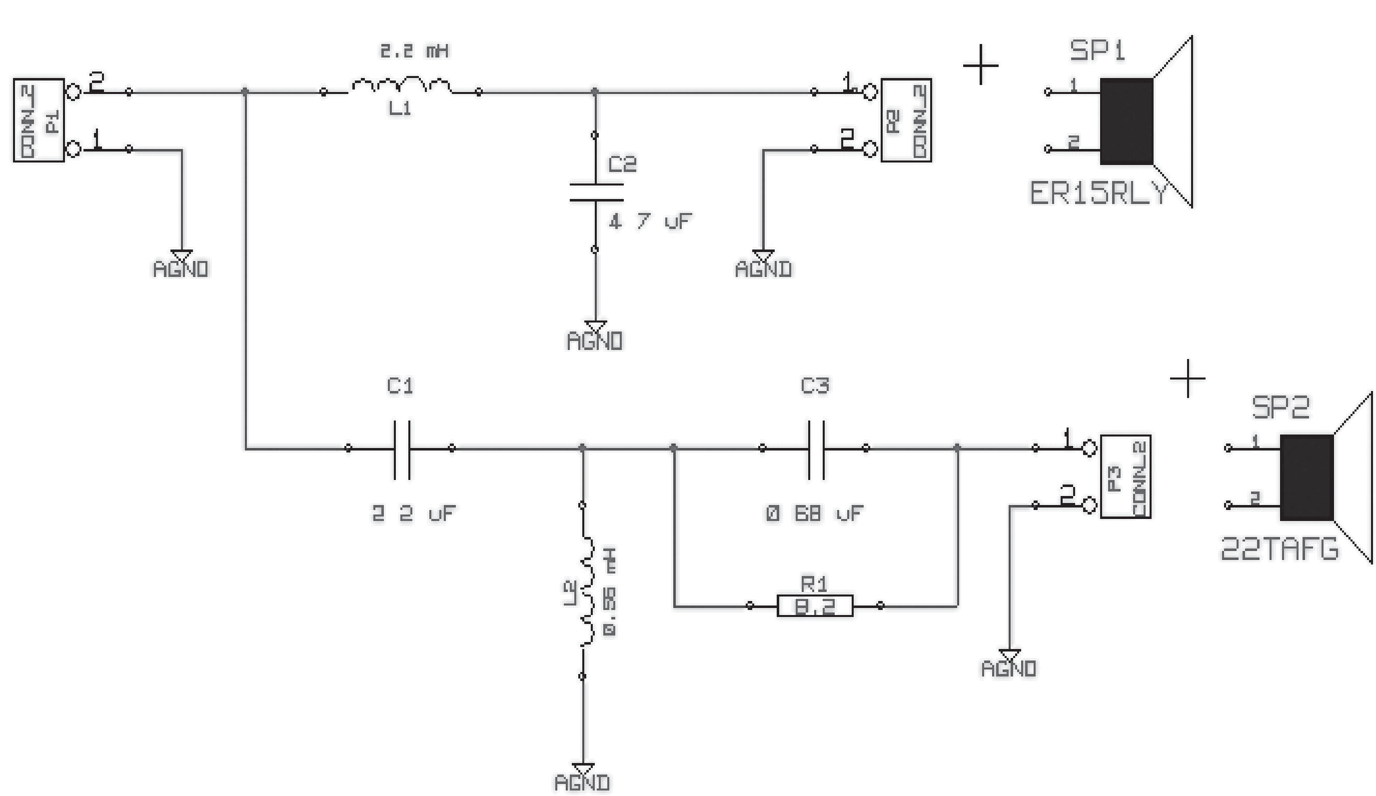

FIGURE 13: Mustang filter.

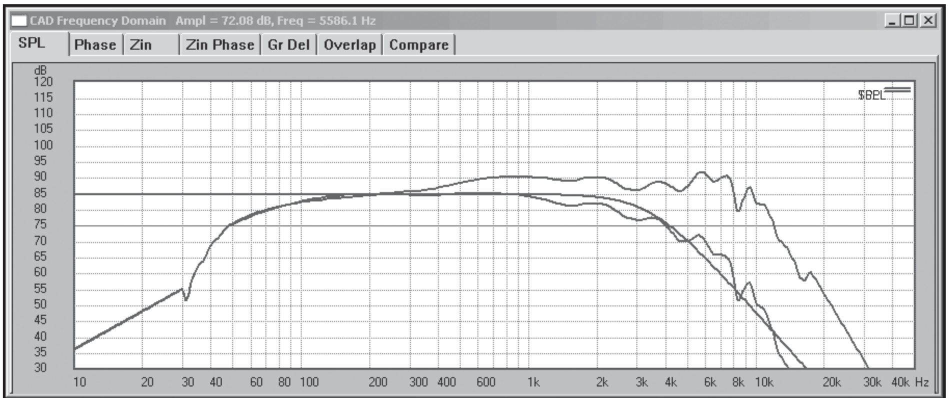

FIGURE 14: Filtered frequency domain for the woofer.

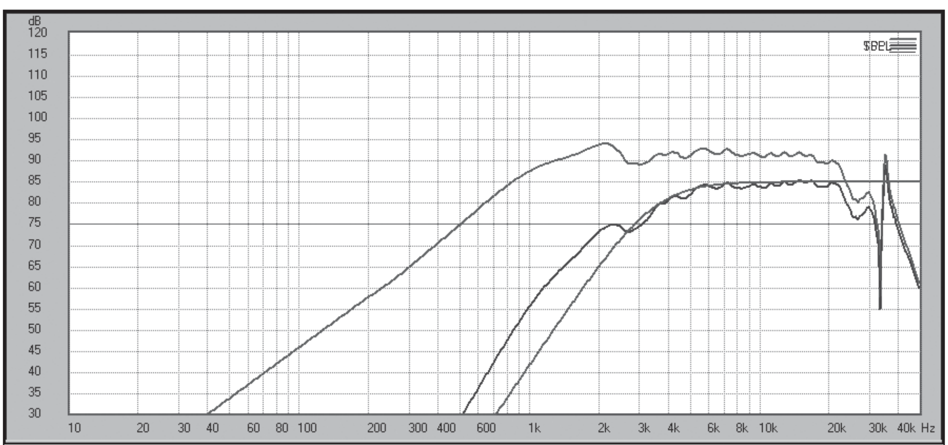

FIGURE 15: Filtered frequency domain of the tweeter.

PCB surface. Solder the components onto the PCB and connect them using point-to-point wiring.

SPEAKER MEASUREMENTS

The diagram in **Fig. 18** shows the frequency response of the speaker which is flat within ±1.5dB from 100Hz to 20kHz. The measurement is the result of a mix between the far-field response and the sum of the near-field response of the loudspeaker and port.

Figure 19 shows the far-field response of the speaker (ignore the response below 200Hz), with and without inverting the tweeter profile. The reality is not quite as good as the simulation, the drop being only –20dB deep, but it is still a pretty good result. The difference is probably due to the tolerance of the components. It's worth noting the measurement was performed before installation of the capacitor C3, and therefore with low-frequency attenuation beyond 10kHz.

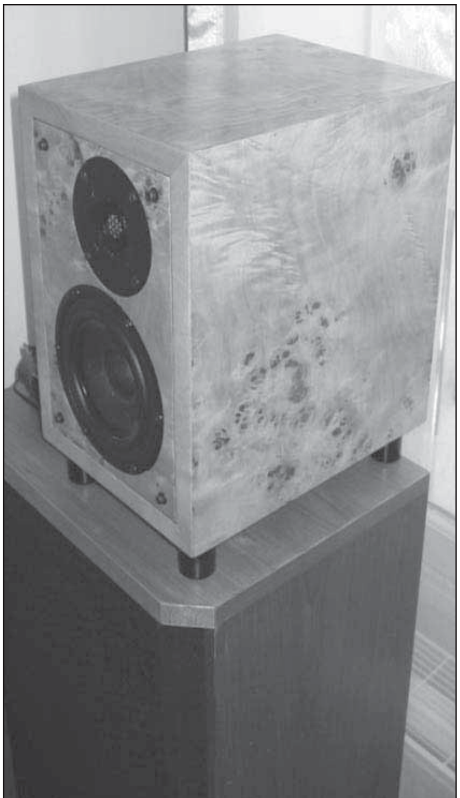

PHOTO 9: A Mustang twin sister (note the tweeter on the right of the vertical axis).

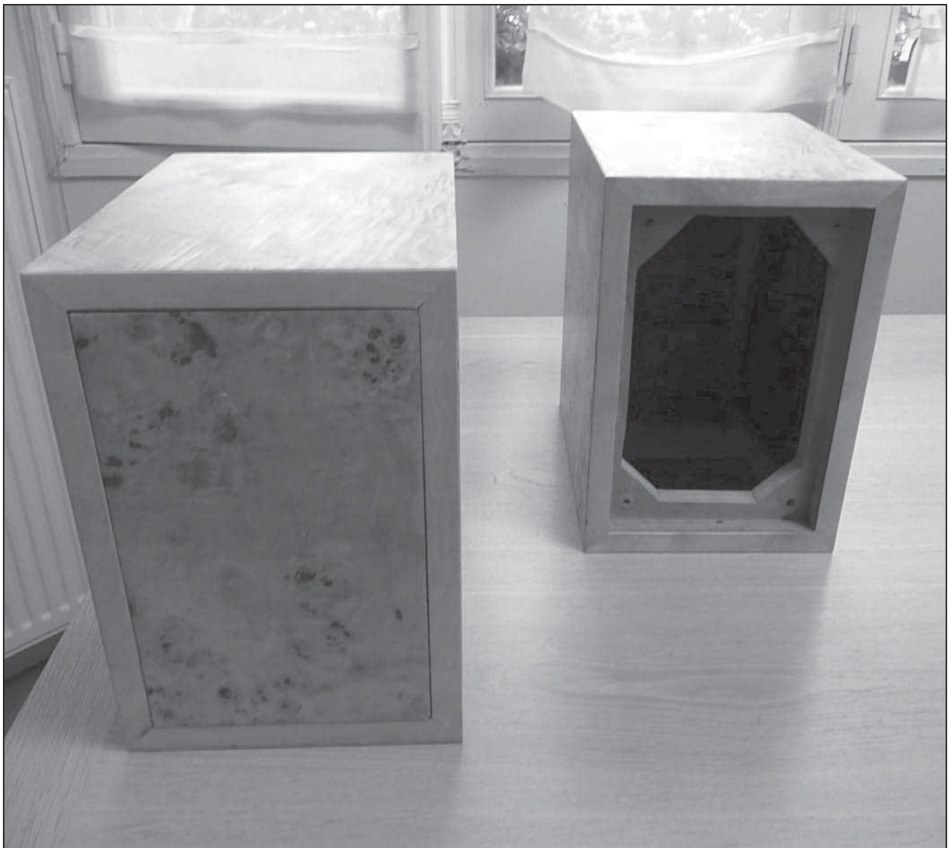

PHOTO 11: A couple of Mustangs being built.

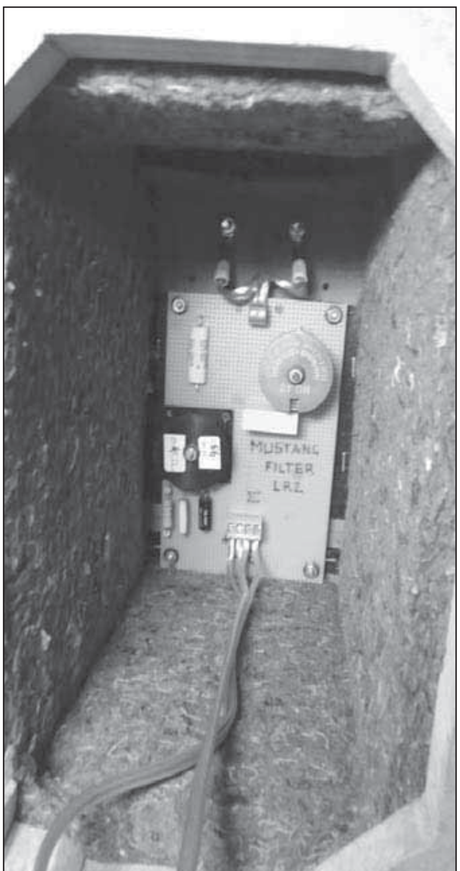

PHOTO 10: The filter, installed on the rear panel.

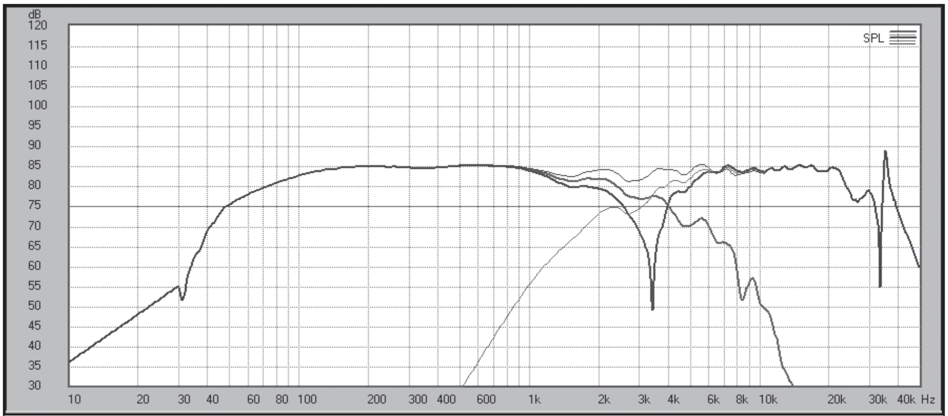

FIGURE 16: Simulated composite response for the speaker.

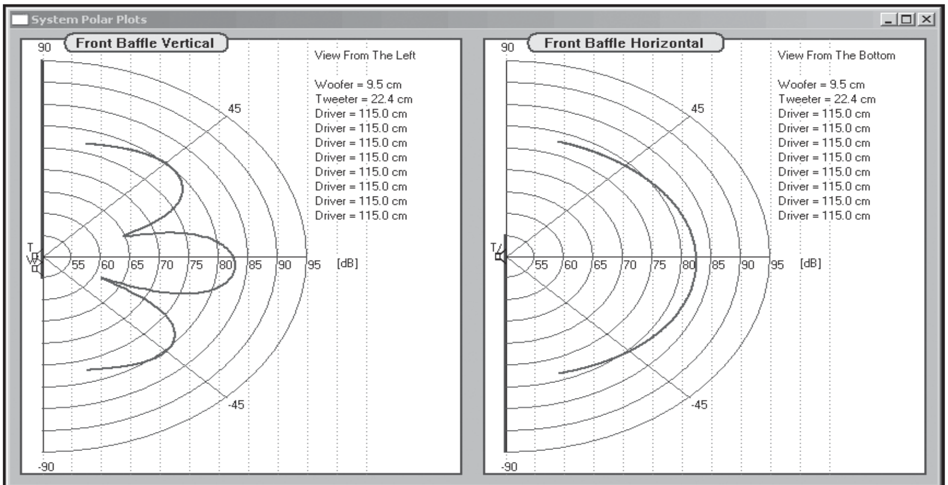

FIGURE 17: Speaker response in the vertical and horizontal axes.

CONCLUSION

This small speaker delights my ears during the increasingly numerous hours I spend in front of a computer screen. Definitely not the solution for a large room, but more than adequate for a student bedroom or home office.

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- *Loudspeaker Design Cookbook*, Vance Dickason: www.audioxpress.com/bksprods/BKSLOUDES.htm

Hardware

- ECM 8000: http://www.behringer.com/
- EMU 0404 USB: http://www.emu.com/

Software

- WinISD: http://www.linearteam.dk/
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- Mustang: http://www.homecinema-fr.com/
- Newbinette: http://www.newbinette.com/

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I would like to thank my son Florent, who did the first French to English translation, and my colleague and friend John Gerrity, who improved the English version and made many valuable comments to correct, simplify, or clarify the paragraphs that were unclear in the initial version. aX

FIGURE 18: Frequency domain of the Mustang.

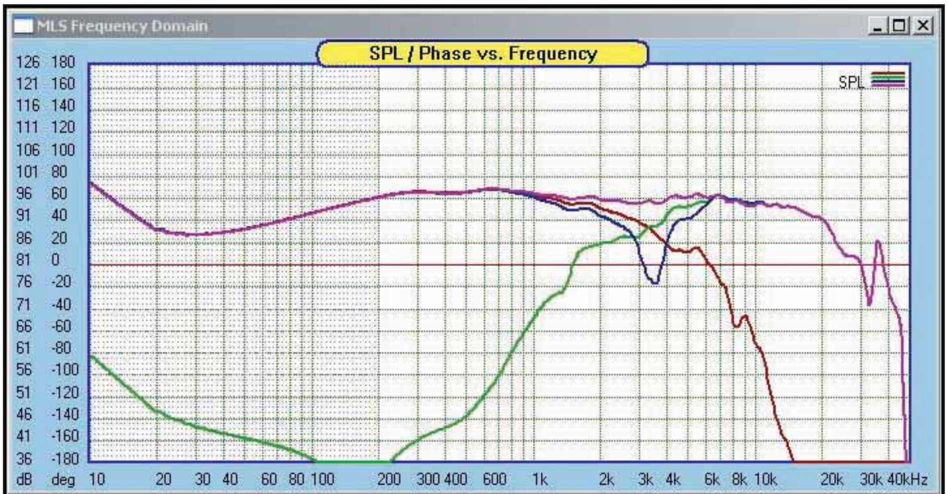


FIGURE 19: Far-field frequency response, with and without inverting the tweeter.

Low-Frequency Horn Speaker

Get down with this large horn speaker.

Popular music is typically produced using synthesizers that effortlessly cover the entire audio band down into infrasonic sound. The audio band is usually regarded in books and magazines as 20Hz to 20kHz, but the audio band is actually 30Hz to 15kHz. Below 30Hz, sound is easier felt or sensed than detected by the ear and requires high SPL for detection. You can sense a 25Hz signal at 130dB directly through air particle waves, but it is easier to sense them from floor or furniture vibrations acting as the medium for sound to travel to the body.

My objective here is to design and build a low-frequency (LF) horn speaker to play the lowest frequencies down to infrasonic sound. The horn must cover from 30Hz and above. To get to 30Hz using a bass horn requires large cabinets in multiples, which provides mutual coupling to make up for the required horn mouth. This is equivalent to using a corner or a wall, made popular by audio pioneer Paul Klipsch with his folded horn speakers.

The first step is choosing a woofer for the project. I selected an 18″driver, Celestion FTR-4080FD (www.celestion.com). I came across the FTR-4080FD Thiele/Small specification and was awestruck. It's perfect for horn loading at very low frequency or a low-tuned vented box. Contrary to marketing hype, using a small driver—less than 12″—for reproduction of very low frequencies defeats the purpose.

MEASUREMENTS

I built a LF horn around about 2000 from a design I created a few years prior. To plot the horn's expansion, I used a simple C program based on a Bruce Edgar design (www.edgarhorn.com). I started from the throat and folded the horn with 180° bends until it reached the cabinet outer dimensions. The horn's outer dimensions were limited to the trap-door height of my basement work space.

Figure 1 shows the horn CAD draw-

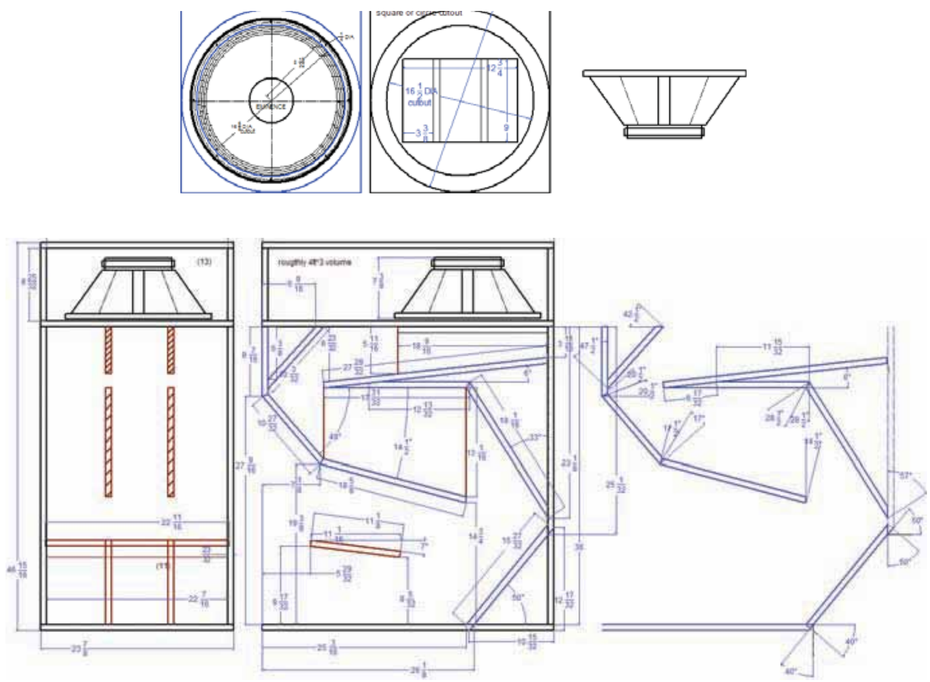


FIGURE 1: A 30Hz horn design using 180° duct bends. Left is front view, middle is side, and right is horn duct showing the 180° angles. The driver baffle mounting board is shown at the top.

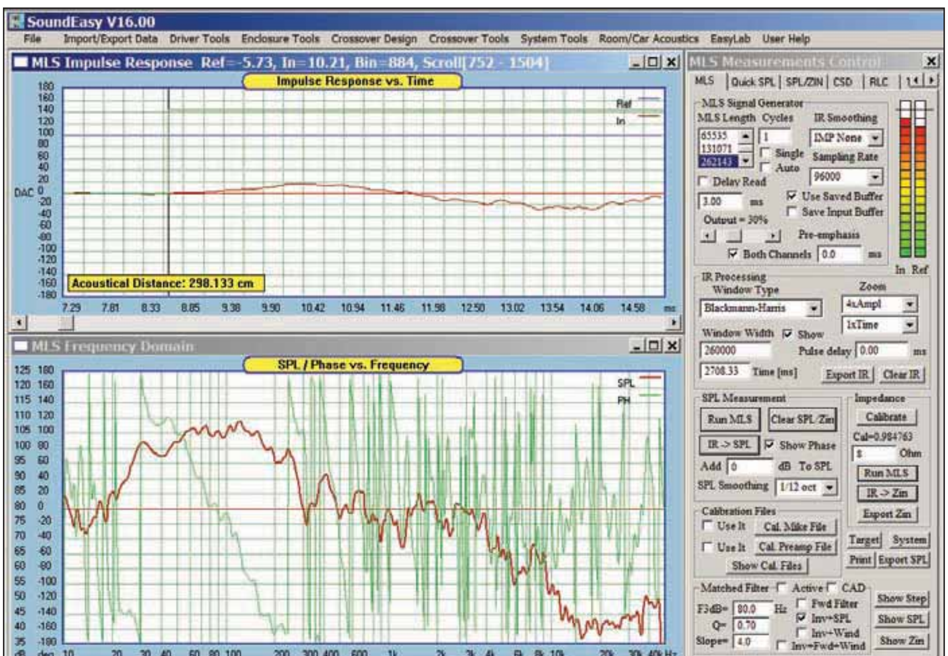


FIGURE 2: The impulse response (top plot), frequency, and phase response (bottom plot) of two cabinets measured outdoors, on the ground and side by side. The frequency is essentially flat between 47Hz to 160Hz, with a 7dB dip at 40Hz and then rising to a 4dB hump at 30Hz.

ing. The external dimensions are 47″H × 24″ W × 36″ D. The cabinet has two 180° bends to attempt to unwrap a 30Hz folded horn. The horn duct uses diagonal reflectors in the corners. Stiffening braces placed in the horn duct eliminate unwanted vibration. The rear cabinet volume is $4ft^3$ (113 ltr).

One flaw is that the volume or space directly to the rear of the driver is not big enough to provide cooling of the voice coil vent. I cooked a few voice coils before realizing the design flaw. At drawing time, I didn't consider a vented woofer, which is now ubiquitous.

Figure 2 shows the impulse response and frequency response of two cabinets measured outside, on the ground and side by side. The frequency band is relatively smooth from 55Hz to 160Hz. Notice the dip at 40Hz, and the hump at 30Hz as the response rebounds. The horn mouth is foreshortened, the mouth and cabinet front face dimensions are not enough area to provide proper loading to 30Hz on a ground position.

Figure 3 shows the dual cabinets' impedance response. The impedance peaks are flattened due to reactance annulling at the throat. The 30Hz hump in **Fig. 2** is a sign the horn cutoff or tuning is 30Hz. So here you have very good loading at the throat, but its mouth is too small to load a 30Hz wavelength at the ground position.

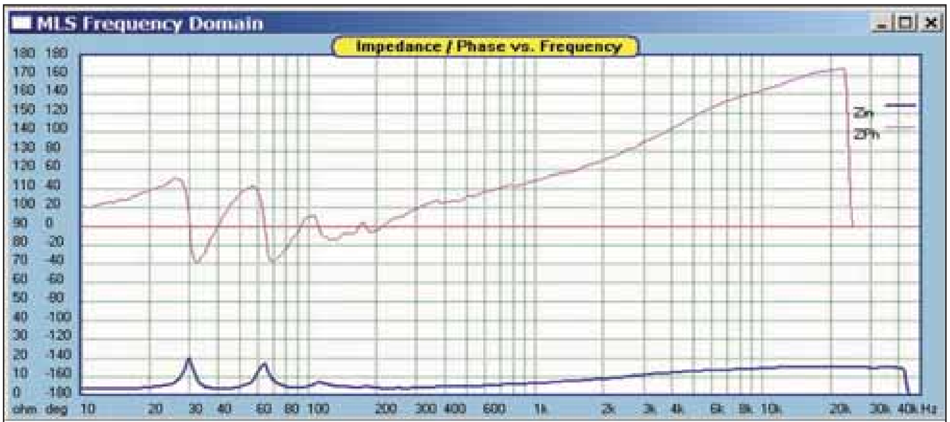


FIGURE 3: Two cabinets in parallel mentioned in *Fig. 2*, impedance response.

I moved the two cabinets to a wall position—that is, the backs against the outside wall of a house—to measure the frequency response and impedance. **Figure 4** shows the measured response is now within ±3dB from 30 to 160Hz. You can expect this result when the speakers are placed indoors near a wall. The 20dB null at 300Hz is due to placing the driver at a midpoint of the rear chamber's dimension. Analogous to cabinet diffraction, placing the driver off-center will cure the 300Hz null. Because this null is outside the passband, the anomaly is of no concern.

One side note: Placing the driver off-center is critical if designing for midrange. This is a well-known design issue, but through my own tunnel vision I failed to quickly identify the problem in my *audioXpress* article, "Designing a Midrange Horn" (March 2009). **Figure A** is from the Harry F. Olson book *Acoustical Engineer-*

FIGURE 4: Two cabinets placed against a wall, outdoors, measured impulse (top plot) and frequency (bottom plot) response.

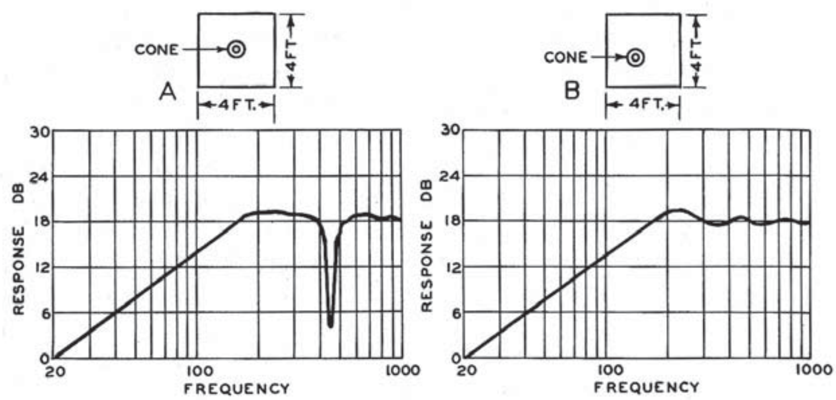


FIGURE A: Harry F. Olson book, Acoustical Engineering, Figure 6.20. "In B the speaker is mounted unsymmetrical to eliminate interference."

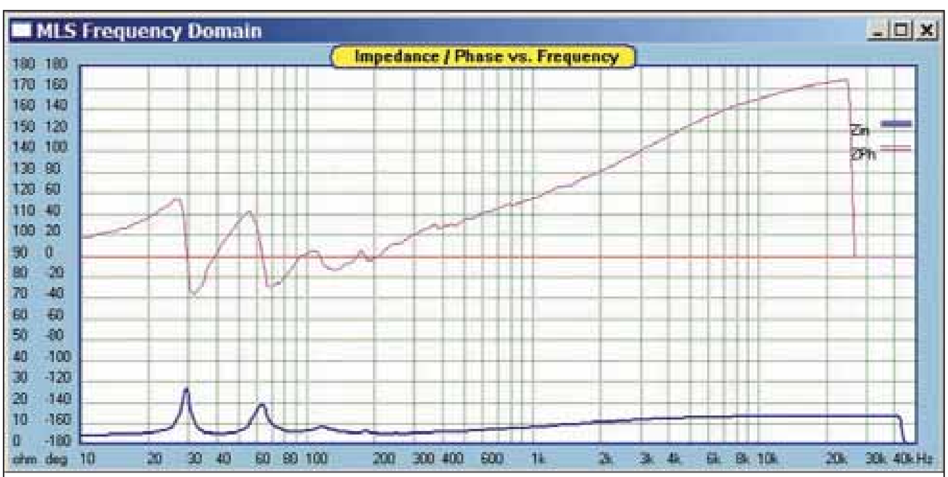


FIGURE 5: Two cabinets against a wall, outdoors, measured impedance response.

ing (www.audioXpress.com), Figure 6.20. The book documents the dilemma.

Figure 5 is the measured impedance response of the wall position. The only discernible change is the impedance phase at 36Hz and slight increase in impedance magnitude from 20Ω to 26Ω.

When I was fortunate to move my work space from a basement to a garage, I set out to build a no-holds-barred very low frequency horn. I used my custom horn calculator, Hyperhorn, to crunch the math and Delta CAD to lay out the horn drawing. (Hyperhorn is freely available from the web (http://djgroundbass.com), and Delta CAD (www.deltacad.com) is available for a small fee.) The math for the horn design is shown in **Fig. 6**.

For an eighth space (corner) horn with an expansion of 30Hz hyperbolic, the calculator shows a throat of 81.5999 in^2, a mouth of 2055 in^2, and rear volume of 123.668 ltr. Already, it became clear that multiple cabinets are needed to meet the 2055 in^2 mouth requirement for a corner horn. Furthermore, four cabinets are needed if the wall position is used.

I cross-checked the design using a freely available horn calculator, David McBean's Horn Response Analysis Program version 14.00. McBean's calculator (www.dmcbean.bigblog.com.au) accepts metric units and calculates an exponential expansion, not the hyperbolic expansion previously mentioned. There is a small difference in length for exponential and hyperbolic expansion—295cm axial length versus 337.82cm axial length, respectfully. McBean's exponential horn calculator plots are shown in **Fig. 7**. The frequency response is modeled for eighth space (corner) position.

FINAL DESIGN

I used a final cross-check—SoundEasy from Bodzio Software (www.interdomain.net.au)—to predict the frequency response of the horn cabinet for eighth space (corner position) and half space (ground position) horns. By setting the horn's length, **Fig. 8** simulates the two horns' lengths frequency response. It shows that when the horn's length and mouth are taken to the intended size, the low-frequency response becomes flat all the way down to the cutoff frequency.

Armed with the simulation information, I drew the no-holds-barred horn

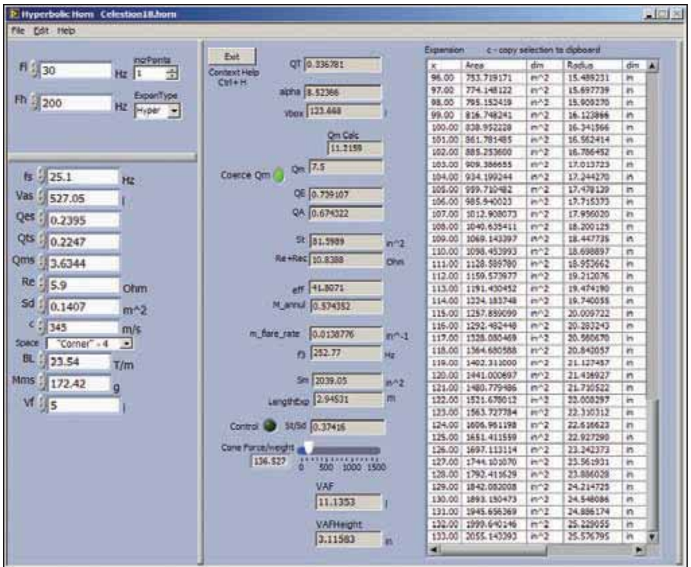


FIGURE 6: Horn expansion is 30Hz hyperbolic, a throat of 81.5999 in^2, a mouth of 2055 in^2, rear volume of 123.668L for an eighth space (corner) horn.

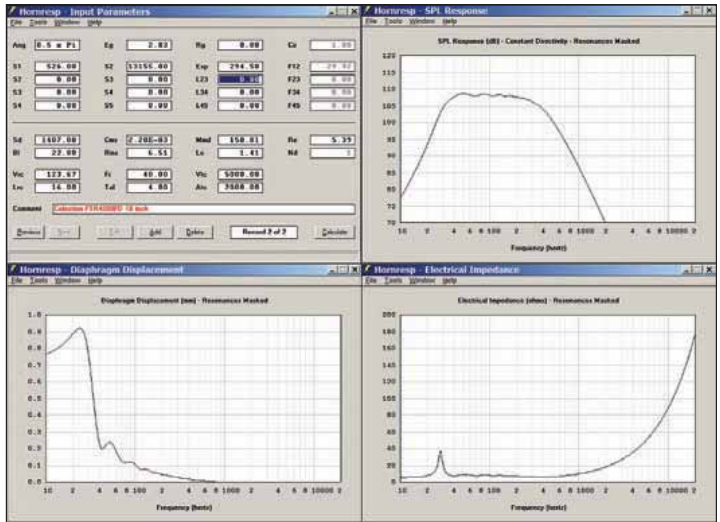


FIGURE 7: From left to right, McBean's horn input parameters, SPL response, diaphragm displacement, and electrical impedance. When you input figures for the throat (526cm^2), mouth (13155cm^2), exponential length (294.5cm), the program calculates a flare cutoff frequency of 29.92Hz.

using Delta CAD. **Figure 9** shows a 30Hz horn with the Celestion FTR-4080FD 18″ woofer. The horn layout uses 90° bends and diagonal duct reflectors, includes enough space behind the driver for vent cooling, and features the entire front face as the mouth opening. Also included are cut-out handles and two rear castors to aid portability.

There is a point of diminishing returns when it comes to cabinet size. The larger the cabinet, the more bracing is needed to stop sound-robbing panel vibrations. **Figure 10** shows the cabinet's side and includes the placement of braces and additional dimensions and angle dimensions. The bracings are cut out with circles to reduce weight. All wood is ¾″ 13-ply Baltic birch.

Two cabinets side by side provides a mouth area of 2304 in^2, i.e., 48 × 24 + 48 × 24. You need two cabinets to meet a 1/8 space (corner) horn for a cutoff frequency of 30Hz! Also, the rear volume is oversized. The required volume is 123.668 ltr (4.36ft^3), and the drawn volume is 226.53 ltr (8ft^3). If acoustical tests indicate tuning

FIGURE 8: SoundEasy simulation of a 30Hz hyperbolic horn. At 30Hz the bottom line is the eighth space size horn simulation. The top line is half space size horn and has greater low-frequency output down to 30Hz.

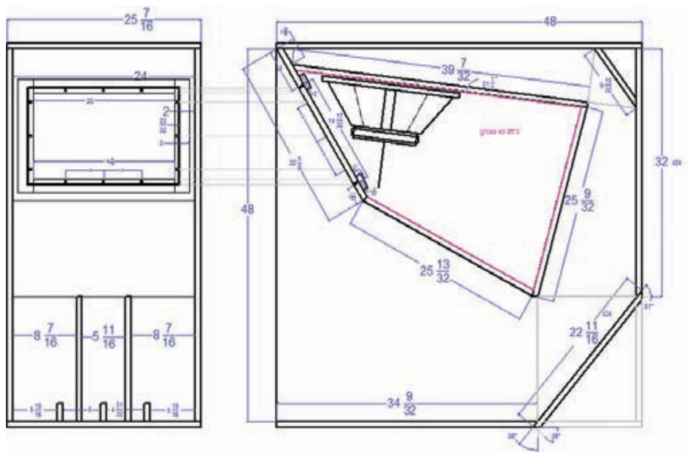


FIGURE 9: A 30Hz no-holds-barred horn drawn using DeltaCAD. The front face (left) and side (right) are shown. The front includes some stiffening braces of the bottom panels.

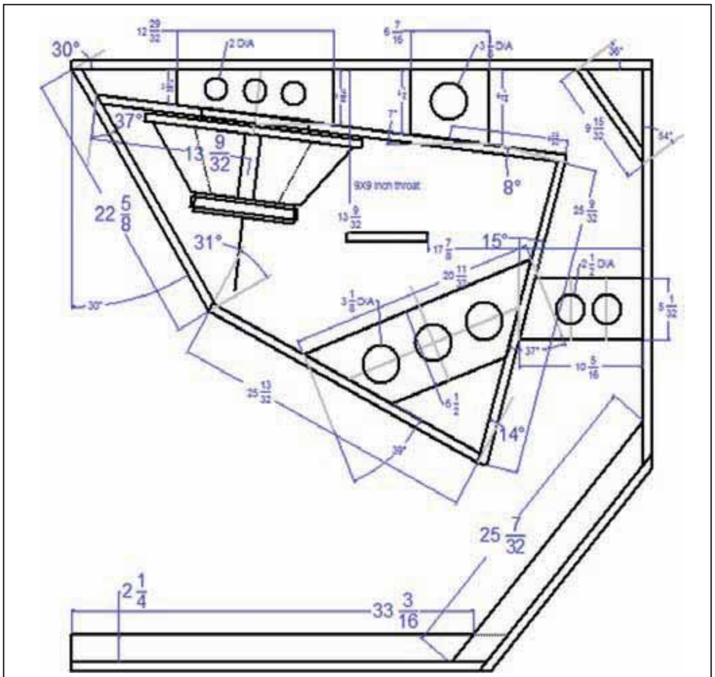


FIGURE 10: Cabinet's side that includes the placement of pairs of braces and additional measurements and angle dimensions. Circles are cut in the braces to reduce weight.

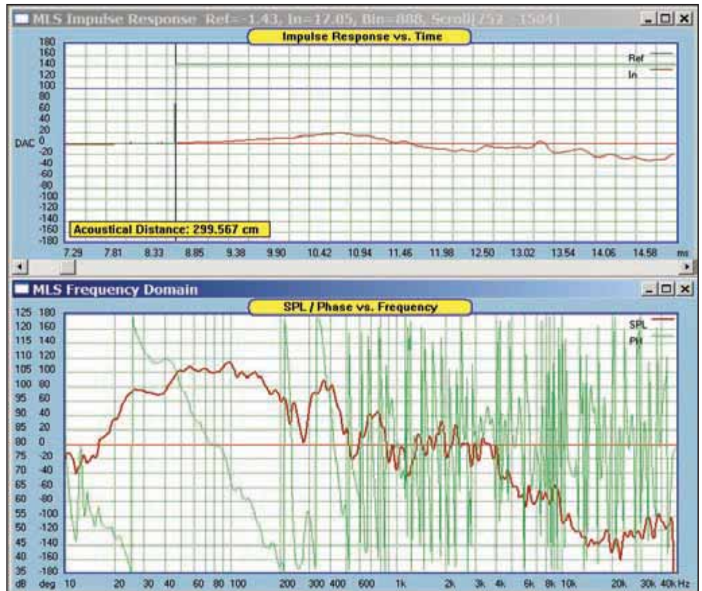


FIGURE 11: The impulse response (top plot), frequency, and phase response (bottom plot) of two cabinets measured outdoors, on the ground and side by side. From 47Hz to 150Hz, the response is flat with the exception of a +3dB hump at 98Hz. Below 47Hz a −6dB shelf is in effect down to 25Hz. Below 25Hz the response rolls off at 24dB per octave.

FIGURE 12: The impedance response of a pair of cabinets set up as in *Fig. 11*. The oversized rear chamber and the 25Hz driver resonance appears to load quite well from the flatter impedance magnitude.

FIGURE 13: The impulse response (top plot), frequency, and phase response (bottom plot) of two cabinets measured outdoors, back against a wall and side by side. Below 45Hz the bottom end is a −5dB shelf down to 25Hz, then the response rolls off at 24dB per octave.

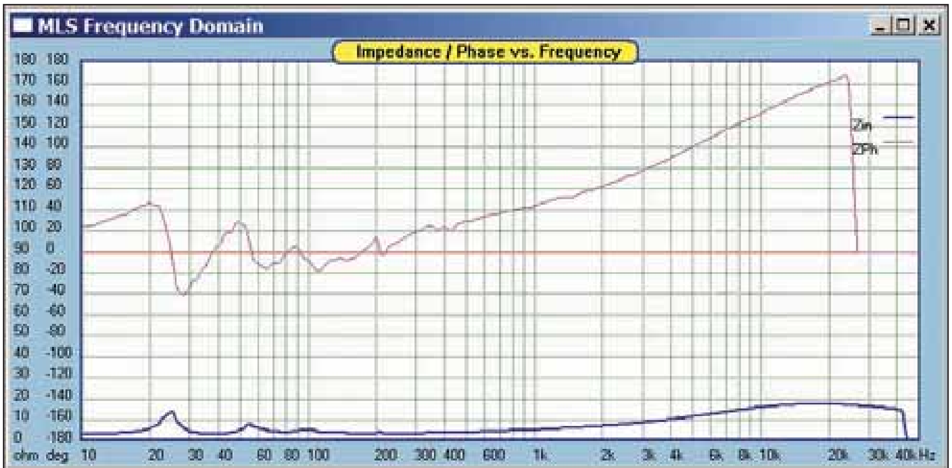


FIGURE 14: Impedance response of the measurement setup of *Fig. 13*. The wall position provides improved horn loading. All improvements are above 40Hz, where the impedance phase approaches zero.

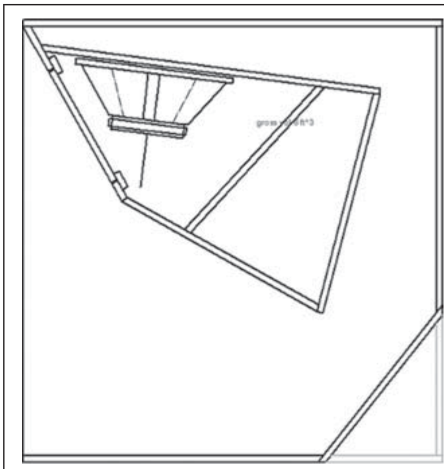


FIGURE 15: The rear volume is reduced, i.e., cut in half, by tack nailing ¾ MDF partition inside the rear sealed chamber.

is necessary, reduce the volume as needed.

TESTING

After building two cabinets, I began speaker testing. I put the cabinets through identical tests of the first horn design mentioned above. The two cabinets are side by side placed outside on the ground. **Figure 11** shows the frequency response.

From 47Hz to 150Hz the response is flat with the exception of a +3dB hump at 98Hz. Below 47Hz a -6dB shelf is in effect down to 25Hz. Below 25Hz the response rolls off at 24dB per octave. Nulls, out of band, occur at 275Hz and 538Hz. The placement of the driver at a midpoint on the baffle of the rear-sealed chamber exacerbates the nulls or box nodal modes. Again, the nulls are of no concern because they are out of band.

Figure 12 is the impedance response of the measurement setup of **Fig. 11**. It's clear that the horn's reactance annulling is in effect from the flat impedance peaks. The horn path and rear chamber combination are an improvement from the first design. The CAD drawing rear chamber is double the target size, and the driver's natural resonance is 25Hz, i.e., below the 30Hz design cutoff frequency. There are no large impedance peaks, so the combination appears to load well.

Continued on p. 26.

FIGURE 16: Rear volume is reduced to 4ft^3. Impulse response is shown in the top plot. Bottom plots are wall position frequency (top line) and ground position frequency response shifted −20dB (bottom line) of two cabinets.

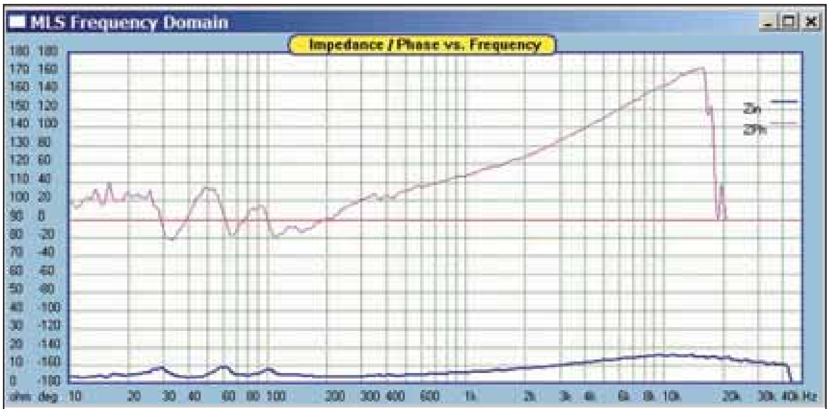


FIGURE 17: The impedance response of the reduced volume rear sealed chamber. Top line is impedance phase and bottom line is impedance magnitude.

Continued from p. 23.

I moved the test setup against the outside wall. **Figure 13** shows the wall position impulse and frequency response. From 45Hz to 150Hz the response is quite flat. Below 45Hz the bottom end is a smoothed -5dB shelf down to 25Hz. At 25Hz the response rolls off at 24dB per octave. The null at 275Hz persists, and the null at 538Hz has summed broader compared to the ground measurement.

Figure 14 shows the impedance response of the measurement setup of **Fig. 13**. As expected, the wall position provides improved horn loading. All the improvement is above 40Hz, where the impedance phase approaches zero (i.e., cancels). There appears to be no impedance changed below 40Hz.

Doubling the cabinets or moving to a corner increases output on the bottom end. The effect raises the frequency shelving to match horn loading that begins at ~45Hz. At this point, I decided to keep the design as is and use at least four or more cabinets when a corner is not available. Another option is to tune the rear-sealed chamber, i.e., reduce the volume thus increasing the cutoff frequency to eliminate the shelving of the low end. Remember the target cutoff frequency is 30Hz and not 25Hz, which is currently measured.

I reduced the rear chamber volume from 8ft^3 to 4ft^3 by tack-nailing ¾ MDF panel inside the rear sealed chamber. A drawing of the new panel, reducing the volume, is shown in **Fig. 15**.

I went back to test for measurements of the modified rear sealed chamber. **Figure 16** shows the frequency response of the wall and ground position measurement. The cutoff frequency is no longer 25Hz but 30Hz. Besides the higher cutoff frequency, all else appears to be identical to the original larger rear sealed chamber result.

The impedance response of the wall measurement setup is shown in **Fig. 17**. The phase line (upper line) shows "squiggles" from the 10Hz frequency mark because the new rear chamber partition is tack-nailed, and air leaks cause measurement anomalies. The impedance magnitude is flat and shows no major changes.

DESIGN CONSIDERATIONS

It's now obvious there is no benefit to reducing the rear volume to 4ft^3 from 8ft^3. What occurs is a reduction in bandwidth by 5Hz. Our ears hear on a logarithmic scale, so 5Hz is a big deal at 30Hz mark.

I decided the final design will use the larger 8ft^3 rear sealed chamber. What I dubbed the no-holds-barred horn actually tuned 5Hz lower than the target design, which is a major benefit. I'll need to use the cabinets in multiples (four or more) to get flat frequency response down to 30Hz or maybe even 25Hz. A theoretical possibility is using a corner, which would require only two cabinets. A test corner outdoors was not available to prove the theory.

This cabinet is large and requires generous bracing, especially for large panels where the sound exits the mouth. Wood tension strength is much greater on its edge dimension than its broad side. I took advantage of this by connecting braces on edge to the horn ¾″ panels, perpendicular to the panel's wood grain. **Photo 2** shows the horn during assembly and the stiffening braces of the horn duct.

Photo 3 shows the mouth braces and where I used ½″ birch glued to the ¾″edge braces that are glued to the mouth panels, perpendicular to the panel's wood grain. All edge braces are cut to a height of 2¼″. I will also install vertical and horizontal cross boards to further stiffen the mouth. A few stiffening boards installed at the mouth have no effect on the passband frequency response. If you intend on pushing the horns to the highest SPL, it is really important to install resonant-eliminating bracing.

Photos 4 and **5** show the cutout handles and castor placement. A single person can easily move this unit around by grabbing the cabinet by the handles, tilting the cabinet back on its wheels, and rolling it away. This works well because the edge of the cabinet is very strong, and I cut costs by using minimal hardware, castors, and metal handles.

If your listening room space is limited and infrasonic frequency is the goal, you're better off with a low resonance woofer of at least 15″ in a vented box tuned to a low frequency. If the room is large (say 100+ people capacity) and a corner or wall is available, then the bass horn will outperform all other types of speakers, plain and simple.

I hope I have provided a view into the horn size requirement to cover very low frequency sound. Could I have built something smaller for 30Hz? Sure, but my goal was a "no-holds-barred" very low frequency horn. I did not want to skimp on size requirements in order to get to the lowest possible frequency.

A large 8ft^3 rear chamber is necessary. I've shown the 4ft^3 rear chamber raises the cutoff. Any manufacturer who claims that a smaller designed rear chamber is a 30Hz frequency range is just not correct. Also, I've read about other DIYers who built bigger monster cabinets that don't go as low. If the rear chamber is not tuned low enough, no corner or stacking of cabinets will make the frequency range lower. On front-loaded horns, no rear chamber is used, the driver resonance effectively determines the lowest tuning frequency.

Bass horns sound smooth when properly built. The frequency response and impedance response above prove it. Manufacturers make claims of small cabinets—even tabletop models—that fill the room with concert sound using horn technology. That's simply marketing hype and not true. Some floor models and all tabletop models are more transmission line than are horn. aX

PHOTO 2: The stiffening braces of the horn duct.

PHOTO 4: Casters and cutout handles aid portability.

PHOTO 3: Mouth braces in which 2¼″ wide pieces are glued on edge to stiffen the mouth panel.

PHOTO 5: The cabinet tilts back and balances on two wheels.

The Low Profile Tuba

This makeover for a popular home-theater sub may be the right fit for your needs.

FIGURE 1: The Low Profile Tuba HT.

The original Tuba Home Theater subwoofer (*aX*, February '07 and Sept. '07) has proven very popular with the high-end home theater crowd, providing unexcelled SPL per dollar from its 22Hz folded horn. But the 36 × 36″ profile isn't a good fit for every room. In response to requests for a lower-profile version, I've refolded the original square shape into a rectangle (**Fig. 1**). I call this subwoofer the Low-Profile Tuba Home Theater, or LPTHT.

PARTS SELECTION

The suggested material for construction is ½″ plywood. You may use Baltic birch if you wish, but any good grade of void-free plywood with at least five plies will do. A 24.5″ wide cab requires about three 4 × 8′ sheets of plywood. You may build the cab in various widths.

You can build a 15″-driver-loaded cab from 18-36″ wide stock, using 17-35″ wide panels. You may use a 12″ driver in 15-30″ wide cabs, using 14-29″wide panels. Response is flatter and low-frequency sensitivity goes higher as the cab becomes wider, so make it as wide as is practical for your room. A good compromise uses 23.5″ wide panels, giving a 24.5″ finished width, for minimum waste with 4 × 8′ plywood.

If you can only fit an 18-20″ wide cab in your space, the Eminence LAB 15″ driver is preferred, because it has a heavy-duty cone that will stand up to the higher throat pressures of a narrower cab. Its specs are Fs 28Hz, Qts .35, Vas 103 ltr, Xmax 12mm. For cabs 21″ and wider the Dayton (Parts Express, www.parts-express.com) DVC 385-88, part #295-190, will give the best result. T/S specs are Fs 19, Qts .36, Vas 232 ltr, Xmax 15mm.

A suitable 12″ driver is the Dayton DVC 310-88, with Fs 22Hz, Qts .38, Vas 120 ltr, and Xmax 15mm. This driver makes sense if your room won't allow an 18″ wide cab, but at that point the advantage shifts to the 15″ driver. Of course, if you have room for an 18″cab or wider and already have an appropriate 12″ driver on hand, by all means use it.

This design is so strong that butt joints are quite adequate. You may secure the joints with screws; if so, you need a pilot/countersink bit and driver bit for your drill. Use drywall or cabinet screws, 1 and 1.25″ long. You may fasten the cabinet with 1 and 1 5/8″ ribbed shank paneling nails, using a nail punch to set their heads below the surface of the wood.

Another option is a pneumatic brad nailer, using 1 and 1.25″ brads. Screws and nails only hold the parts in place while the adhesive in the joints sets. Use a polyurethane base construction adhesive, such as PL Premium, applied

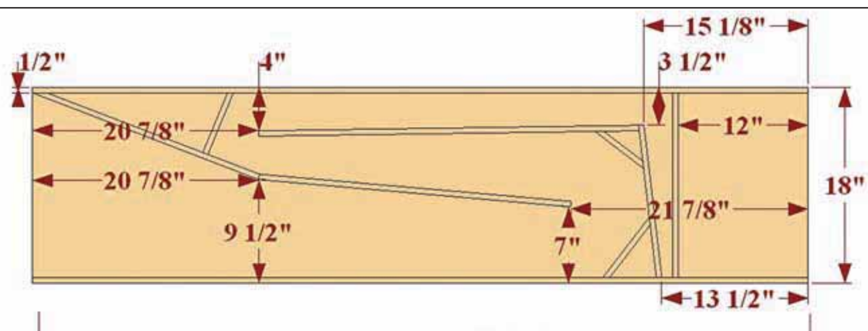


FIGURE 2: Major inner panel joints.

with a caulking gun. Polyurethane expands as it cures, filling gaps to ensure an airtight joint.

CONSTRUCTION

Cut out the sides. Draw the major inner panel joints on one (**Fig. 2**). The dimensions shown are to the edge of the side. Add the minor panel joints per **Fig. 3**. Write the assembly order of the panels per **Fig. 4** on the side. Drill two 1/16″ holes through the center of each joint, about 2″ from either end of the panel. Flip the side over and draw lines connecting each pair of holes. The lines show you where to drive your fasteners into the panels.

Clamp the two sides together and drill through the holes in the first side all the way through the second. Unclamp the two sides, draw connecting lines on the second side, so you'll know where to drive the fasteners when attaching the second side. If your plywood has a better side, make sure you lay out the sides so that both "better" faces are facing out.

The ten panels are the same width. **Table 1** shows the panel rough lengths, leaving some selvage should you wish to pre-cut them. Because there is no such thing as plywood that truly measures a half-inch thick, the actual finished sizes of all parts are figured by dead reckoning. Measure and then cut them to finished length as you install them to be sure of the final size required, using the panel figures as guides.

TABLE 1: ROUGH LENGTHS FOR PANELS

1. 18″	2. 72″	3. 72″	4. 15″	5. 8″
6. 35″	7. 6″	8. 30″	9. 23″	10. 7″

For perfect joints, clamp a straight 2 × 2 guideboard along the joint line, and then clamp the mating part in place. If you can't reach the guideboard with clamps, screw it in place instead, filling the screw holes with adhesive when you remove it. Adjust the part position until you get it right, remove the mating part, apply adhesive to the joint line, and then re-clamp the part in place. Drill pilot/countersinks, no closer than 2″ from the end of the boards, about 6-8″ apart.

In the middle of a panel make the countersinks just deep enough so that the screw heads don't stand proud, but on edges take them about halfway through the top sheet, so that you can round over the cabinet edges later without hitting the screws. Drive your screws, using 1″ where you don't want to penetrate too far, 1.25″ everywhere else. With nails or brads the procedure is the same, sans the pilots.

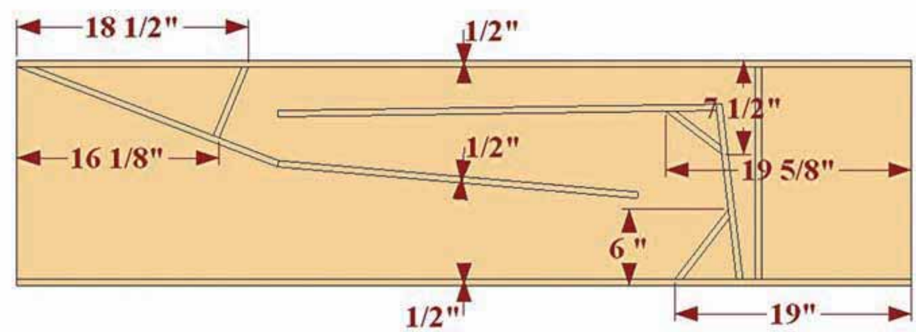


FIGURE 3: Minor panel locations.

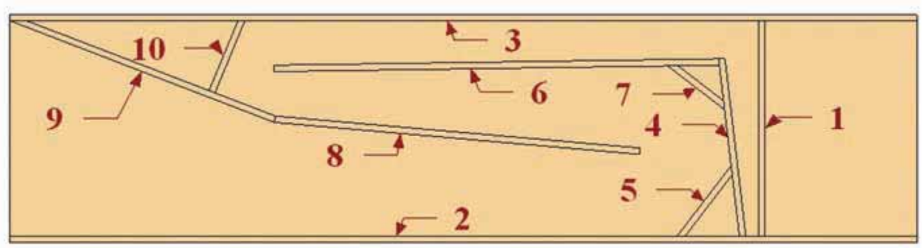


FIGURE 4: Assembly order.

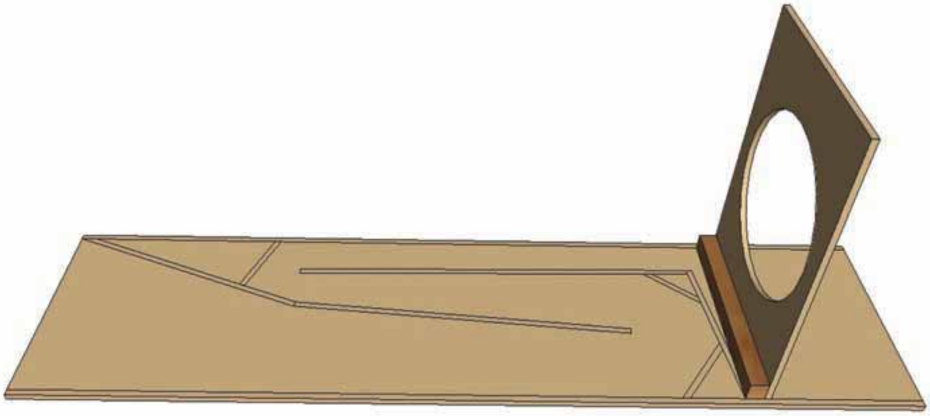


FIGURE 5: Panel 1 installation.

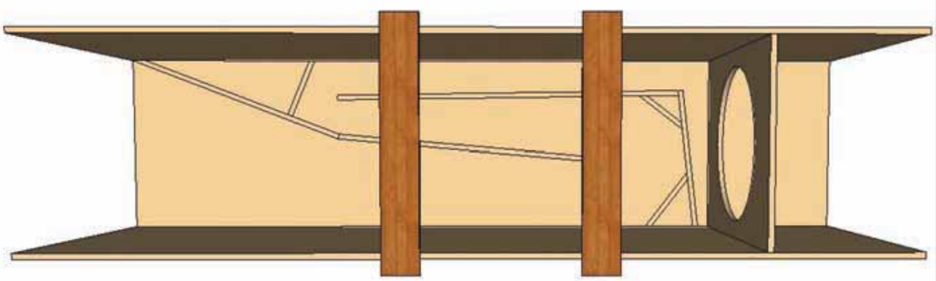


FIGURE 6: Top and bottom added.

For stability leave the guideboard in place until you attach the next part to the assembly. If your panels are warped, clamp or screw guideboards to the panel face to push or pull out the warp until the joint to the next panel is fastened. Some of the assembly figures show guideboards, but not clamps. They also show 3″ wide scrap plywood braces, temporarily screwed to the free edges of the panels to hold them in alignment.

Cut out panel 1. The driver hole is centered on it; size the driver hole per the manufacturer's driver datasheet, or an eighth of an inch smaller than the inner diameter of the frame gasket. The height of Panel 1 is that of the 18″ finished cabinet size less the thickness of the top and bottom. Two layers of half inch plywood is usually about 15/16″, so the height of panel 1 would end up at 17 1/16″.

Position the driver and use a nail or punch driven through the bolt holes to mark where to drill holes for fasteners. Fasten the driver with 3/16″ bolts and T-nuts or hurricane nuts. When using

FIGURE 7: Panel 1/4 braces.

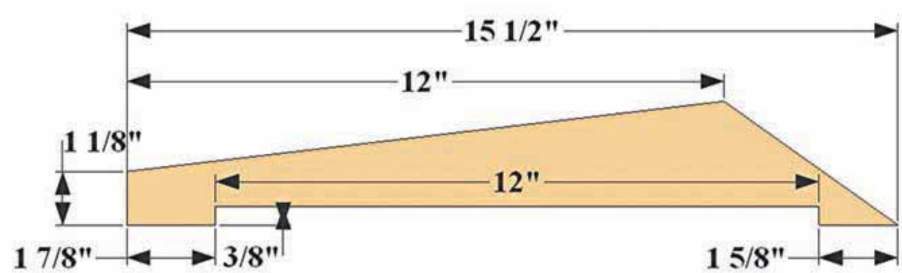


FIGURE 8: Panel 4.

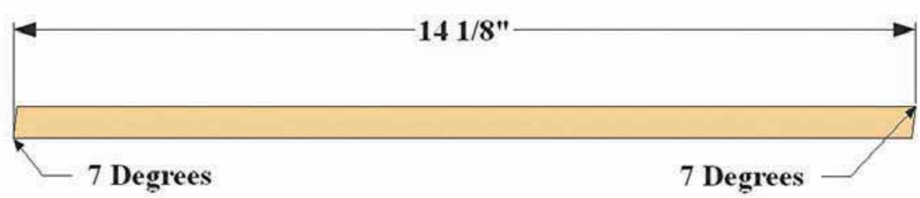
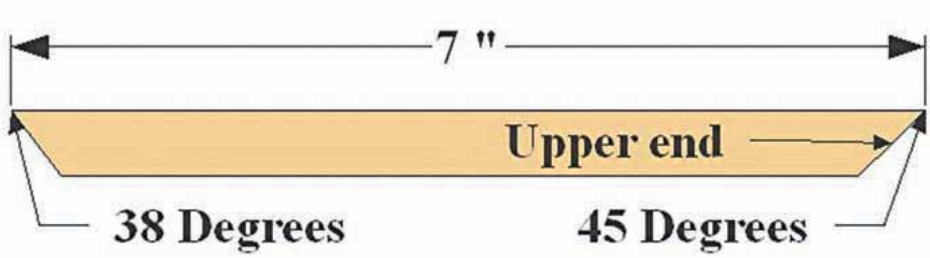


FIGURE 9: Panel 5.



blind nuts there often isn't enough wood on the baffle to seat them well near the baffle cutout. A way around this is to cut the hole with your jigsaw shoe set at a 30 to 45° angle, so the hole diameter on the opposite side of the baffle is smaller.

Coat the nuts with some Gorilla Glue or the equivalent before inserting, taking care not to glue the threads, chasing them with a tap if you clog them. The nuts install from the side of the baffle opposite the driver; use a bolt to pull the nut into place, hand driving it only, because a power driver can easily strip the threads. Allen socket-head bolts are best, because they can easily be driven by feel in tight spaces. Trial-fit the driver, bolting it in place to be sure everything lines up correctly.

INITIAL TEST

This is a good time to break the driver in. While the driver doesn't need to be broken in, it will initially work better if it is. Run a 25-30Hz tone through it, at 10 to 12V AC, for 12 hours or so. You should see the cone move about one-quarter inch. Doing this with the driver face-up on a workbench will make very little sound and also allow you to see whether the driver suspension hits the baffle in long excursions.

Lower the test tone to 10 to 15Hz to get maximum excursion from the driver, about a half inch. If you hear slap but can't tell where, turn off the amp and rub some chalk dust (from a carpenter's chalk line) on the inside of the driver cutout. Bring the signal back up until the suspension slaps, and the chalk dust will show you where it hits. Remove and store the driver.

Attach panel 1 to the side, leaving an equal amount of space above and below it where the top and bottom will fit (**Fig. 5**). Attach panels 2 and 3, the top and bottom. Stabilize the assembly with a couple of plywood scraps (**Fig. 6**).

ASSEMBLY

Braces which connect all the major panels may be made of one-quarter to one-half inch plywood. Space them no more than 8″ apart. A single brace is thus adequate for 15″ wide panels, two sets of braces for 18-25″ panels, and three sets of braces for wider than 25″.

Cut the panel 1/4 braces (**Fig. 7**).

FIGURE 10: Panel 6.

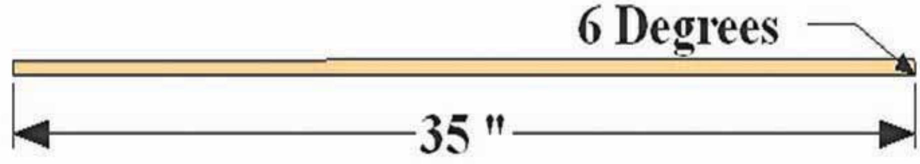


FIGURE 11: Panel 3/6 braces.

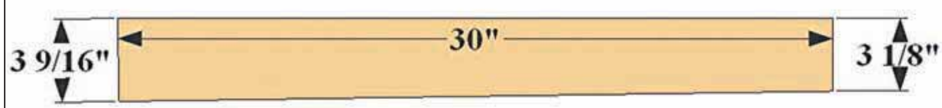
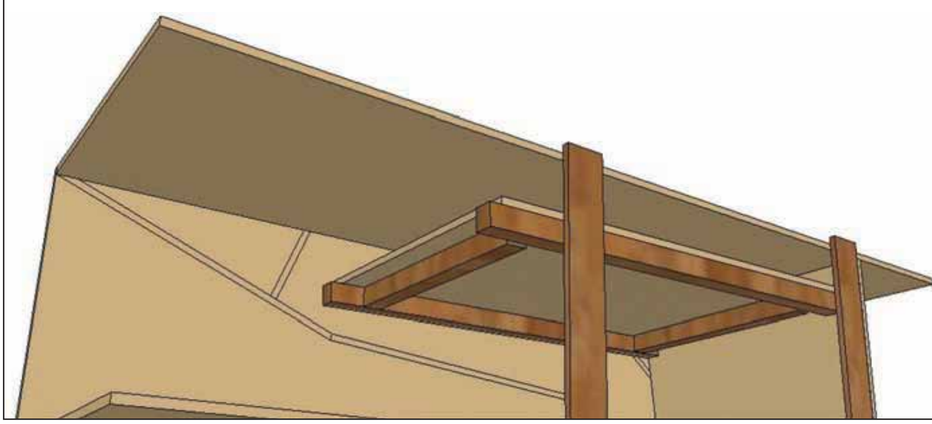


FIGURE 12: Panel 6 straightened out with guideboards.



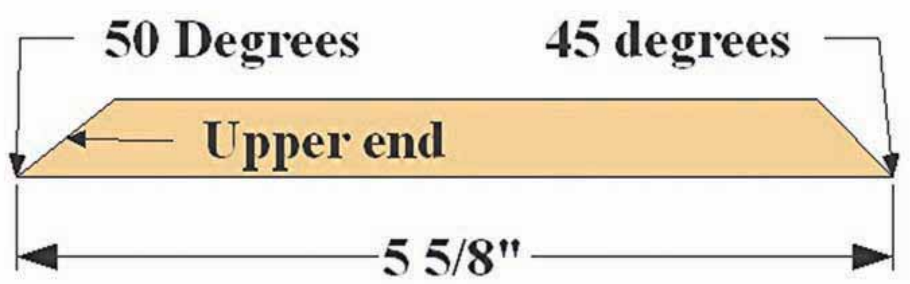


FIGURE 13: Panel 7.

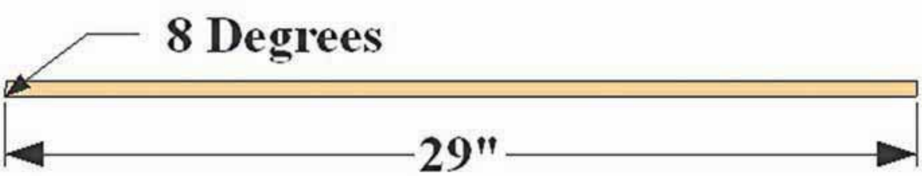


FIGURE 14: Panel 8.

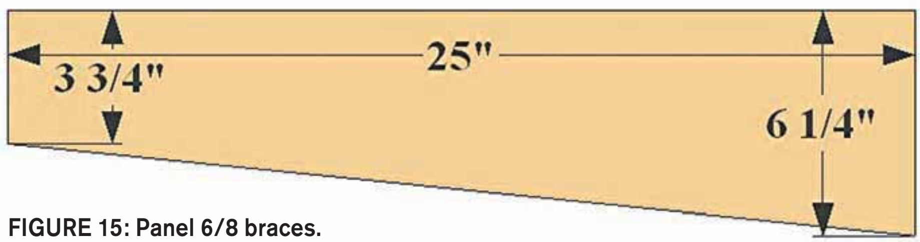


FIGURE 15: Panel 6/8 braces.

The 3/8″ deep relief prevents the driver surround from hitting it in long excursions. The relief length shown is for a loaded cab with 15″ driver; you can shorten it for a 12″ driver. Attach the braces to panel 1. To get the position right, lay a brace on the side, against panel 1, lining it up vertically so that its edge matches the joint line with panel 2. Use a carpenter's square to transfer the vertical positioning to panel 1.

Cut and install panels 4 and 5 (**Figs. 8** and **9**), and panel 6 and panel 3/6 braces (**Figs. 10, 11**). Install these and all remaining braces by applying adhesive to their edges and sliding them into place, pushing in far enough to just be snug without bowing the panels. Secure them in place with nails or screws.

You may dado grooves in the panels for the braces to fit into, especially if you use ¼″ thick braces. If you do so, be sure to size the braces to account for the dado depth, and dado the panels before you attach them to the assembly. **Figure 12** shows how to straighten a badly warped panel 6 with

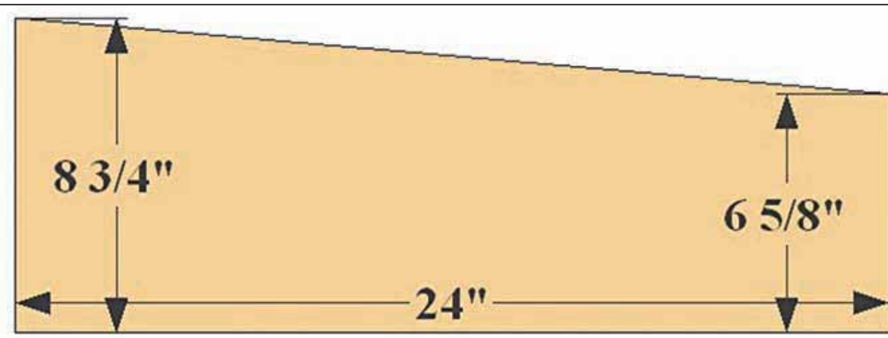


FIGURE 16: Panel 2/8 braces.

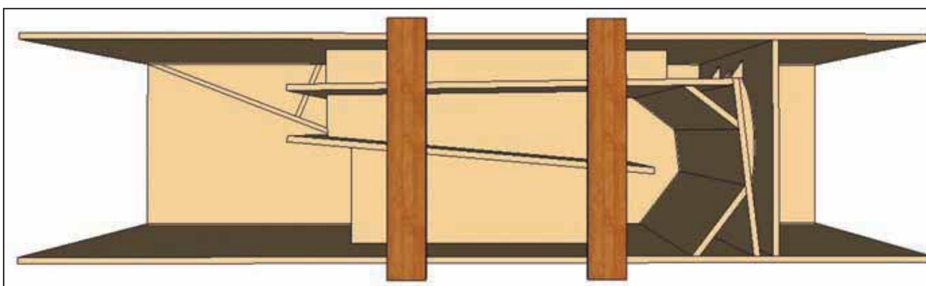


FIGURE 17: Panel 2/8 braces installed.

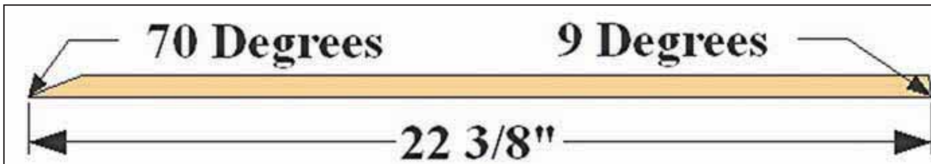


FIGURE 18: Panel 9.

guideboards, screwed and/or clamped to it, for installation. Once in place the braces will hold warped panels true, and then you can remove the guideboards.

Cut and install panel 7 (**Fig. 13**). If your saw can't do a 50° angle, cut it 45° and sand it to fit. Cut and install panel 8, panel 6/8 braces, and panel 2/8 braces (**Figs. 14-17**). Cut and install panel 9, panel 10, and panel 2/9 braces (**Figs. 18-20**). Cutting a 70° angle on panel 9 requires a tablesaw with a special jig; if you don't have one cut it at 45° and sand it to fit. Cut 1″ × ½″ notches into the leading and trailing edges of the panel 2/9 braces, about halfway up. Those notches provide nesting for 4″ wide transverse braces that span from side to side. Attach those once the Panel 2/9 braces are in place (**Fig. 21**).

Clamp or temporarily screw the second side in place, using a long pipe clamp to square the assembly to the side. Reach through the back hole and trace the joint with panel 1 on the inner face of the second side. Trace the joints of panels 8 and 9 from the front of the horn. Remove the second side. If you did an exceptional assembly job, the previously drilled pilot holes will all be in the right places; but if not, just drill new holes in the right spots. The remaining panels aren't accessible for tracing, so measure their exact locations relative to panels 1, 9, and 10 and transfer that to the second side.

Lay the cabinet on its side and apply a generous bead of adhesive on all the panel edges. Put the second side atop the assembly, making sure that it is oriented so that the pilot holes line up with the panels. Fasten it, with at least a few screws to pull the joints tight, using a long pipe clamp or two to pull it into perfect alignment with the rest of the cabinet as you drive the fasteners. Look inside the rear and the horn mouth to be sure there is adhesive squeeze-out on all the joints, caulking them with more adhesive as required.

Cut eight driver chamber wall stiffeners (**Fig. 22**). Install the stiffeners on the driver chamber walls and panel 1, ensuring that they don't interfere with the driver installation. Rim the driver

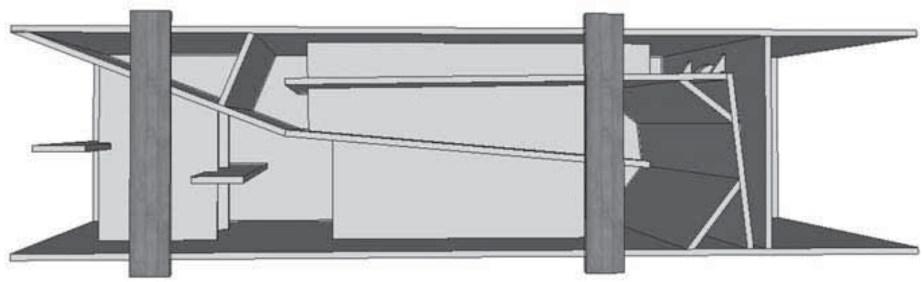


FIGURE 21: All panels and braces installed.

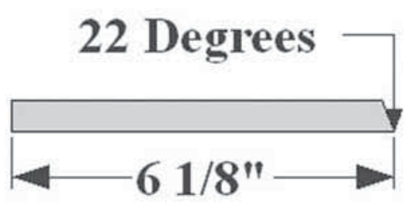


FIGURE 19: Panel 10.

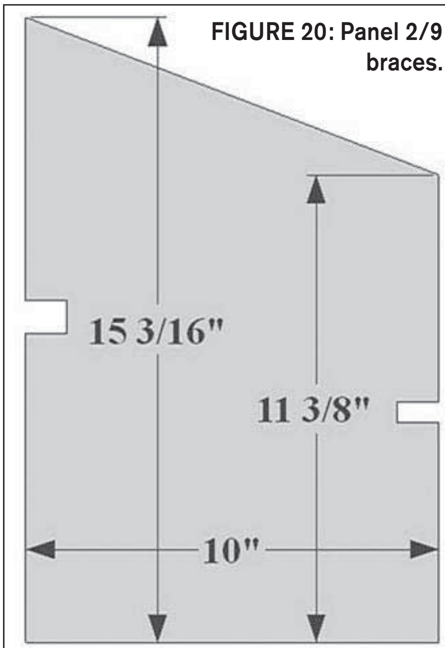


FIGURE 20: Panel 2/9 braces.

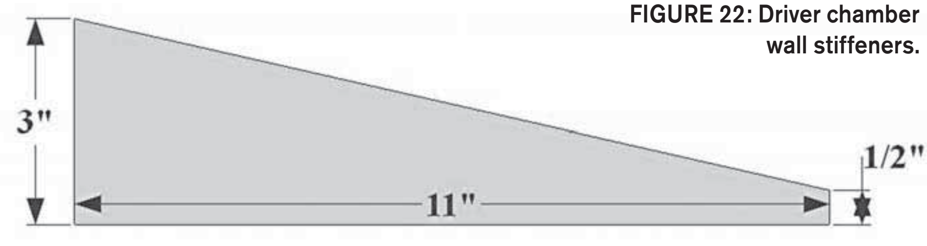


FIGURE 22: Driver chamber wall stiffeners.

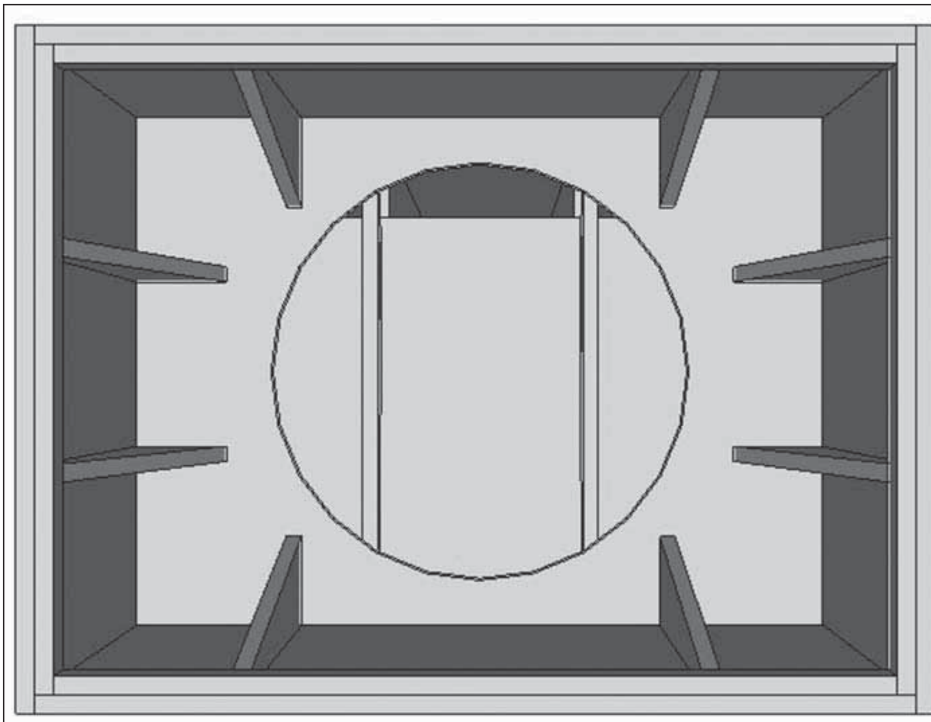


FIGURE 23: Stiffeners and flanges installed.

chamber opening with 1 × ½″ plywood strips to serve as flanges for the cover attachment (**Fig. 23**). Produce a jack/binding post mount by drilling or sawing a hole through one of the driver chamber walls, backing the hole with a piece of ¼″ plywood. Cut out the chamber cover and chamber cover braces (**Figs. 24, 25**). Attach the braces (**Fig. 25**), spacing them wide enough to not hit the driver magnet or frame.

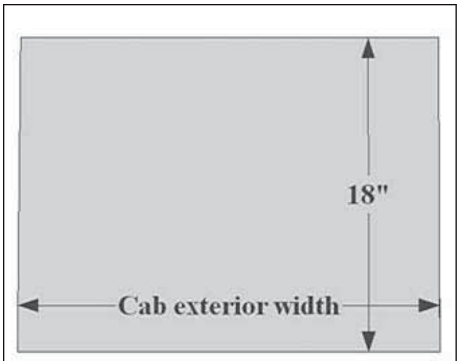


FIGURE 24: Chamber cover.

FINAL TOUCHES

Treat the exterior edges with a sander or router to remove excess/overlapping trim. If you're painting, fill the holes over the nail or screw heads before sanding and finishing the box. I prefer polyester auto-body filler for this job, because it sets fast, holds tight, and doesn't shrink. If you're veneering the box, filling the heads is optional. An easy finish is a sheet or other piece of cloth draped over the box, because the low frequencies will pass through even the tightest of weaves.

Install and wire the jack or binding post. Install and wire the driver; lock washers are a must, or the bolts will vibrate loose. It is absolutely critical that the driver flange has an airtight seal to the baffle. Leaks may occur near the bolt holes, so if there's any doubt at all about the integrity of the seal, caulk over the area. Test for leaks before attaching the access cover.

Buy a 3′ length of flexible 3/8″ to ½″ plastic or latex tubing. Run a 20-25Hz test tone at 10-12V. Use the tubing as a stethoscope, with one end to your ear, and run the other along the joints of panel 1 and around the driver seal. The test tone will be very hard to hear, but the noise of air rushing through leaks will be easy to hear. Fill any leaks with a dab of adhesive.

The chamber is not lined or stuffed with damping material. Install the cover, using screws every 3-4″. Seal the flange/cover joint for an airtight fit. An excellent material for that is the foam rubber weather stripping used between truck camper top and body, which comes in full inch and wider widths. After attaching the access cover, check its joint with the cab for leaks, and those of panels 8 and 9.

Use boundary loading whenever practical. Having subs next to a wall gets you up to 6dB of additional sensitivity below about 80Hz, and putting them in a corner up to 12dB. In most cases you'll have best results with the cabinet mouth against one wall, about 18″ from the adjacent wall. If the cabinet mouth is a quarter-wavelength from a boundary, there will be up to a 24dB deep cancellation at that frequency.

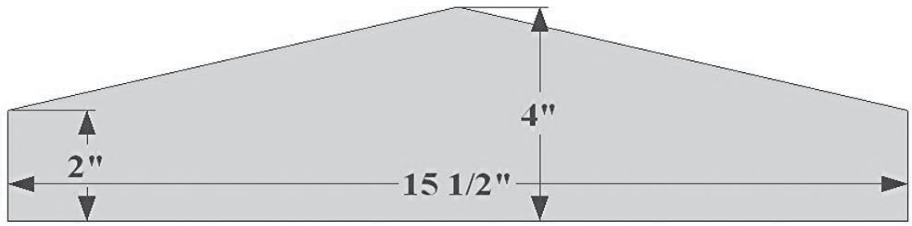


FIGURE 25: Chamber cover braces.

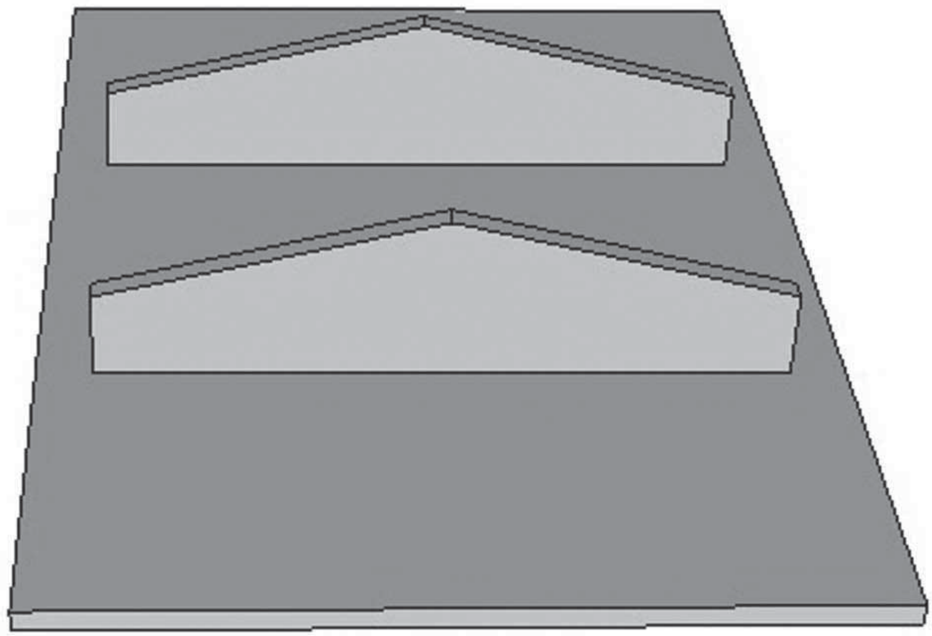


FIGURE 26: Finished chamber cover.

Within the nominal bandwidth of the LPTHT a quarter-wavelength ranges from 19′ at 15Hz to 2.8′ at 100Hz, so middle of room placement usually won't work well. You may put the LPTHT upright aimed at the ceiling, or on 16-24″ legs aimed at the floor. Every room is different, so try a variety of placements to find one that works best. *aX*

A Subwoofer for the Reflection

A bass add-on for the author's sound system.

PHOTO 1: The subwoofer.

The "Reflection" (*audioXpress*, March 2009) is a small, low cost, good-sounding loudspeaker. Because of its small woofer, the addition of a subwoofer is absolutely necessary for the reproduction of the full spectrum. A welcome benefit in this case is that the satellite loudspeakers are released from the large cone movements required for the reproduction of the low frequencies reducing their distortion dramatically.

This article describes the procedure that I followed to add a subwoofer to the Reflection loudspeakers. I used an analog active crossover, which drives the Reflection loudspeaker with its high-pass section and the subwoofer with its low-pass section. Both sections are Linkwitz-Riley filters with a slope of 24dB/octave (4^{th}-order). A built-in Linkwitz transform circuit modifies both the low-frequency response of the satellites and the subwoofer so that a correct total frequency response is achieved.

SUBWOOFER

The subwoofer is based on the INFINITY KCS-120IB 12″ woofer (www.infinitysystems.com). According to the manufacturer, this woofer features IMG injection-molded graphite cone, high-temperature Kapton/Nomex voice coil formers, Kapton laminated copper ribbon voice coil lead wire, and extremely high power reception.

The Thiele/Small parameters of the woofers I measured are given in **Table 1**.

TABLE 1

Parameter	Woofer #1	Woofer #2
Qes	0.712	0.72
Qms	10.14	10.23
Qts	0.66	0.67
Fs (Hz)	31.5	30.7
Rdc (Ohm)	3.85	4.03
Sd (square meters)	0.053	0.053
Vas (ltr)	190	200
Xpeak (mm) according to the manufacturer	6.75	6.75

I used a box with a net internal volume of about 35 ltr. This volume will give rather high Fs and Qts for a subwoofer, but this will be corrected electronically with a Linkwitz transform circuit topology. The external dimensions of the box are 38 × 38 × 35 cm. I constructed the subwoofer box from 18mm plywood and covered it with a wood imitation varnish (**Photo 1**).

THE DESIGN PROCEDURE

Before examining the design of the analog crossover, you need to know the resonance frequency Fo and the quality factor Qo of the satellites and the

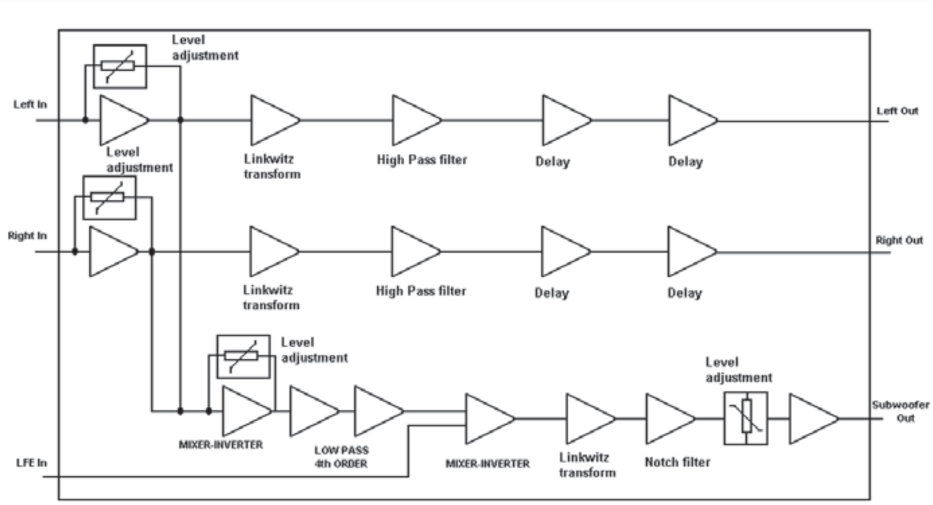


FIGURE 1: The block diagram of the analog active crossover.

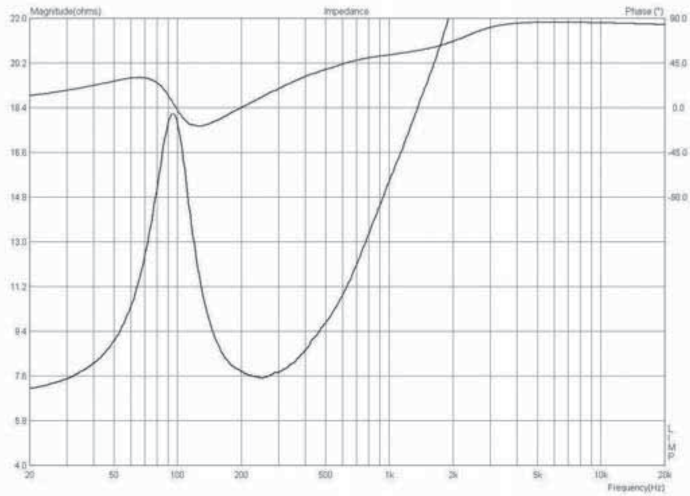


FIGURE 2: The impedance of the woofer section of the Reflection loudspeaker.

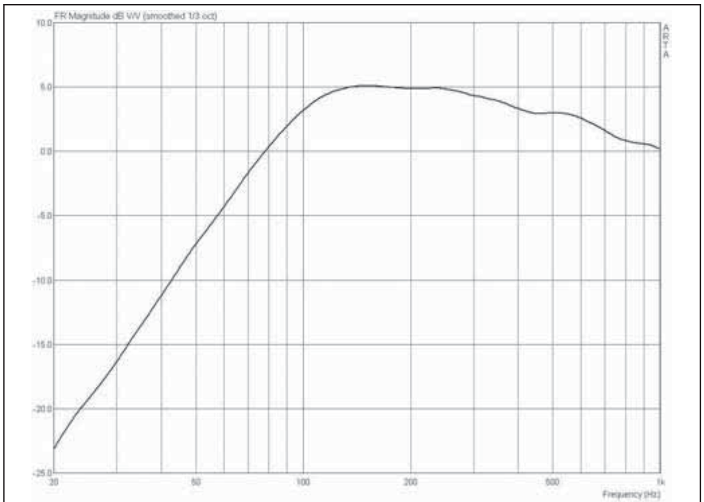


FIGURE 4: Near-field frequency response measurement of the woofer section of the Reflection.

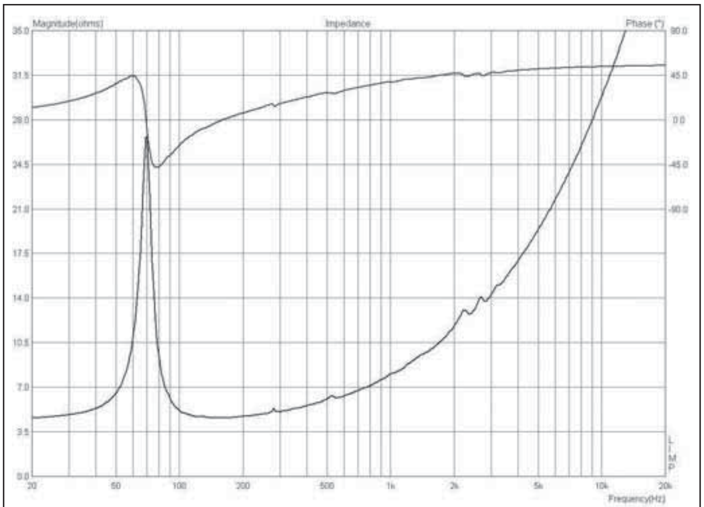


FIGURE 3: The impedance of the subwoofer.

	Impedance measurement		Near field measurement	
	Fo (Hz)	Qo	Fo (Hz)	Qo
Reflection	96	0.71	105	0.8
Subwoofer	69.5	1.3	73	1.6

Table 2: Comparison of the measurement of Fo and Qo using the impedance and the near-field method

subwoofer. I measured these parameters using the demo version of ARTA software (http://www.fesb.hr/~mateljan/arta/), which consists of three separate parts. One of them, the Limp, measures the impedance of a loudspeaker and then extracts automatically the Thiele/Small parameters of this loudspeaker.

The impedance of the Reflection loudspeaker as measured with the LIMP is shown in **Fig. 2**. Only the woofer section of the Reflection crossover was connected. This is the reason for the very high impedance measured after the 2kHz.

From this measurement the program extracted the following results:
Fo = 96Hz and Qts = 0.71.

I measured the impedance of the subwoofer again with the LIMP as shown in **Fig. 3**. The program gave the following results: Fo = 69.5Hz and Qts = 1.3.

In addition to these measurements, I also used another method to compute the Thiele/Small parameters of the loudspeakers. I put a microphone very close to the center of the woofer and the subwoofer and, using the ARTA program, measured the near-field low-frequency response of the Reflection woofer and the subwoofer. The measurements are shown in **Fig. 4** for the Reflection and **Fig. 5** for the subwoofer.

Then I used the Orcad Pspice student version simulation program, entering an ideal second-order high-pass filter using the Laplace function of the program where the transfer function can be entered. Then I modified the Fo and Q of the transfer filter function until I got a frequency response similar to the one I measured in the near-field of the woofer.

Table 2 compares the result of the two methods. As expected, the results are different but the deviations are small and can be explained by the assumptions taken. For example, it was not easy to decide in what level to set the 0dB line in **Fig. 4** and especially in **Fig. 5**.

Interesting results are included in **Figs.**

6 and **7**. In **Fig. 6**, two computer-simulated responses for the Reflection loudspeaker are shown. One is for a loudspeaker with fo = 96Hz and Q = 0.71 (as measured with the ARTA software), and the other for a loudspeaker with fo = 105Hz and Q = 0.8 (as estimated using the near-field measurement). The maximum difference between the two curves is about 1.5dB in the region of 20Hz.

Similarly, in **Fig. 7**, two computer-simulated responses for the subwoofer are shown. One is for a loudspeaker with fo = 73Hz and Q = 1.6, and the other for a loudspeaker with fo = 69.5Hz and Q = 1.3. Again, the maximum difference between the two curves is about 1.5dB in the region of 80Hz. For the design of the analog crossover, I used the parameters as computed with the near-field frequency response.

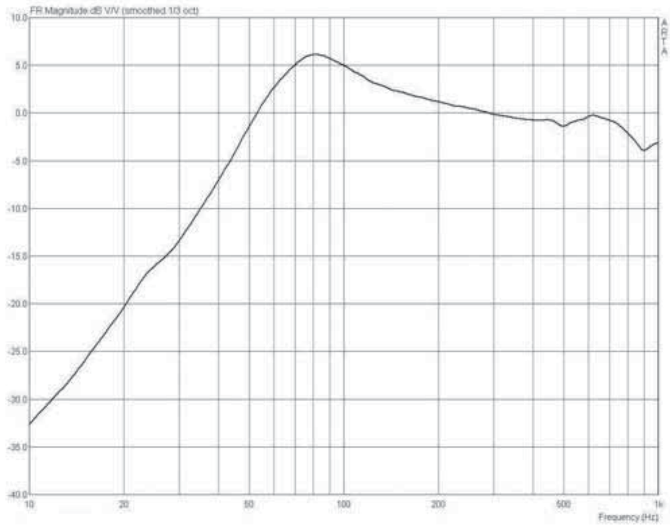


FIGURE 5: Near-field frequency response measurement of the subwoofer.

ANALOG ACTIVE CROSSOVER

The block diagram of the analog active crossover is shown in **Fig. 1**. The signal in the left and right channel follows a similar path, so I will describe only the left channel.

After the input inverting buffer with the level adjustment, the signal is divided into two paths. One drives the high-pass section and the other the low-pass section. The high-pass includes a Linkwitz transform circuit topology and a second-order Linkwitz-Riley filter so that a fourth-order Linkwitz-Riley filter is realized. There are also two delay circuits, in case they are needed. The second delay stage is also used as a buffer to drive the output to the power amplifier.

The low-pass section is common for both channels.

The buffered signal from each channel is driven to a mixer circuit and a mono signal is produced. A trimmer adjusts the level of the signal. After the mixer, the signal is driven to a fourth-order Linkwitz-Riley low-pass filter realized with two second-order stages.

The next stage is a mixer circuit, which is used to add the bass signal from an external low-frequency effect (LFE) channel. A Linkwitz transform circuit topology which follows is used to equalize the bass response of the subwoofer. The next stages are optional and include a notch filter for room equalization and an equalizer, which also drives the output to the power amplifier. Between the two stages a trimmer adjusts the total level of the subwoofer output.

The complete electronic diagram of the analog active crossover is shown in **Fig. 8**. The topology of the right channel is similar to the left channel, so I will describe only the operation of the left channel. No DC blocking capacitors are included at the inputs of the crossover, so you should check that the previous stage is free from any DC offset; otherwise, you should use a DC blocking capacitor with a value greater than 10μF.

The IC1A buffers and inverts the input signal. The components around IC2A form the Linkwitz transform circuit topology, which places a pair of complex zeroes (Fo, Qo) on top of the pole pair of the Reflection loudspeaker to exactly compensate their effect. A new pair of poles (Fp, Qp) is then placed at a different frequency to obtain the desirable frequency response.

The components around IC3A form a second-order high-pass filter that, together with the Linkwitz transform circuit and the loudspeaker, form a fourth-order Linkwitz-Riley filter. The next two circuits around IC4A and IC5A form delay circuits and are optional in case they are needed. Resistor R33 buffers the output from any external capacitive load, and capacitor C21 blocks any output offset. If the next stage already has an input DC blocking capacitor, then you can replace the capacitor C21 with a short circuit.

The operation of the subwoofer channel is as follows: From the outputs of the input buffer stages of each one of the two channels (outputs of IC1A, IC1B), the signal goes to the mixer amplifier IC6A. The gain of the mixer is adjustable with the R66 trimmer. The output of this mixer drives the first stage of the fourth-order low-pass Linkwitz-Riley filter, which is formed with the components around IC7A. The second stage is real-

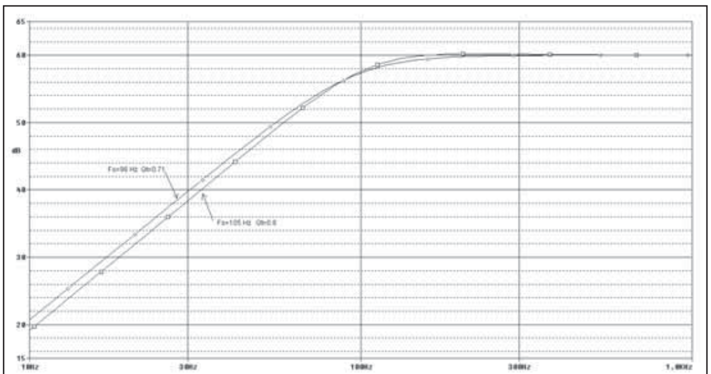


FIGURE 6: Frequency response comparison for the Reflection loudspeaker with two different sets of Thiele/Small parameters.

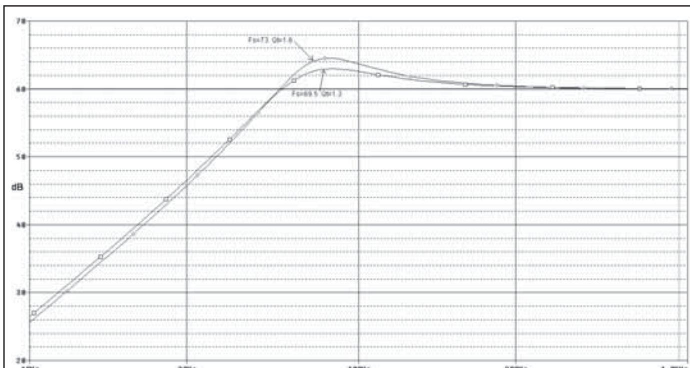


FIGURE 7: Frequency response comparison for the subwoofer with two different sets of Thiele/Small parameters.

ized with the components around IC8A.

The next stage is again a mixer, which adds the mono low-frequency signal of the L and R channels with the external low-frequency effect channel (LFE) of a multichannel system (if this exists). The next stage around IC9B forms the Linkwitz transform circuit for the equalization of the subwoofer. This circuit places a pair of complex zeroes (Fo, Qo) on top of the pole pair of the subwoofer to exactly compensate their effect. A new pair of poles (Fp, Qp) is then placed at a different frequency to obtain the desirable frequency response.

The output of this stage drives an optional notch filter around IC7B, which can be used for the equalization of the room response. The signal is buffered with IC8B. The trimmer R65 adjusts the total volume of the subwoofer. The circuit around IC6B is an optional equalizer for future use. Resistor R61 again buffers the subwoofer output, and the bipolar capacitor C37 blocks any offset from the signal.

THE PCB

The construction of the analog active crossover is too complicated to be realized on a general-purpose PCB, so a PCB was designed using the demo version of the Eagle Layout editor, which you can download for free from Cadsoft (www.cadsoftusa.com). The demo version is fully operational but has a limitation for the maximum dimensions of the PCB, which was not a problem for this design.

The placement of the components on the PCB is shown in **Fig. 9**. You should place the jumpers on the PCB prior to the installation of all other components. I used pin headers for the connection of the input and output cables. I placed the trimmers for the volume adjustment on the PCB but you can use potentiometers of similar value if there is a need for external adjustment. All the op amps were placed on gold-plated sockets.

As in every filter construction, I strongly recommend the measurement of the resistors with an ohmmeter and of the capacitors with a capacitance meter, at least for the components that are used for the filters and the equalizer. A tolerance of about 1% for the capacitors will guarantee that the response of the filter will be very close to the theoretical response, which is very important.

I usually buy a large quantity of the capacitors that I need. Then I measure the value of each capacitor and choose the components with the closest value to their nominal value. This method provides high accuracy filters with a minimum cost. **Photo 2** shows the assembled PCB of the analog active crossover. The IC4 and IC5 were not used for this version, so no IC sockets were placed in their position.

TESTING THE CROSSOVER

After the assembly of the components on the PCB, you should test the analog active crossover to verify its proper operation, using some basic equipment such as a low distortion audio frequency oscillator, an oscilloscope, and an AC voltage meter. I attached an oscillator at a frequency of about 130Hz to the input of each channel of the crossover, then checked every output to ensure they were clean without noise or high-frequency oscillations. I chose 130Hz because with this frequency both the high-pass and the low-pass sections are active.

If you have a distortion analyzer, then you can perform a distortion measurement.

The most important is to verify that the crossover has the correct frequency

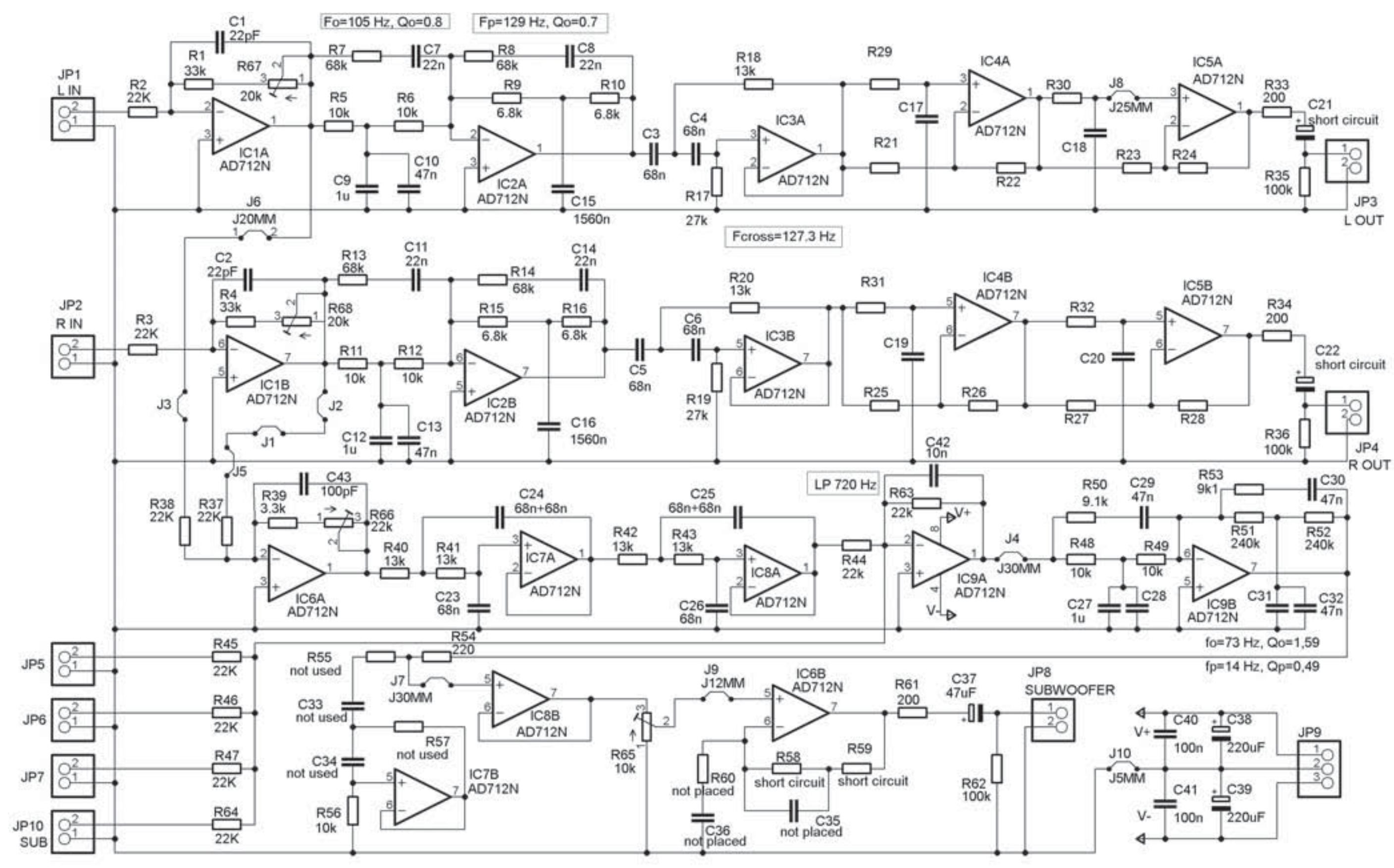


FIGURE 8: The complete electronic diagram of the analog active crossover.

response. With the oscillator connected to the JP1 L IN of the crossover, you should measure the following values at the JP3 L OUT connector. The 0dB reference is the output signal at 1kHz.

1kHz	0dB
240Hz	- 0.73dB
140Hz	- 3.8dB
82Hz	- 11.3dB
58Hz	- 17dB

You should measure similar values for the right channel also, and next measure the response of the subwoofer channel. Measure the following values in dB at the JP8 subwoofer connector when the input signal is connected only at the L IN (and also at the R IN connector). The 0dB reference is the output signal at the JP8 subwoofer connector at the frequency of 20Hz.

240Hz	- 41.5dB
140Hz	- 27.8dB
82Hz	- 24dB
35Hz	- 8.3dB
20Hz	0dB

If your measurements are similar to these, the testing of the analog active crossover is concluded.

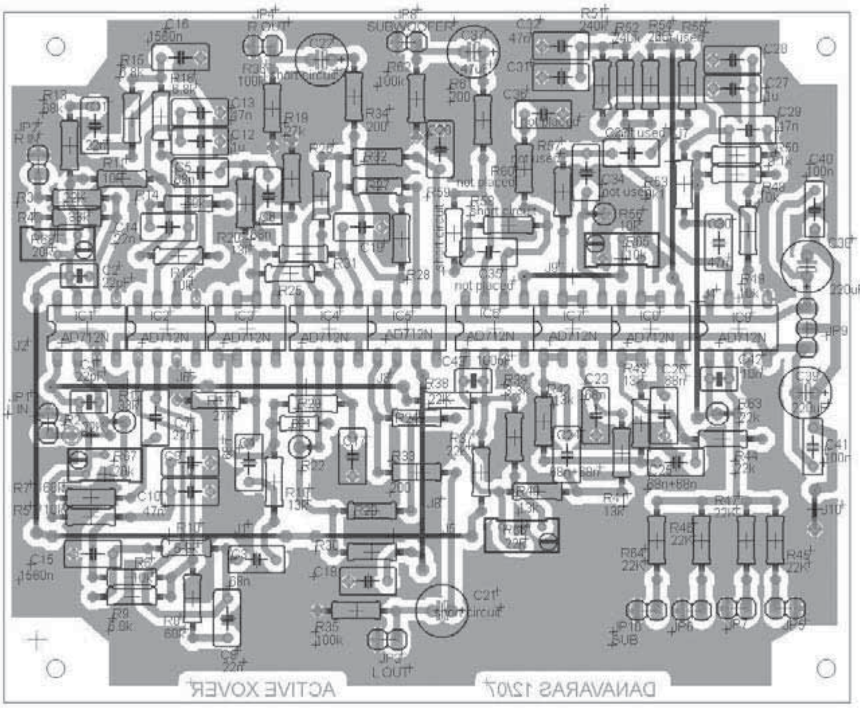


FIGURE 9: The PCB assembly of the analog active crossover.

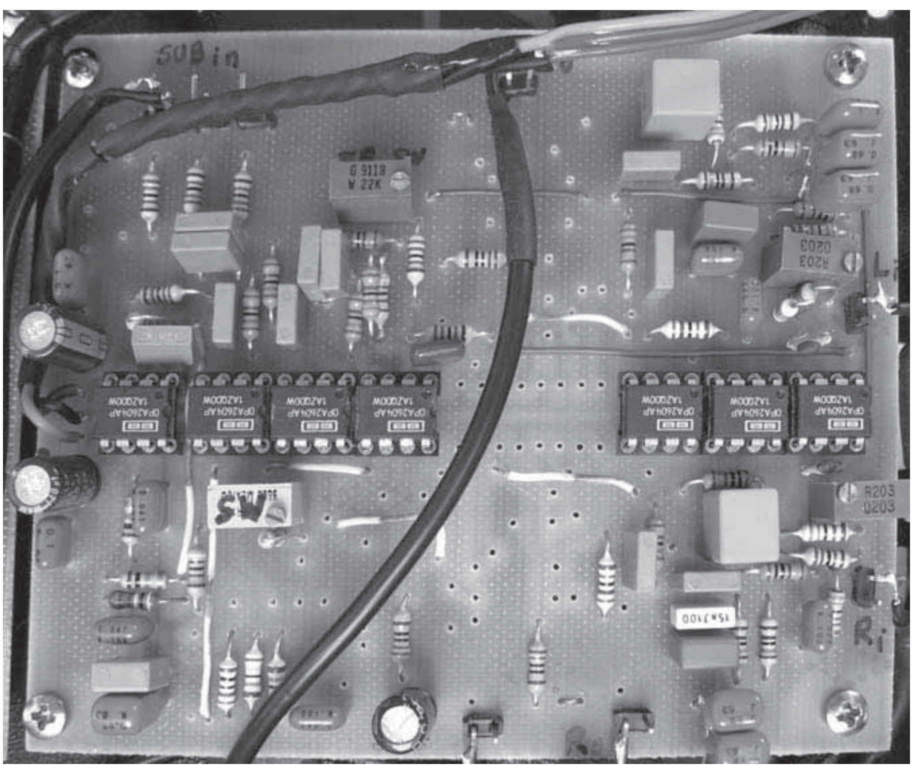


PHOTO 2: The PCB of the analog active crossover.

LEVEL ADJUSTMENTS

After the testing, connect the analog crossover to the system for final level adjustments. For this reason, the crossover includes four trimmers (R65, R66, R67, and R68). For the level adjustment, I used the ARTA software and measured the frequency response of the surround loudspeakers with the crossover and the subwoofer connected. The units were placed in the room in their normal positions.

When the level adjustment is completed, perform a listening test with a very well-known music piece in order to verify the correct adjustment. *aX*

TABLE 3: PARTS LIST OF THE ANALOG CROSSOVER

All resistors 0.4W, 1%
All capacitors 5%, MKT, except as noted

C1, C2	22pF
C3-6, C23, C26	68n
C7, C8, C11, C14	22n
C9, C12, C27	1µ
C10, C13, C29, C30, C32	47n
C15, C16	1560n
C17-20, C33, C34	not used
C21, C22	short circuit
C24, C25	68n + 68n
C28, C31, C35, C36	not placed
C37	47µF
C38, C39	220µF
C40, C41	100n
C42	10n
C43	100pF
IC1-IC9	AD712N
R1, R4	33k
R2, R3, R37, R38, R44-47, R63, R64, R66	22k
R5, R6, R11, R12, R48, R49, R56, R65	10k
R7, R8, R13, R14	68k
R9, R10, R15, R16	6.8k
R17, R19	27k
R18, R20, R40-43	13k
R21-32, R55, R57	not used
R33, R34, R61	200
R35, R36, R62	100k
R39	3.3k
R50	9.1k
R51, R52	240k
R53	9k1
R54	220
R58, R59	short circuit
R60	not placed
R67, R68	20k

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Anechoic correction may have limited resolution at bass frequencies (20Hz - 200Hz), where room measurement and correction can be used.

CORRECTIONS

The following contact information was omitted from the recent review of the Bryston BCD-1 CD player (June *aX*, p. 22):

Bryston Ltd. is a Canadian company with authorized dealers in countries worldwide as well as throughout the US and Canada. To find the dealer nearest you, go to bryston.com and click on "dealers."

6L6 AMP

Like many subscribers, I, too, have built the Joseph Still 20W 6L6 amplifiers (*GA* 5/00, p. 20). The amplifier works and sounds great; I am using it in our (small) home theater system.

I have a question for Mr. Still on the value of a particular capacitor. It's C5, the capacitor used in the feedback circuit. The schematic lists 100pF, but the parts list says 100nF. I suspect it is picofarad.

Note: I am currently running it in open loop mode.

Douglas L. Castle
dougcastle@comcast.net

Joseph Norwood Still responds:

I'm very happy you enjoy the 6L6 amplifier. The 6L6 (later the 6L6GC) was the "Cadillac" power tube of the 30s and 50s. Since I presented the 6L6 amplifier using a single driver stage, many commercial amplifiers have appeared using this same concept. Ed Dell told me that the higher output CD players and FM tuners of present-day design made this possible. I would like to caution anyone who has built this amplifier to never substitute the 6L6 with other more recent power tubes, because they "all" exceed the power output limitations of the power output transformers.

Again, I'm very pleased you are satisfied with the performance of your amplifier. This capacitor is 100pF as you surmised. Thanks for the letter.

SHIELDING

In reference to the "Cable Hum" letter in *audioXpress* (6/10, p. 28), I have a polite question: Why do you print a letter which includes a Figure 1 which does absolutely nothing to explain what the letter writer says?

Specifically, drawing "H" shows better performance at both 400Hz and 50kHz than drawing "F," which is identified as the "preferred circuit," but there is no explanation given for this.

Strange! Or am I missing something?

Doug Pomeroy
audiofixer@verizon.net

Owen Gallagher responds:

The letter was just a cover letter for the old Boeing chart. I expected the chart to get used and just did a quick note.

So the numbers are a little off. In measurements such as this, numbers are fuzzy. Repeating the same setup from scratch the next day can get ±3dB, and at another site it can be ±6dB. EMI engineers are used to this.

It seems to be honestly presented data. Whoever took it is definitely retired by now. The chart is all we have.

The lesson from the chart is that a twisted pair with the load end isolated is the best for shielding effectiveness at low frequencies where the predominant coupling is magnetic. This means that the little RCA connectors are problematic, as they are coaxial with shield to chassis at both ends.

High-frequency shielding is best with shield braid coaxially grounded at both ends to chassis. Aluminum-coated Mylar wrap with aluminized side facing out under the braid is important because it reduces shield leakage at high frequencies.

Shielding effectiveness can be improved over this twisted pair by using Magnetic Shielding Corp. braided 4 wire with mu metal braided magnetic shielding. The 4 wire braid connected correctly has less magnetic field pickup than the twisted pair. It is basically two twisted pairs twisted in opposite directions braided together. The mu metal braid diverts, to some extent, the magnetic field away from the braided 4 wires inside.

Magnetic field shielding is never as effective as electrical field shielding, so reducing pickup is more important than shielding.

I had used this braided 4 wire cable and similarly braided the 600A magnet drive cables in an application. I did not get quite enough attenuation. I used a different transducer with higher output, so it achieved adequate signal-to-signal plus noise ratio.

When faced with the problem again recently, I recommended using analog fiber optic link for the signal cable that was in the same tray with a 250kW magnet drive cable untwisted pair. Big stuff.

GOOD DESIGN

In Bard Kallestad's otherwise excellent article "Designing for Everyone" (July '10, p. 14), there is an error in calculating the output voltage under a 5mA load, starting with his Fig. 1. For easy analysis, it is best to replace the two 47K resistors with their Thevenin equivalent as shown in Fig. 2.

Now, when 5mA is drawn from the emitter of the 2N3904, Mr. Kallestad correctly calculated the base current at 50μA, but this current is drawn through 23.5K, not 47K. Figure 3 shows the correct output voltage under a 5mA load.

In this analysis I have assumed 0.7V for Vbe. (0.6V or 0.7V is used by different designers; at 5mA I think 0.7V is more accurate.) The actual output voltage is closer to 2.6V than Mr. Kallestad's 1.55V. This is still very poor regulation, just not quite as poor.

I thought the balance of the article was excellent and clearly stated many valid and valuable goals for robust and repeatable designs.

Thomas Bohley
Colorado Springs, Colo.

Bard Kallestad responds:

Thank you for the kind remarks regarding my article. I'll take them as high praise from a long-time HP design engineer. You are correct regarding the Thevenin equivalent base resistance. As you say, it does make a small difference in the numbers, though it does not upset the overall point of the piece. I thank you for pointing it out, and for enjoying my article.

SIMPLE BTO

I read the article about the construction of a Bridged Tee Oscillator (BTO) by Dick Crawford in July issue of *audioXpress* (p. 5) with interest. I have long wanted to design a simple audio source for speaker and amplifier testing, and this simple BTO seems to fit that bill quite well.

The article, however, seems to be missing a complete schematic showing all the connections. The author mentions a compensation capacitor C3 = 0.018µF. According to the "wiring schematic" it goes from node 23 to node 1, but node 1 is found nowhere else.

Also, node 42 goes nowhere, but it should obviously connect to node 38. The text also mentions two high-frequency compensation capacitors of 220pF nowhere to be found on the wiring schematics.

The author does cover himself in that he will provide more construction details if the interest arises. I understand how the oscillator works and appreciate the design, but I believe some information is missing.

Hans J Weedon
Salem, Mass.

Dick Crawford responds:

Thank you for your letter. I don't like making mistakes, but it is gratifying when someone reads an article closely enough to catch them. In answer to your queries:

1. Node 1 is a contact on switch S2b. My schematic capture software doesn't do switches. Sorry.
2. Node 42 does indeed go somewhere. It is the wiper on potentiometer P2k. The obscure clue here is "P2k," which is a 2kΩ pot. My software doesn't do pots. Sorry.
3. The 220pF capacitors are indeed missing from the schematics. My error.

One of the 220pF capacitors goes in parallel with R43 in Fig. 3, the other 220pF capacitor goes in parallel with R47 in Fig. 3.

PVC SPEAKERS

I found the Fire Sticks speakers to be a nice construction article (July, p. 10). It took me back to the 70s when I built a set of sewer pipe eights. They used a set of 10″ fired clay sewer pipes with an 8″ driver mounted on the tops with a funnel as a dispersion cone. I decided I needed to build a pair.

I have a couple of questions and a comment:

1. The plans show a 1″ port below the woofers, but none of the actual pictures show this. Do they need a port or not?
2. The part number on the woofers is transposed. It should be 296-220 not "269-220" (this is a really hard error to catch)—just pointing it out to save someone some frustration.
3. I am not sure about the "bargain" nature of the drivers. While the 4 × 1″ drivers are $1.37 each, the 3″ woofers are $170 each. This brings the total cost for drivers and crossover components to over $400. Sorry, but I am "cheap" (although I prefer "thrifty").
4. Parts Express lists the 3″ woofer as having a 16Ω impedance, and the 1 × 4″ drivers as 8Ω each. The wiring diagram shows the small driver series-paralleled so that they would be an 8Ω impedance. Is this correct?

For this kind of money, I think they deserve a wood cabinet. I did some quick calculations and made a trip to the local "super" lumber store. If you buy hardwood shrink wrapped in plastic (most expensive way to buy, but is ready to cut to length, glue, sand, and finish), you can build the cabinets out of wood for about $50-70 each depending on the wood you use. I am not sure the plastic pipe and fittings will cost much less than this.

I still love the article and hope my comments do not offend. They are pretty cool and the pair I built of wood sound very good.

Bruce Brown
Tuninfork@aol.com

Ken Bird responds:

Thank you for your input on the Firesticks speakers.

The published drawing did show a port and I should have removed it from the original I sent with the article. I found that porting the A3 in the pipe will not produce a smooth bass response. Those speakers are designed for a sealed enclosure and work much better in the sealed pipe.

I chose the PVC pipe to produce a unique enclosure that would appeal to the younger generation. I did my original testing in a wood "pipe" constructed from 1 × 6 pine and I agree that a hardwood version properly finished would work just as well and the speakers would be easier to mount on a flat surface. I was a bit surprised to see them as described as "bargain" speakers on the cover as I never referred to them as such in my text.

The "sewer pipe eights" you referred to were a David B. Weems project published in *Popular Electronics* in June of 1962 (see www.audioXpress.com/magsdirx/ax/addenda/index.htm). Dave designed a number of ceramic pipe speakers over his career. See my article on Dave in the October 2009 issue of *audioXpress*.

As to the Audience drivers, I got mine directly from the manufacturer but used Parts Express as the source because they carry the Audience line. Audience changed the design of their driver to a dual voice coil version making it easier to configure line arrays, which is what they were originally designed for. It was my error to assume that Parts Express would have the new version in stock. I did check with them on the availability of the 1″ × 4″ drivers but did not mention the 8Ω version of the A3. I apologize for any confusion it caused you and other readers.

I spoke with Mark Liptak, the buyer at Parts Express on July 8 and he said they will soon have the Audience A3 8Ω dual voice coil in stock. In the meantime the 16Ω version can be used, and except for the impedance, it has the same specifications as the DVC version. You will need to change the crossover coil specified to a 2.54µH coil.

CHALLENGE

I would like to see an up-to-date Red Book tube DAC project. I am still a believer that you can get a good emotional, dynamic, three-dimensional sound stage out of digital. Unfortunately, designing digital is not my forte. I know it is a tall order. Are any writers up for the challenge?

Don Smith
donsmi@cox.net

ENDURING HOBBY

I would like to make a comment about the article, "Amplifier Comparison Using Oscilloscope Waveform Plots" by Kent Smith (May '10, p. 20).

Figure 1 (Tube Amp) and Figure 2 (SS Amp) compare amplifier output to input. Channel 3 (shown purple in the figures) was connected to the amp OUTPUT, and Channel 4 (shown green in the figures) was connected to the INPUT.

There do not appear to be noticeable temporal anomalies in either figure. The "slurs" on the downsweeps seem to line up quite well for both amps. What are apparent are amplitude errors. These errors are likely related to the low-frequency behavior of the amps. Tube amps with output transformers, in particular, cannot go very low in frequency because of factors relating to the core (leakage inductance, and so on).

My tube "re-education" occurred 15 years ago when a colleague gave me a Dynaco ST-70 in need of some TLC. I rebuilt it over the course of a few months and I was really surprised at the result. The music sounded "alive" instead of "dry" or "clinical" (my other amp was a Kenwood that I had thought was quite good). My enjoyment of music was greatly enhanced.

The first time you turn on a tube amp there is always the fear that something is going to blow up (that did happen with my Audio Research D78 when the regulator tube exploded). Was it adrenaline or the fact that I started listening intently to the music? Anyway, my "free" amplifier ended up costing me thousands in upgrades as I went in pursuit of audio nirvana.

What makes our hobby so enduring is that it is a journey with an unreachable destination.

Andre Routh
Medford, N.J.

Kent Smith responds:

Your explanation about the output transformer would sound good had the plots been labeled correctly. However, since the captions were accidently switched somewhere, we face the odd problem that the "chip" amp actually has worse accuracy than the tube amp with its transformer!

I do believe that it really is more of an amplitude problem, but it produces a frequency error problem also of several percent or so. This is double the time error of the tube amp. In addition, the negative overshoot is more pronounced in the chip amp, which shouldn't have any weird reactance issues. Its response is flat to 20Hz.

On top of all this, the chip amp has a THD level at least one order of magnitude lower than the tube amp, yet its plot looks worse! So, I'm just baffled. You do seem to indicate that you don't think the errors are significant, and that may be true. This is a subjective issue. However, these plots do indicate that there is an issue that needs explaining, to me, anyway.

My real point here is that there are differences that can be measured and yet seem to go uninvestigated and unexplained. Is no one else curious? Are we really that happy with the status quo in this field? Perhaps everyone just enjoys seeing the Emperor walk around with no clothes. I guess I'm just not enjoying the joke.

All I'm asking is for people to try this experiment for yourself. Make the measurements, then listen for yourself. Listen to a real drum, and listen to a good recording. This is not a subtle thing, it's easy to hear. And if I'm wrong, I'm only asking you to show me.

By the way, thanks for your kind letters. When I wrote last year, I got really nasty letters from people who were incensed that I dare challenge the status quo. Pehaps offering up real data helped solve that problem. I have learned my lesson! If I can get a setup working to look at spectrum differences, I will try to do an article on that, but I hope someone will do a better article than I can before I get to that point. Anybody?

TUBES OR TRANSISTORS?

With 22 years of experience in researching and manufacturing world-class solid-state guitar amplifiers, I maintain the approach taken to distinguish solid-state and tube amplifiers in the article, "Amplifier Comparison Using Oscilloscope Waveform Plots," by Kent Smith (*aX*, May '10), is not definitive. It is not definitive because the investigation was in the wrong realms.

First, Russell Hamm, in "Tubes Versus Transistors—Is There An Audible Difference?" (*JAES*, May 1973 and reprinted in *Glass Audio* 4/92), finds a difference in the amplitude versus overdrive of low order harmonics. He examined a variety of microphone preamplifiers. Their harmonic structures are influenced by circuit topology and feedback. Single-ended structures have more even harmonics than push-pull structures. Higher feedback has faster rising harmonics. Some harmonics even rise and fall before rising again.

I developed a solid-state emulator and used it to build a typical two-stage microphone preamplifier to compare against Hamm's triode preamplifier. The results were published in *dB Magazine*, as "The Tube Sound and Tube Emulator," July/August 1994 (www.pritchardamps.com). After adjusting for input and output scales, the scope traces were virtually identical from clean to distortion. And after adjusting the bias on the tube emulators, they produced virtually identical amplitude versus overdrive of the same harmonics. For this I used an old Tektronix 5403 with a mating 5L4N swept frequency spectrum analyzer plug-in.

Second, I used these emulators in an early guitar amplifier prototype. It did not function as a tube guitar amplifier because the tube output stage needed more investigation. As my patents mention, the effects revolve around the modulating nature of the push-pull tube output stage. The non-ideal nature of tube amplifier power supplies provides modulating signals, which embellish the signal and make it resilient and fat.

Note, however, that most transistors do not have second inputs akin to the screen grid, and most transistors have output resistances far higher than tube output resistances. Thus, the comparison between tube and transistor output stages depends upon the assumption that these differences have no effect.

Engineering believes the perfect amplifier is one that replicates its input without any embellishments because engineering incorrectly assumed that the human hearing process does not produce harmonics. Guitar players, however, pick their amplifiers by their embellishments. They find the typical solid-state amplifiers like white wine or glass and lacking in dimension. They prefer an amplifier that is akin to red wine or flesh and being multi-dimensional and full-bodied. I suspect that tube audiophile amplifiers are a bit more subtle because some jazz players do not want as much embellishment as blues and rock players.

Eric Pritchard
Pritchard Amps
eric@pritchardamps.com

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Contact Ed at njvet01@yahoo.com.

"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

CONTRIBUTORS

Claude Goeuriot ("The Mustang Speaker," p. 6) resides in France. This is his first article for *audioXpress*.

Rich Johnson ("Low-Frequency Horn Speaker," p. 18) became excited about high-performance horn speakers in his teens when he started DJing. He once begged a guy at the wood shop to cut dimensioned material for his first speaker cabinet. After finishing school, he bought himself all the wood shop tools he could use to build speakers. He earned his BS EE at the New Jersey Institute of Technology. He has experience with wireless handset development and cellular technologies. He currently develops graphical user interfaces (GUI) using Labview for the test and measurement industry.

Bill Fitzmaurice ("The Low-Profile Tuba," p. 28) has been a professional musician since 1966 and has been constructing instruments, amplifiers, and speakers for just as long.

George Danavaras ("A Subwoofer for the Reflection," p. 36) 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.

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Oppo BDP-83 Measurements

By Chuck Hansen

I performed audio measurements on the BDP-83 using the following test discs:

- Sheffield/A2TB "My Disc," CBS Labs CD-1 and Pierre Verany test CDs (the latter two available from Old Colony Sound Lab, www.audioXpress.com)
- CD test tracks that I ripped to 128Kbit sampled MP3 and burned to CD-R
- Philips Super Audio CD DAC Test Disc
- Chesky Super Audio Collection & Professional Test Disc, Part II (despite the name, this is a DVD-A disc)
- Analog test tracks that I recorded in 24/96 AIFF format and had burned to DVD+R (on a Mac using Digidesign ProTools|HD with a TDM system).
- Gary Galo's detailed De-emphasis Test CD (from db Systems http://www.dbsystemsaudio.com/)

I put the BDP-83 in CD playback with a CD test disc for one hour before making any measurements. The player was cool to the touch over its entire surface after this period. One interesting feature is that when a disc is loaded, it begins playback at the track at which it was last stopped rather than at the beginning.

The output impedance, for both two-channel and the 7.1/5.1-channel outputs, measured 210Ω at 20Hz, and 190Ω at 1kHz and 20kHz. The DC resistance exceeded 15MΩ, indicating that the audio outputs (via TI NE5532A dual op amps) were capacitively coupled. Strangely, the composite and component video outputs measured over 10MΩ rather than the 75Ω I usually measure for video outputs. Perhaps Oppo has capacitively coupled their video outputs rather than using transformer coupling.

All the analog outputs had normal polarity, a positive-going test pulse producing a positive-going output. The digital black test track measured < -120dB, indicating that circuitry was probably shunting the outputs to ground during this test. Front left and right channel separation measured –110dB from 100Hz to 10kHz, with both unweighted and A-weighting filters. The center and surround channels showed about 10dB higher residual noise levels than the front channels.

In CD mode, the player performed perfectly in the track defect dropout tests out to track 36 on the Pierre Verany test disc, which contains a 2.5mm gap (Red Book requirement is 0.2mm). At the 3mm defect (track 37) there were audible clicks until the unit muted after 45 seconds.

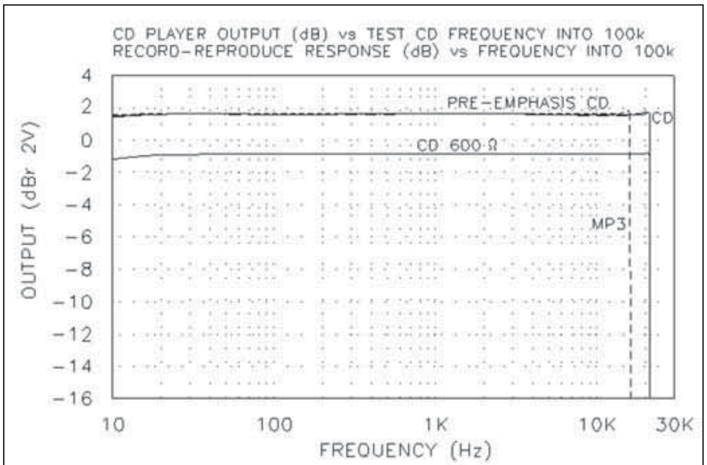

FIGURE 1: Frequency response: CD, MP3.

The 1kHz 0dBfs CD output at the front channels was 2.41V RMS at 1kHz, or 1.62dB higher than the CD Red Book standard of 2V RMS. Balance between the two stereo channels was within 0.03dB. Frequency response for CD and MP3 is shown in **Fig. 1**. There is a slight peak just before the response drops at the limit imposed by the digital filters. The MP3 frequency response with identical CD test tracks ripped to 128k MP3 format was identical to the CD response out to 6kHz, then it dropped off just above 15kHz.

With stereo test tracks, all eight of the multichannel outputs, including subwoofer, had the same frequency response as the stereo outputs. I did not attempt to delve into the menus to revise the response of any output.

Reducing the output load to 600Ω caused the output voltage to drop to 1.817V RMS, or -2.5dB at 1kHz, with a bit more droop (-2.6dB) at the lowest frequency due to the capacitive coupling at the analog output stages. The THD+N at 997Hz increased from 0.0056% into 100k to 0.0069% with the 600Ω load.

The higher definition 24/96 DVD-Audio and SACD tests show the extended frequency response of which these formats are capable. Setup for the DVD-Audio measurements always requires an annoying trip to the TV to map the on-screen menus for the test tracks I need[1]. With either a 24-bit/48kHz

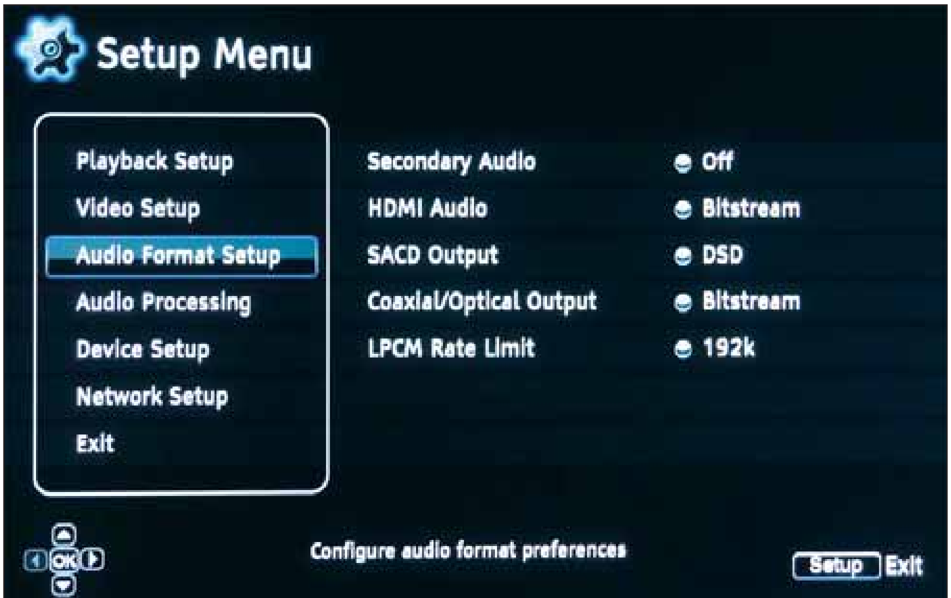

PHOTO A: BDP-83 audio format setup menu.

(24/48) or 24/96 1kHz 0dBfs DVD-A sine wave (**Fig. 2**) the output was 2.42V RMS, or +1.66dB compared with the 2V RMS requirement with the same response peaking seen with CD and MP3.

With two-channel SACD playback, the 1kHz 0dBfs test track produced the same 2.42V RMS, or +1.66dB above the Scarlet Book standard (**Fig. 2**). The SACD test tracks usually do not require a video monitor. However, except for this Oppo unit, I needed the audio format setup menu to select between SACD PCM or DSD data to the DACs (**Photo A**). There is also the same response peaking seen with the other digital formats with the BDP-83. I assume this is due to the digital filter characteristics chosen by Oppo.

THD+N versus frequency is shown in **Fig. 3** for each audio mode. I engaged the 22kHz LP filter in my distortion test set. Note that the SACD test tracks are recorded at –3dBfs, and the DVD-A test tracks are recorded at –6dBfs. This presents a problem in displaying the data because THD increases as the output level drops. I normalized all the distortion curves to the 0dBfs output level of the CD/MP3 tracks by adjusting them to the THD I measured for the 1kHz 0dBfs tracks recorded on the SACD and DVD-A test discs. Note the sound quality penalty paid for the extra storage space that MP3 files provide for portable audio devices.

THD+N versus output voltage is shown in **Fig. 4**, at 1kHz for each audio mode. Despite the fact that the 0dB output voltage is higher than the 2V RMS specification, the distortion continues to drop out to the 2.41V RMS maximum.

The spectrum of a CD 50Hz sine wave at 0dBfs is shown in **Fig. 5**, from DC to 650Hz. The calculated THD based solely on harmonics is 0.0063%, which is the same value of THD+N measured with the distortion test set. The resulting spectrum is near the display noise floor of my 16-bit analyzer. There are no 60Hz power line harmonics or other spuria evident in the spectrum. Engaging the BDP-83 Pure Audio mode (display turns off) did not change the distortion readings.

The MP3 spectrum in **Fig. 6** tells a different story. The 50Hz fundamental sits on a broad band of noise out to 400Hz or so, with the THD+N measuring 0.108%, although it drops to 0.034%

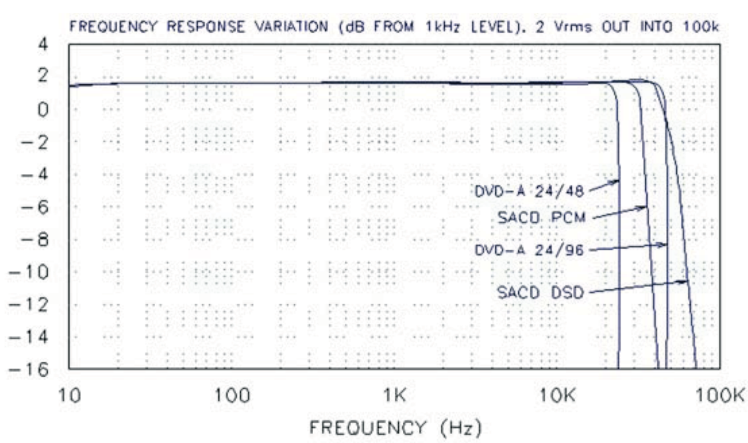

FIGURE 2: Frequency response: DVD-A, SACD.

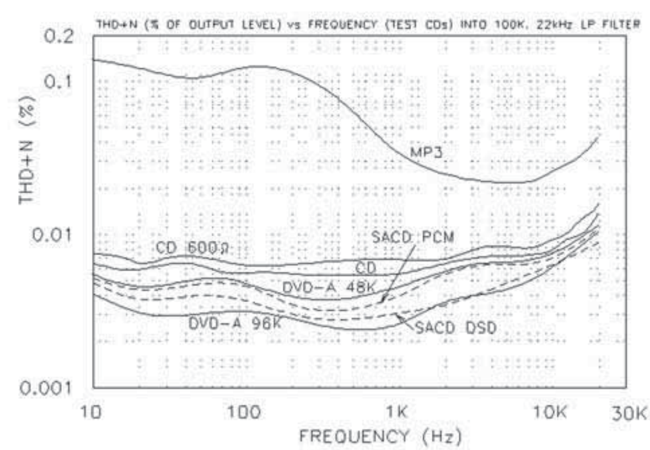

FIGURE 3: THD+N vs. frequency.

at 1kHz. The distortion calculated from the harmonics of 50Hz is 0.037%.

The DVD-A test disc does not have 50Hz test tracks, so I used the 100Hz test signals at -6dBfs. The spectrum analysis to 1.3kHz with 24/48 DVD-A data (**Fig. 7**) yielded a THD+N of 0.0122%. The FFT-calculated distortion was 0.0047%. Increasing the DVD sample rate to 24/96 produced a bit cleaner spectrum, but the true noise floor is still masked by the 100dB display range limit of my analyzer.

The SACD spectrum of 50Hz with the BDP-83 set to SACD PCM is essentially the same as that with DVD-A PCM test tracks, with the THD+N measuring 0.004%. Setting the unit to SACD DSD produces the 50Hz spectrum shown in **Fig. 8**. Note that in both cases the 50Hz test track is recorded at -3dB. The measured THD+N for DSD data is 0.011%, while the FFT-calculated THD is 0.0031%.

The spectrum of a 1kHz 0dB DVD-A 24/48 sine wave shows a clean noise floor with no visible harmonics (**Fig. 9**). THD+N and THD calculated from the 1kHz harmonics were identical at 0.004%. Increasing the sample rate to 24/96 (not shown) shows a spectrum that essentially looked the same. This was also the case with SACD in PCM output format (also not shown).

Switching to SACD DSD mode, there is about 5dB higher noise floor in the spectrum of 1008Hz 0dBfs (**Fig. 10**) and THD+N measures 0.0063%. The FFT-calculated THD is 0.0053%. You can just see the noise floor start to increase above 20kHz, where the DSD noise shaping begins to take effect.

The residual distortion signal in CD mode for 997Hz at 0dBfs (**Fig. 11**) consisted primarily of low-level noise. The THD+N at 0dBfs measured 0.0056%. The MP3 residual distortion signal (**Fig. 12**) shows significant ringing that seems to occur just after the positive and negative peaks of the sine wave. The MP3 THD+N at 0dBfs measured 0.034%.

The residual distortion signals for the two DVD-A modes with a 1kHz at 0dBfs signal show decreasing levels of noise as the sample rates increase. DVD-A mode with 24/48 data is shown in **Fig. 13**, with 24/96 data in

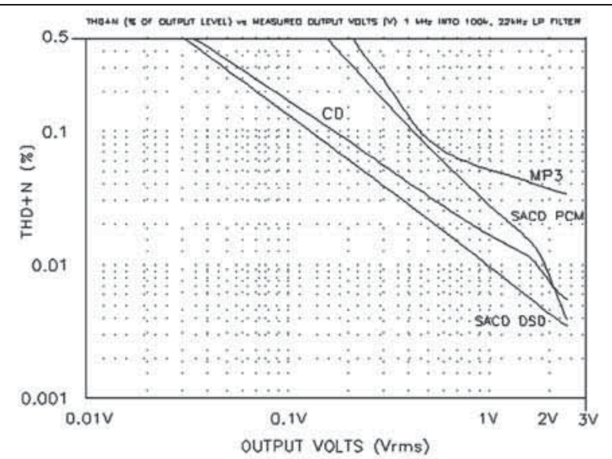

FIGURE 4: THD+N vs. output.

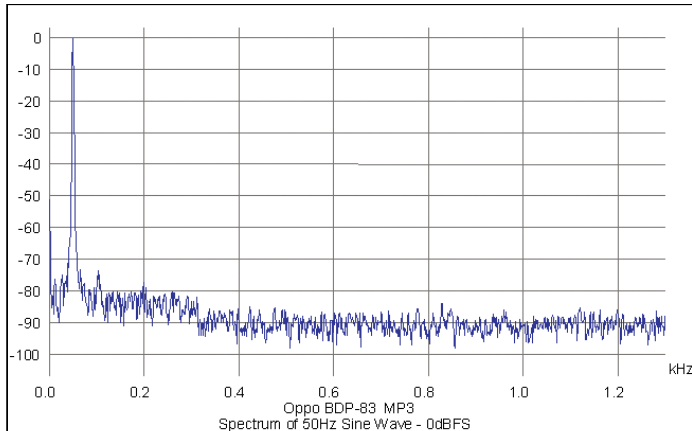

FIGURE 6: MP3—50Hz spectrum.

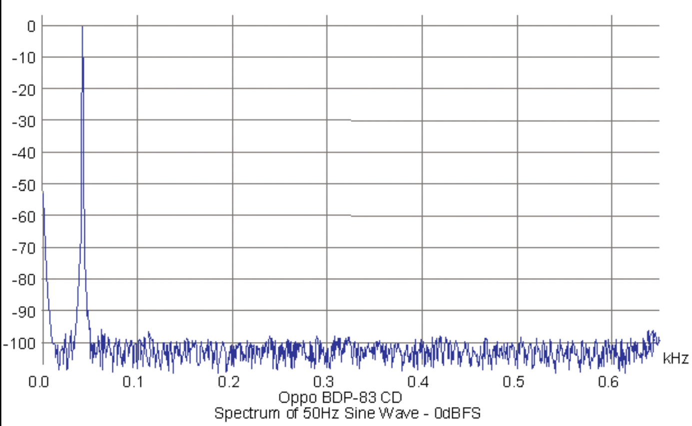

FIGURE 5: CD—50Hz spectrum.

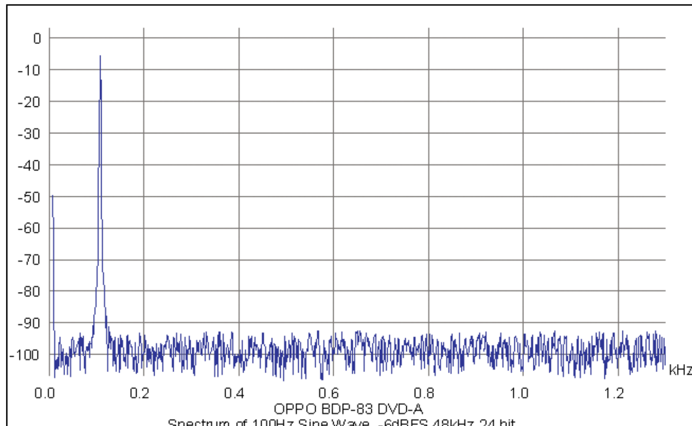

FIGURE 7: DVD48—100Hz spectrum.

Fig. 14. The residual noise level is noticeably higher with SACD PCM (**Fig. 15**).

The distortion residual for the SACD playback of a 1008Hz sine wave in DSD mode (**Fig. 16**) shows what appears to be a lower noise level than the high resolution PCM formats. This is because the residual signal is not to scale, and I needed to switch ranges to avoid overloading the analyzer front end with the DSD high-frequency noise shaping energy. The residual noise, when viewed on a wideband analog oscilloscope, also shows a much higher frequency content than is shown here in the DSO capture.

I ripped the CD 11kHz+12kHz intermodulation distortion (IMD) test signals to 128kHz MP3 format, and the results are shown in **Fig. 17.** The 1kHz and 2kHz products are –75dB, and the 10kHz and 13kHz products are –78dB. The skirts around the 11kHz and 12kHz stimulus signals are broad, indicating more overall noise. There are also spikes of about –80dB at 20kHz and 21kHz.

A test in CD mode with the more difficult 19kHz + 20kHz IMD test track

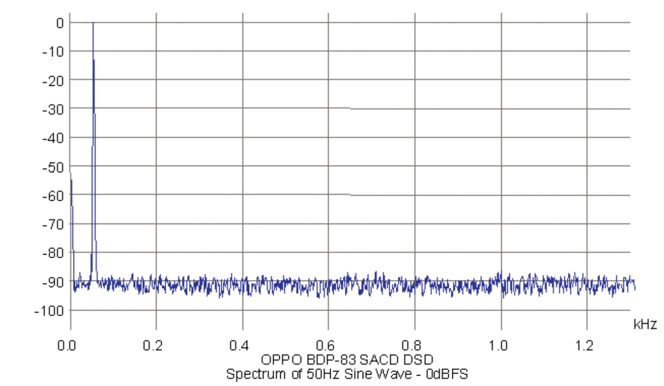

FIGURE 8: SACD DSD—50Hz spectrum.

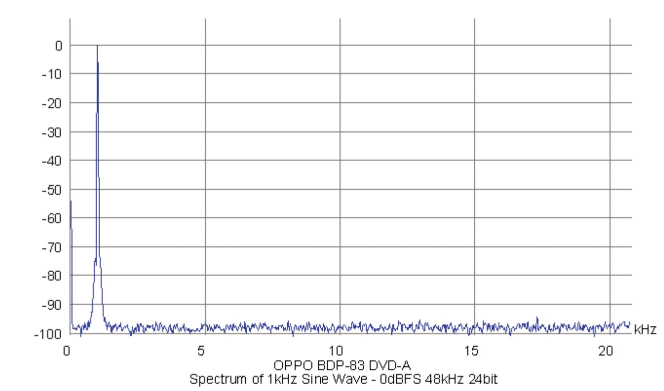

FIGURE 9: DVD48—1kHz spectrum.

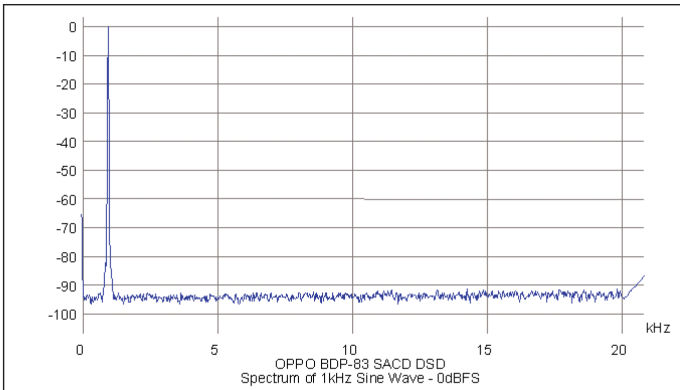

FIGURE 10: SACD DSD—1kHz spectrum.

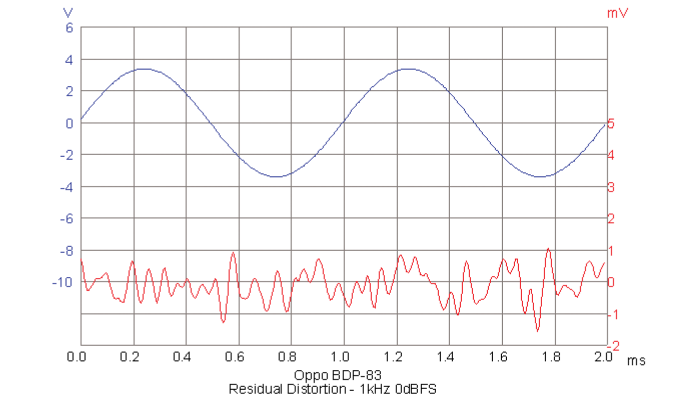

FIGURE 11: CD—1kHz residual distortion.

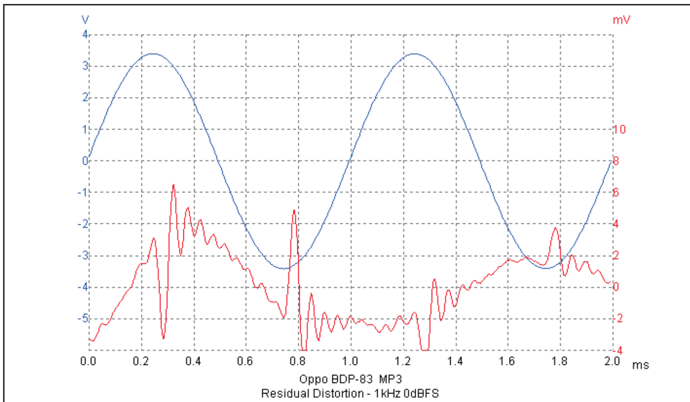

FIGURE 12: MP3—1kHz residual distortion.

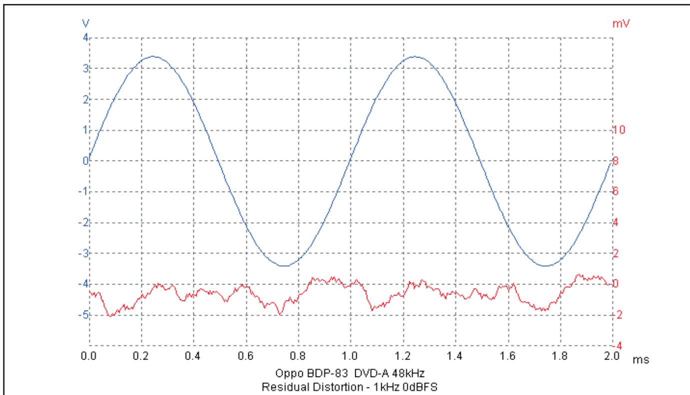

FIGURE 13: DVD48—1kHz residual distortion.

(**Fig. 18**) shows the 1kHz intermodulation difference product to be -87dB (0.0045%). The 18kHz product is below –90dB and the 19kHz and 21kHz products are -85dB and -82dB, respectively.

Figure 19 shows the DVD-A spectrum of response to the 19kHz + 20kHz IMD signals using 24/48 test data, from DC to 20.8kHz. The 1kHz intermodulation distortion (IMD) difference product measures –94dB, a bit lower than the CD test. The 19kHz and 21kHz products are at –87dB and there is a spike at 2kHz of –98dB. Recording DVD-A 24/96 and SACD PCM data (not shown) produced essentially the same results as the 24/48 test data at the same IMD frequencies.

All the IMD products in the SACD DSD mode reproduction of the 19kHz + 20kHz intermodulation test signal in **Fig. 20** are below –90dBfs. Note the continuously rising noise floor, which is the result of the HF noise shaping.

The CD playback of a 0dBfs square wave at 997Hz (**Fig. 21**) exhibits the Gibbs-phenomenon ringing associated with the steep digital filters used in the BDP-83. This ringing is supposed to be

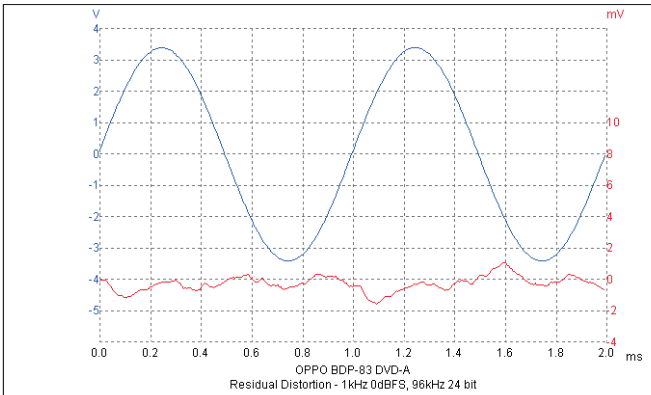

FIGURE 14: DVD96—1kHz residual distortion.

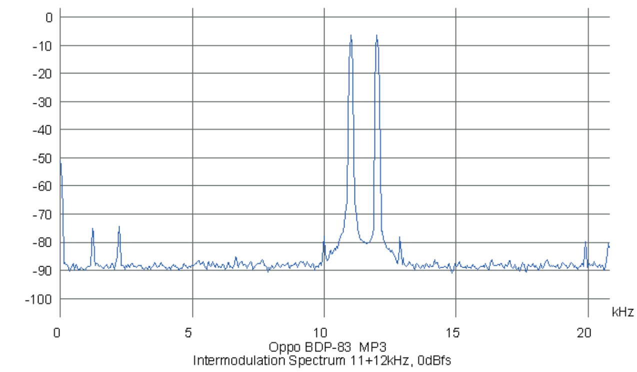

FIGURE 17: MP3—IMD 11±12kHz.

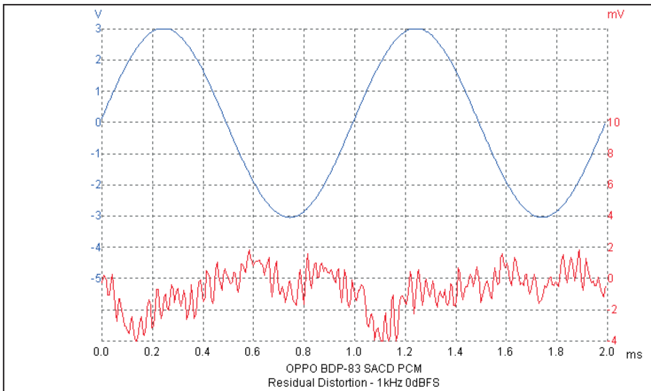

FIGURE 15: SACD PCM—1kHz residual distortion.

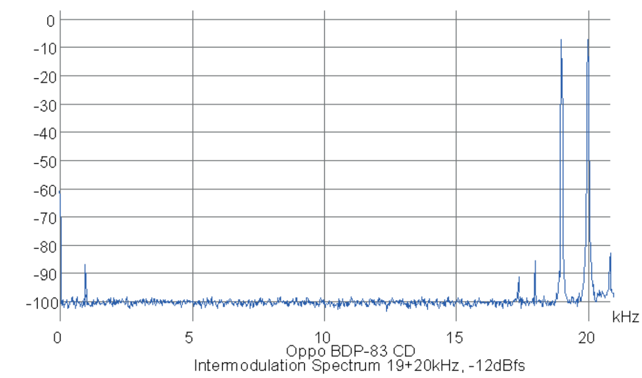

FIGURE 18: CD—IMD 19±20kHz.

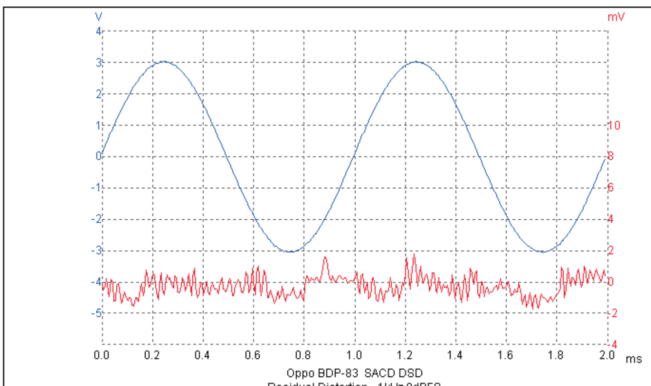

FIGURE 16: SACD DSD—1kHz residual distortion.

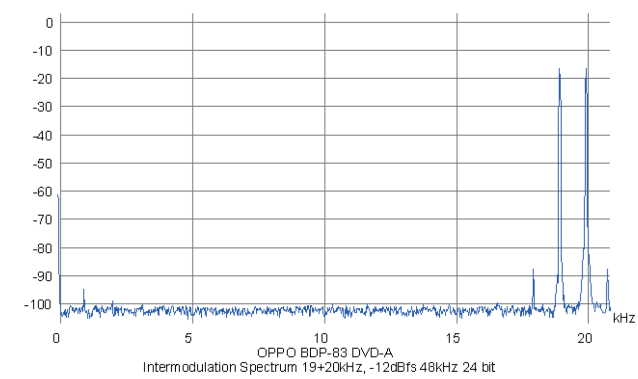

FIGURE 19: DVD48—IMD 19±20kHz.

a symmetrical damped sinusoid with peaks at the leading and trailing edges, and a minimum amplitude in the center. However, the square wave ringing here peaks at the rise/fall and fades down toward the end of the square wave instead of being symmetrical. This response is probably associated with the HF peaking I see in the analog output versus frequency curves.

Reproduction of the same square wave ripped to MP3 showed similar clipping (**Fig. 22**). Note that the Gibbs pre- and post-echo ripple has a total of 16 "pulses" per full cycle, compared with the 22 pulse CD playback. You can approximate the PCM Nyquist frequency response limit by multiplying the number of pulses by the fundamental square wave frequency. This demonstrates the more limited high-frequency response available as a result of the MP3 compression algorithm.

The playback of a 1kHz 0dB square wave in 24/48 DVD-A mode (**Fig. 23**) shows the 24 pulse Gibbs ripple associated with the higher resolution 24-bit DAC performance. There is again a noticeable nonsymmetry associated with this test. The 48 pulse Gibbs ripple associated with the 24-bit 96kHz DAC output (**Fig. 24**) accentuates the nonsymmetry of this higher resolution test.

The SACD PCM mode 1008Hz square wave Gibbs-phenomenon ringing in **Fig. 25** appears to be near 80kHz, but I can't be sure because the ringing peaks at the rise/fall and fades before the trailing end of the square wave, making it difficult to count the ripple pulses.

The leading edge peak of the SACD 1008Hz square wave output at 0dBfs in **Fig. 26** indicates the presence of high-frequency peaking in the response, just like that which would occur with all-analog test signals. I don't believe the Gibbs phenomenon occurs with the 1-bit delta-sigma (ΔΣ) DAC operating in the SACD DSD mode. DSD reproduction does not require the steep digital filters needed for PCM conversion to analog.

Figure 27 shows the CD reproduction of an undithered 997Hz sine wave at –90.31dBfs. At this level the signal consists of ±1 bit of data, producing two different voltage levels that are symmetrical about the horizontal axis (time). These noisy discrete voltage steps are recognizable, but not ideal. Repeating the test with a dithered 997Hz sine wave at –90.31dBfs (**Fig. 28**) further detracts from the discrete voltage steps.

The same CD track ripped and

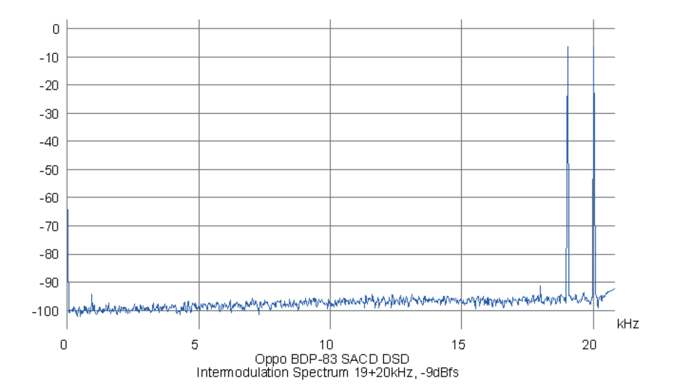

FIGURE 20: SACD DSD—IMD 19±20kHz.

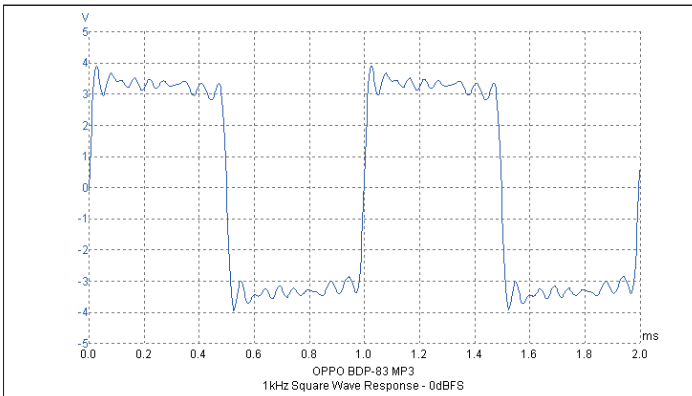

FIGURE 22: MP3—1kHz square wave.

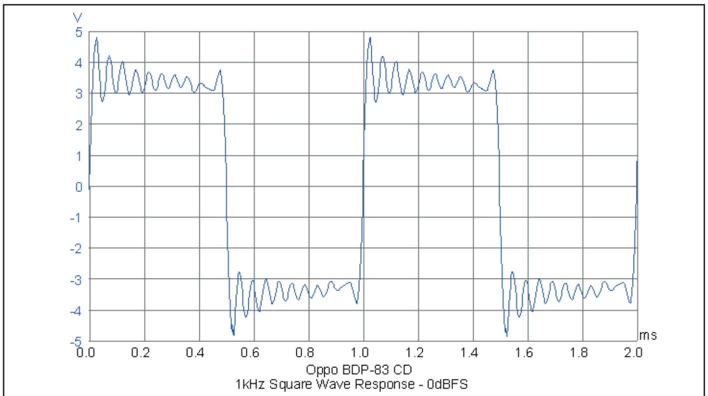

FIGURE 21: CD—1kHz square wave.

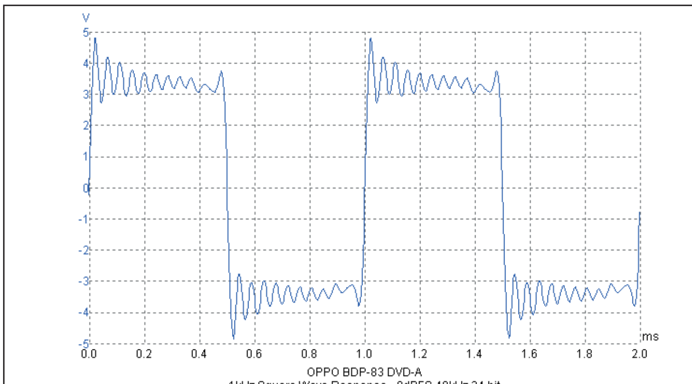

FIGURE 23: DVD48—1kHz square wave 24 bit.

played in MP3 mode shows that noise and compression artifacts have obliterated the vertical transitions in the output voltage (**Fig. 29**). The amplitude is only about 40% of what it should be if the MP3 conversion were perfectly linear.

A repeat of the -90dBfs signal level using the 16-bit 48kHz DVD-A test track (**Fig. 30**) shows a less noisy resolution of the ±1 bit sine wave. There is a noticeable skew in the horizontal portions of the waveform, however. Extending the bit depth to 24-bit, but still at 48kHz sample rate, shows the improvement produced by the ±9-bit sine wave (**Fig. 31**).

Extending the -90dBfs signal sampling to 24/96 DVD-A (not shown) improves the sine wave by virtue of the lower noise floor, but there is not a significant improvement with my DSO captures compared to **Fig. 31**. A similar situation exists with SACD in PCM mode.

Noise took its toll on the reproduction of the –90dBfs SACD test track at 1008Hz in DSD mode. While the waveform in **Fig. 32** is recognizable as a sine wave, it is modulated with signifi-

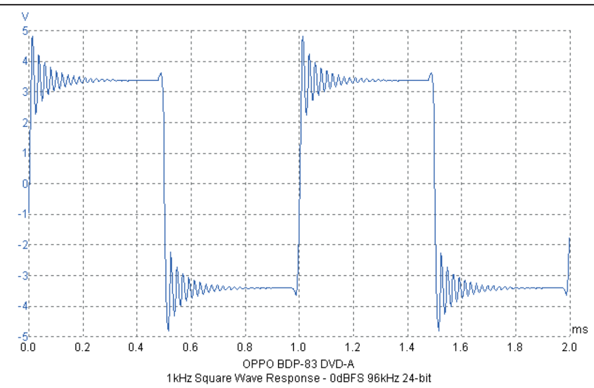

FIGURE 24: DVD96—1kHz square wave 24 bit.

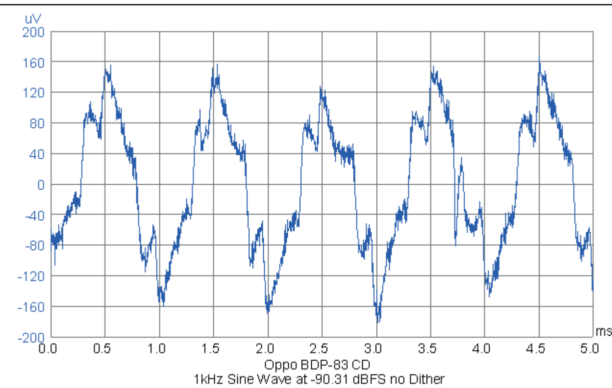

FIGURE 27: CD: -90dB 1kHz sine, no dither.

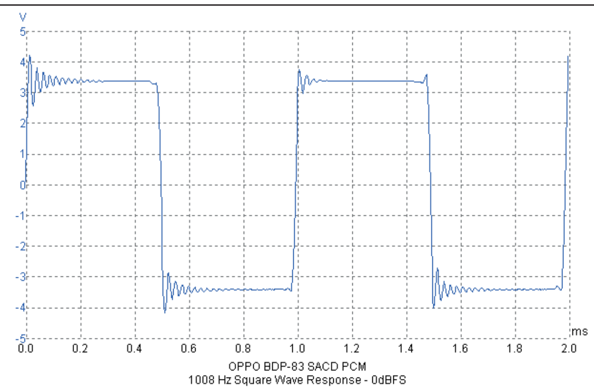

FIGURE 25: SACD PCM—1kHz square wave.

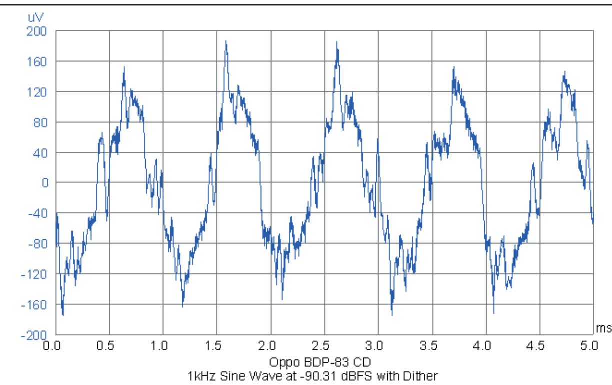

FIGURE 28: CD: -90dB 1kHz sine with dither.

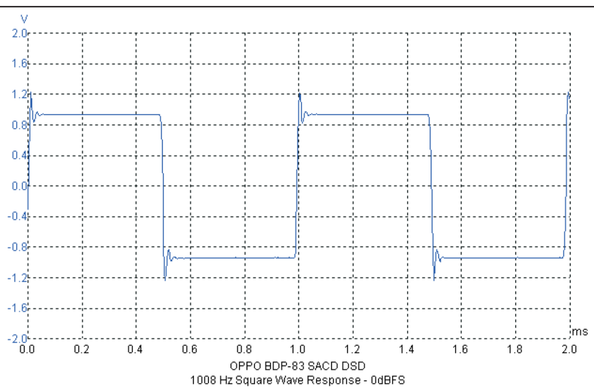

FIGURE 26: SACD DSD—1kHz square wave.

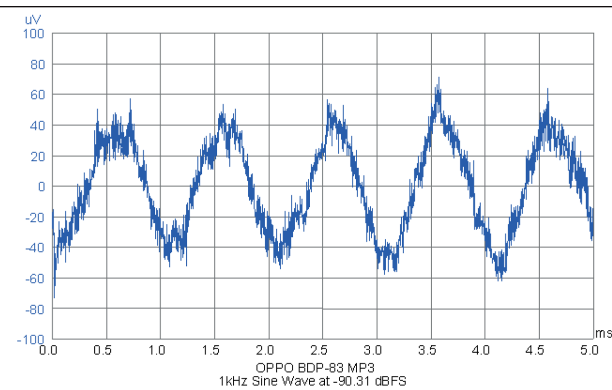

FIGURE 29: MP3: -90dB 1kHz sine, no dither.

cant amounts of HF noise.

To further explore the true noise floor of the BDP-83, I used a 997Hz -100dBfs test signal with dither from the CD test disk that I preamplified with a low-noise external gain block. **Figure 33** shows a CD noise floor of about -125dB, with barely noticeable 3rd and 5th harmonics. Compare this with the BDP-83 in STOP mode in **Fig. 34,** which only shows very low levels of power line harmonics.

The noise floor is much higher with MP3 files. The -100dB test signal produces a picket fence of 1kHz harmonics (**Fig. 35**). I can't explain the large spike at 20kHz because MP3 frequency response does not extend out that far, as you can see back in **Fig. 1**.

The spectrum of a 1008Hz SACD DSD test signal at -160dB (preamplified, in **Fig. 36**) shows the higher noise floor evident in earlier figures in this report. It appears that no muting occurred during this test, but I'm not sure why given the low level of the test track. The 1008Hz signal and any of its harmonics are obscured below the noise.

For the final test, I examined the spectrum extended to 166kHz while playing a 1kHz SACD DSD test track with the BDP-83 in DSD mode, at –90.31dB to positively prevent output muting.

This is a product of the 1-bit technology used in DSD. The delta-sigma DACs alter the noise spectrum of the audio signal, so the SACD DSD processor uses aggressive low-frequency noise shaping to move the noise levels up beyond the audio band, increasing the out-of-band noise. A conventional 2nd or 3rd-order analog filter at the output of the DAC then limits the HF noise that is produced at the analog output. You can see this effect in the extended spectrum analysis of the 1kHz –90dB SACD test sine wave with the passive 20kHz LP pre-filter removed (**Fig. 37**). The HF noise levels out, or shelves, above the DAC analog LP filter breakpoint.

The SACD noise shaping should produce a roughly constant downward slope in the noise floor below the HF shelving point. However, the BDP-83 SACD output seems to hit a LF noise floor that is higher than the noise floor seen in the PCM modes (CD and DVD-A). This higher noise floor is reflected in the THD+N measurements.

CONCLUSION

The Oppo BDP-83 offers a fine measured performance with a low noise floor. At this price point it is a real bar-

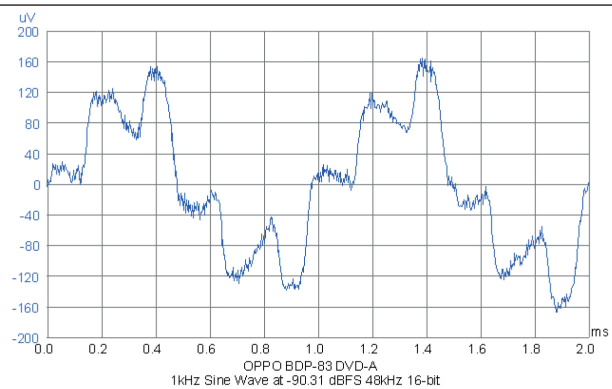

FIGURE 30: DVD48: -90dB 16bit 1kHz sine.

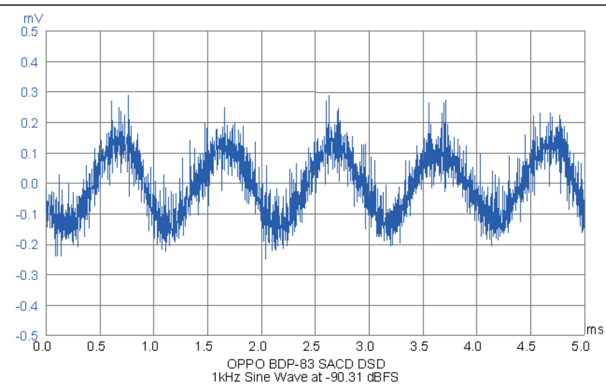

FIGURE 32: SACD DSD: -90dB 1kHz sine.

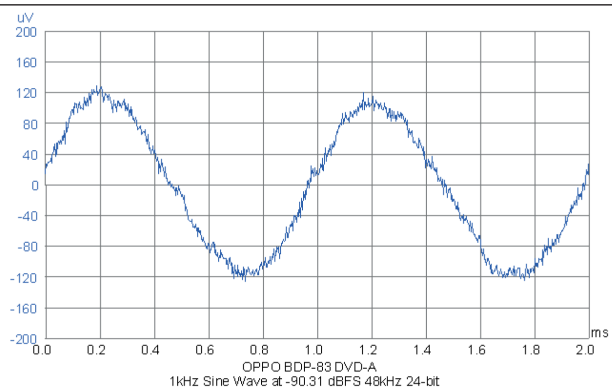

FIGURE 31: DVD48: -90dB 24bit 1kHz sine.

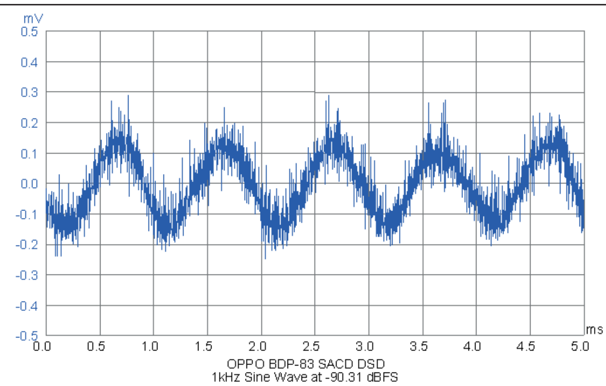

FIGURE 33: CD spectrum: -100dB 1kHz with dither.

gain as well. (Part 1 of this review appears in the August issue.)

MANUFACTURER'S RESPONSE

On behalf of everyone here at OPPO Digital, I would like to thank Mr. Gary Galo, Mr. Chuck Hansen, and the editorial staff at *audioXpress* for the insightful review and detailed measurements. Building a universal Blu-ray player with solid performance is not an easy undertaking. We are proud that the BDP-83 is selected by *audioXpress* for such in-depth coverage. OPPO is a relatively small brand in the world of audio/video equipment manufacturers. We owe our progress to the invaluable input from our enthusiastic customers and industry experts like the two reviewers here. By listening to their feedback, we hope to continue improving the products and supporting our users with excellent customer service.

Jason Liao, OPPO Digital, Inc.

REFERENCE

It is a real bother to run the DVD-Audio tests. I need to string three long composite video cable sets from my lab to my wife's 26″ Toshiba widescreen and connect the video cables together with RCA couplers so I can see the menus. Then I must run back and forth from the lab to the TV to select test tracks and volumes. Many of the tracks are only 30 seconds, so it means much repetition to capture DVD-Audio test data. The CD, MP3, and SACD do not require a monitor. Am I the only one who can't understand why stereo DVD-A disks need to use video chapters and menus? ***rr***

TABLE 1 MEASURED PERFORMANCE

Parameter	Manufacturer's Rating	Measured Results
Frequency Response (-3dB)	20Hz – 20kHz (±0.4dB)	20Hz – 20kHz, ±0.04dB
S/N ratio	>110dB	125dB, 1kHz, "A" weighted
Total Harmonic Distortion	<0.002%	0.0056% 0dB 997Hz CD 0.034% 0dB 997Hz MP3 0.004% 0dB 1kHz DVD-A 24-bit, 48kHz and 96kHz 0.004% 0dB 1008Hz SACD PCM 0.0063% 0dB 1008Hz SACD DSD
Power Requirements	35W maximum, 0.5W Stand-by	verified

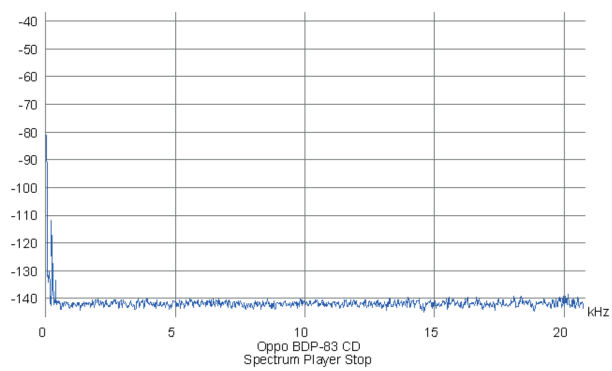

FIGURE 34: CD spectrum player stop.

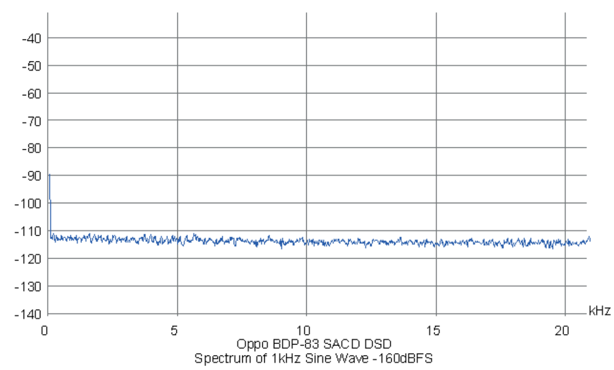

FIGURE 36: SACD DSD spectrum: -160dB 1kHz sine.

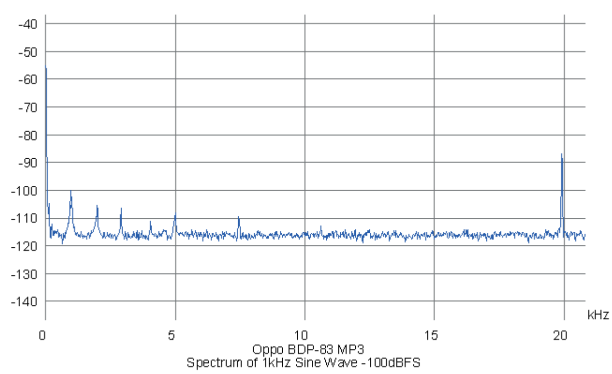

FIGURE 35: MP3 spectrum: -100dB 1kHz sine.

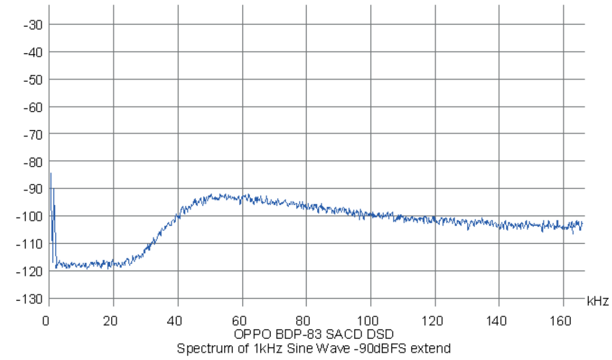

FIGURE 37: SACD DSD spectrum: -90dB 1kHz sine extended noise shaping.

By Keith Levkoff

Personal Digital Recorders, Part 3

We conclude this series on the advantages of using a PDR for recording

(see Parts 1 and 2 in 3/10 and 4/10 aX, respectively).

Zoom H2.

WORKFLOW #1—DIGITIZING AN ALBUM OR TAPE

When I digitize an album, I usually produce the following: a backup copy at the best possible quality (for future use), a copy on CD, and CD-quality digital files to put on my Squeezebox music server. I end up with a separate file for each song so I can work individually with them later.

The most sensible way to start the project is to record the album at 96/24 (96k sample rate/24 bits of depth), and to record the entire album onto a single file (or one for each side). Odds are that if I need to adjust the equalization at all, the settings will be the same for all the songs on that album, and I'll want to maintain the same relative level setting for all the songs. (Everything is about the same for a tape source or LP, except that I probably wouldn't bother with a high sample rate for a cassette source.)

When I open the song in my editor, I see a display that looks like an oscilloscope presentation with a very long time scale (the entire song will fit onto the screen at once). My editor will let me zoom the time frame up to a millisecond or so occupying the entire screen. The important thing is that I can see where the loud and quiet parts are, and the spaces between songs (**Fig. 1**).

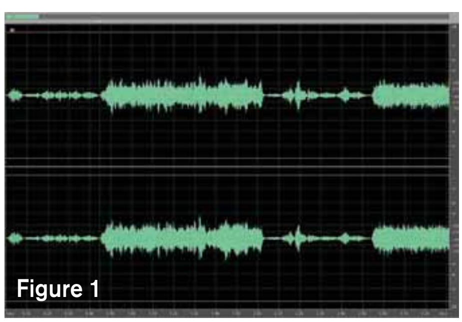

Figure 1

Because I want to keep the loudness of all the songs the same relative to each other, I adjust the overall loudness while I've still got them all in one single file. I prefer to set levels so that the loudest part of my entire project is at about -3dB. My next choice depends on whether there are any loud ticks or pops on the recording. If they are not present, I can use something called Normalize, and simply tell the editor to adjust the overall level so that the level of the loudest part of the file is at -3dB (the editor will take care of all the details, and even fix the channel balance at the same time if I tell it to).

HERE'S HOW TO NORMALIZE THE LEVEL:

In Audition:
Press Ctrl-A to highlight the entire recording

Select **Effects > Amplitude and Compression > Normalize**

Check the top three boxes, enter the level to normalize to (-3dB) and click OK.

In Audacity:
Press Ctrl-A to highlight the entire recording

Select **Effect > Normalize**

Check both boxes, enter the level to normalize to (-3dB) and click **OK**.

If there are loud ticks, the editor will see them as the loudest parts and set the levels accordingly, which isn't what you want. Therefore, you have the choice of either cutting out or squashing the loud ticks before normalizing the levels, or using the level control and display to manually adjust the level (and letting the ticks clip). To delete short ticks or pops, simply highlight each and press the Delete key. This will delete the highlighted area and seamlessly join the ends together (both editors have the ability to automatically move the cuts to the nearest zero crossing for more precision—read the documentation for details).

HERE'S HOW TO REDUCE THE LEVEL OF SELECTED SOUNDS OR INTERVALS:

In Audition:
Highlight the area or sound you wish to reduce in volume.

Notice the little volume control icon which appears toward the top of the selected area. Click on this icon and move the mouse to the left while holding the button down to reduce the level (simply slide the mouse left or right; you are not trying to turn the little knob and you don't need to stay over the icon while sliding).

In Audacity:
Highlight the area or sound you wish to reduce in volume.

Select **Effect > Amplify**

Slide the Amplification slider to where you prefer and click **OK**.

(If you don't like the result, you can use Undo (Ctrl-Z) to cancel it and try again.)

Once the loud spots and clicks are gone, you can apply the normalization successfully.

At this point, apply any necessary equalization. Most of the equalizer modules have a gain control, which you can adjust so the overall level remains the same if you apply a lot of boost. If planning to apply a huge amount of boost, you could also have done the equalization before the normalization. (Of course, you want to avoid clipping. You also want to avoid lowering the level excessively and then raising it. Since we're editing at 24 bits of depth, level reduction of up to 30dB or so won't hurt anything.)

All audio editors also include various noise removal features (one for background hiss, one for ticks and pops, and usually a few assorted notch filters and dynamic processors). This is the point where I apply those (if necessary) as well. You'll need to read the advanced book—or the manual—to find out about those.

Audition 3 offers several different equalizer modules under **Effects > Filter and EQ**. These include a graphic equalizer (10, 20, or 30 bands), a parametric equalizer (up to five bands, everything variable), and an FFT filter (where you can draw a frequency response graph and set several other options). It also offers several dynamic processing options (actually, an awful lot of them). Audacity offers a sort of combined equalizer which does both under **Effect > Equalization**.

HERE'S HOW TO ADJUST EQUALIZATION USING THE GRAPHIC EQUALIZER:

In Audition:
Highlight the area or sound you wish to equalize (probably the entire song).

Select **Effects > Filter and Eq > Graphic Equalizer**

Use the sliders to get the equalization curve you want. Use the Master Gain slider to compensate for any net increase or decrease in level.

Click the little Play icon in the lower left corner of the Equalizer screen to get a preview (in Audition, the Preview is live—you hear the changes applied to a sample as you make them). Click **OK** when you're satisfied to apply your changes to the selection.

In Audacity:
Highlight the area or sound you wish to equalize (probably the entire song).

Select **Effect > Equalization**

Use the controls to get the equalization curve you want (try different options). Click **Preview** to hear a short sample played using your settings (in Audacity, the preview is not live; you must click the button whenever you want to hear a preview). Click **OK** when you're satisfied to apply your changes to the selection.

Finally, I divide the file into separate songs by selecting each song separately and saving it to a file. I prefer to save each song as a separate file at the original sample rate and bit depth, leaving a bit of extra space before and after each. I then reopen each file, clean up the beginning and end carefully, and re-save it. Of course, you can just do it carefully the first time.

HERE'S HOW TO SAVE EACH SONG AS A SEPARATE FILE:

In Audition:
Select the song you want to save

Select **File > Save Selection**

Enter a file name, select the folder where you want to save it, and click **Save** (the Save As Type box should say Windows PCM (WAV)—the same as our original). (remember to use a descriptive file name)

In Audacity:
Select the song you want to save

Select **File > Export Selection**

Click **OK** when asked to Edit Met data (the dialog appears because of a bug in the current release of Audacity). Enter a file name, select the folder where you want to save it, and click **Save** (the type should show as Wave (Microsoft) Signed PCM of the appropriate bit depth).

I did this step last because I specifically planned to set the level and equalization the same for all the songs—so I set that while they were all still one single file. At this point I can adjust any song individually if I want to. The song name should be descriptive. Also remember that most computer programs list files in alphabetical order by name so it makes sense to start the names with numbers to keep the songs in a particular order (if you're going into double digits, use 01 rather than 1; otherwise, the sorting will become messed up). The most important thing is to be consistent, and use a system that makes sense to you, so those names make sense when you see them again years from now.

(There are file name length limits in Windows. Also note that some storage methods, such as CD-Rs, have slightly different length limits than Windows. Upper and lower case don't matter.)

I usually use something like:
00-LadyGaga_NYC_Oct2009_9624.wav
01-FIRST_SONG_4424.WAV
01-FIRST_SONG_9624.WAV
02-SECOND_SONG_4424.WAV
02-SECOND_SONG_9624.WAV

If I'm making different copies at different sample rates or bit depths, I usually save all the individual songs at the highest sample rate and bit depth first. I then open each file, clean up the lead-in and lead-out carefully, and resave it at the same quality. Finally, I convert it to the lower sample rate and bit depth, and then save it again under a different name. Remember to use different names. Also remember that converting to a lower sample rate or bit depth is a one-way process; information is actually discarded when you convert to a lower sample rate or bit depth.

(When you convert a file to a lower sample rate or bit depth, the extra information is discarded. When you convert a file to a higher sample rate or bit depth, the extra information storage space produced is filled with zeros.)

HERE'S HOW TO CHANGE THE SAMPLE RATE AND BIT DEPTH (THIS ASSUMES YOU ALREADY HAVE THE FILE OPEN):

In Audition:
Select **Edit > Convert Sample Type**

For CD quality set Sample Rate: 44100, Channels: Stereo, Bit Depth: 16

Leave the defaults for the rest

Click **OK** (save the new song file)

In Audacity:
Select **Tracks > Resample**

Select 44100 from the pull-down box

Click **OK**
(save the new song file)

(Audition does offer superior options for this process, and the dither provided by Audition should provide a slightly better signal-to-noise ratio in the final output. In practice, however, the noise present on any record or tape provides the same dithering effect as the software is offering. Dithering is critical in situations where the signal-to-noise ratio of the source material approaches that of the bit depth being used—which is over 120dB for 16-bit files.)

WORKFLOW #2—LIVE RECORDING

When I do a full editing job on a live recording, I usually want about the same things as from a digitized album: a backup copy at the best possible quality (for future use), a copy on CD, and CD-quality digital files to put on my Squeezebox music server. I want to end up with a separate file for each song so I can work with it individually later. I may record the original at either a 96k or 44k sample rate—depending on the venue and which microphones I'm using. (It doesn't make much sense to use 96k when I'm using the internal microphones on the Zoom H2 to record a band playing through the PA system at a bar.)

The bit depth of 24 bits is even more important, however, since I'll be dealing with wide dynamic range and somewhat unpredictable levels. I always record the entire session into a single file, and I expect that the equalization requirements won't change for each song. I also usually prefer to preserve the live dynamics (so I leave loud songs loud and quiet ones quiet).

When I edit a live recording, I usually prefer to leave some applause at the end of each song for atmosphere. Specifically, I usually arrange for each song to end with a short clip of 10 or 20 seconds of applause that fades to a few seconds of silence before the next song. If the songs are run together, and I can't separate them neatly, I sometimes leave a few seconds of the song before or after attached and do a quick fade-in or fade-out. To me, this is part of why live recordings sound live.

My preference for leaving those segments of applause causes me one complication. With live recordings, the applause is often by far the loudest part of the recording (especially if I was recording from a seat in the audience). This will interfere with my adjusting the overall loudness because those segments of loud applause will mess up the Normalize option and, if I adjust the levels manually, the applause will probably clip badly. I want to keep some of that applause, but at a reduced level.

To accomplish this, I usually go through the entire recording and delete any areas of applause other than the ones I want. Then, if any of those remaining areas are louder than the loudest part of the music, I select it and reduce the level a bit (it doesn't matter how much at this point as long as it ends up quieter than the loudest parts of the music). This process sounds difficult, but with the oscilloscope view in the editor, it's very simple to visually locate song breaks and areas of applause. I can also position the cursor and click the Play button to hear any specific spot I like (no need to fast forward or rewind to it).

At this point I also reduce or remove any short loud sounds (such as chair clunks and coughs) that are louder than the music. (If you're using Audition, the Spectrum View is great for this.) Now I can adjust the levels using the normalize feature (again, I prefer my loudest spots to be around -3dB).

I usually prefer to keep the relative level of the songs as they were originally—which is why I normalize the level of the entire file before dividing it. If you prefer, you can normalize each song individually after you separate them (in which case the loudest part of each song will be set to the value you enter). You can also, of course, manually adjust the level of each song and forego the normalize function entirely. (Note that the Normalize process changes the level of the entire file being normalized equally; it does not alter the dynamic range or compress the audio within that file.)

At this point, I correct the equalization for the entire recording. Live recordings often require a lot of equalization, so you must be extra careful to avoid clipping. Luckily, with many live recordings, the main equalization requirement is a reduction in bass, which lowers the average level. You can also do the equalization before the level normalization (or you can re-normalize the entire file after equalizing).

Most editors include several different equalizers, including standard graphic and parametric ones as well as ones where you can draw the curve you want. Most also include all sorts of notch filters, dynamic compressors and expanders, and even intelligent noise removal filters that learn the noise profile for a particular file and then remove it. (Most of the serious filters work very well, but they are intrusive, so I try and avoid using them if possible.)

Now I divide the file into separate songs. I usually want all of them to include several seconds of applause after the song; I also want any that don't start cleanly to include a few seconds of the previous song or applause—so I can apply a fade-in later. Now, after all the songs are saved to separate files, I open each one individually and do the final editing.

On some songs I may choose to have the song start immediately; on those I delete any extra audio before the beginning. On others, I may leave a few seconds of applause, or the end of the previous song, and fade it in gradually (this is easy on a computer). On most, I'll leave about 10 or 15 seconds of the applause at the end, and fade that out to silence at the end. This is where you must listen very carefully; it sounds awful if you clip off the end of a long fading note.

This is also where I apply any additional equalization, level adjustment, or other changes that are necessary for each individual song (which should be done before applying the fade-in and fade-out). As I said in the previous workflow, use descriptive names when saving files, and save the high sample rate and high bit depth versions first since the reduction process is one-way.

HERE'S HOW TO FADE A SONG IN AND OUT:

In Audition:
Set your screen view so you can see the beginning of the song (try the very cool Top/Tails View for this). (It's best if the part you want the fade to apply to fills about one-tenth of the screen horizontally). Locate the little Fade-In icon (a little two-tone square at top left of the screen at the beginning of the song). Left click on the icon, hold the mouse button, and slide the fader toward the center of the screen (moving the fader up or down changes the shape of the fade envelope—this is a visual representation of what the amplitude is doing). You can see the change reflected in the actual waveform display. When you let go the change takes effect (you can undo it if you want to try again). Repeat this at the end of the song (the Fade-Out icon is at top right at the end). (There is also a menu option that allows you to draw any amplitude envelope you want.)

In Audacity:
Select (highlight) the part you want to apply the fade-in effect to (the part of the song you want to fade from silence to full volume)

Select **Effect > Fade In**

Select (highlight) the part you want to apply the fade-out effect to (the part of the song you want to fade from full volume to silence)

Select **Effect > Fade Out**

SAVING, COPYING, AND BACKING UP

I won't say much here except to remind you that computers occasionally lose or damage files, so make multiple copies and backups of anything that you value. I suggest making copies of your original recordings before editing them, and keeping them afterwards. If, for example, you later decide that your equalization was a bit aggressive (or not aggressive enough), you can then re-edit the original as you prefer.

When doing a live recording, I always copy my recordings immediately onto the computer when I finish (or get home). Then I make a backup copy from the computer and put it away somewhere. Only then do I proceed to the editing and duplicating.

BURNING TO AN AUDIO CD

There are many programs designed for producing audio CDs, and virtually all of them accept WAV files as their preferred input format. Some popular commercial ones are made by Sony, Sonic, Pinnacle, Nero, and Adobe. Most computers with CD or DVD writers installed come with a CD or DVD burning application already installed (all

DVD writers also write CDs). As you might expect, they range from simple ones with no options to complex "authoring" programs. Some programs do both disc editing and burning, while others split the functionality between two separate programs.

Some of the software included with computers doesn't include the ability to burn audio CDs (it's limited to making data backups). If so, you'll need to get another program. Unfortunately, many PC manufacturers include OEM versions of popular programs with their machines—which often lack some of the features of the regular commercial versions. Also, annoyingly, some programs don't work with certain brands or even certain models of drives.

For burning simple audio CDs, almost any audio CD writing program will do. (The fancier ones offer control over such functions as track spacing, index marks, and such). With most popular ones, creating a CD is as simple as dropping the files into the program and telling it to write a disc. Many of the popular programs offer demonstration versions or free trial periods. ImgBurn is a very sophisticated free program for authoring and burning CDs and DVDs (unfortunately, the process for creating audio CDs is rather arcane and the documentation is nonexistent). I included both the link for ImgBurn and for a tutorial about burning audio CDs with it. You can also Google "ImgBurn audio CD" and get several tutorials on the subject.

The process for burning an audio CD is different for each program, and is usually pretty simple, so I'm not going to describe it here. I will, however, include a few tips and hints:

1. Creating an audio CD is not the same process as copying WAV files to a data CD; audio CDs use a special format and data structure. Be sure to select the option in your program to "create an audio CD." Some programs intended strictly for data backup may not offer this option.
2. Most programs automatically make each WAV file you add into a separate track on the finished CD—which is usually what I would want anyway.
3. There are different methods of writing an audio CD—including disc at once (DAO) and track at once (TAO). Some of them insert a mandatory two-second gap between songs, while others do not.
4. After a disc is recorded, the disc must be closed (or finalized). Some programs offer the option of not closing the disc so that more content can be added later. I strongly recommend you avoid this option—always close your discs. Open discs can usually only be read on a recorder; they cannot be played on standard audio CD drives. You may also lose your content entirely if the disc is later improperly closed or added to.
5. Audio CD-Rs recorded on a computer will play reliably on virtually any computer, and on most modern audio CD players (including portables and DVD players). Some older audio players won't play them, or will do so unreliably. If that happens, you can try different brands of discs, but the only real solution is to get a player that is rated to play CD-Rs.
6. Along with standard CD-Rs (CD recordables), you can buy re-recordable discs called CD-RWs—which can be erased and reused. These are more expensive and less reliable than CD-Rs, and will not play at all on most audio CD players; I avoid them.

USING A PDR AS A POWERED MICROPHONE

Because most PDRs—including the Zoom H2—include a pair of microphones and microphone preamps, and provide line-level outputs, you can use them as powered, amplified microphones. You can connect them to the line-level input of a tape recorder or amplifier, or to the line-level input of a computer sound card.

USING A PDR AS A DIGITAL SOUND CARD

You can connect some PDRs (again including the Zoom H2) to a computer via USB cable and use them as a sound card. When connected this way, the Zoom H2 simply becomes an external sound card for the computer. The computer's audio output is available at the line outputs of the Zoom, and the Zoom's input (which can be line in, internal microphones, or external microphones) is seen by the computer as a sound card input. This is especially convenient with laptop computers, which often have very poor microphone inputs (which are often not stereo). If you use the internal microphones, this might also be referred to as a stereo USB microphone.

To configure the Zoom this way, you simply connect it to your computer via the USB cable, and choose the appropriate setting on the Zoom's screen (it will ask when the connection is detected). Depending on the configuration of your computer, you may also be required to choose the Zoom ("USB sound card") as the active sound card in Windows.

USING A PDR AS A SIGNAL GENERATOR

You can use any PDR or computer sound card as an audio signal generator. All audio editing programs allow you to produce low distortion test tones (including filtered noise and sweep tones) quite easily. You can play those test tones directly, or record them to a file, and play them later. If you record a series of test tones, then copy them onto an SD card for your Zoom H2, you can use the Zoom as a convenient portable test generator. (You can also record test tones to a CD or DVD.)

The benefits of using a PDR for this include portability, low noise, and low distortion. You must remember, however, that the frequency response is limited by the sample rate. For example, a standard 44.1k sample rate file cannot contain any frequencies above about 21kHz. Any tones or audio you record or produce electronically will appear as if filtered by a very sharp high-cut filter at about 22kHz, so a 13kHz square wave will become a sine wave. By using a sample rate of 96k, you can theoretically extend this limit to about 44kHz (I haven't tested the Zoom, so I don't know whether or not the frequency response of the analog circuitry will support this).

USING A PDR AS A DIGITAL OSCILLOSCOPE

Because any digital editor will display a waveform, you can use any PDR or computer sound card as a digital oscilloscope. Unfortunately, the bandwidth is limited by the Nyquist frequency. At best, you'll have a 44kHz oscilloscope—which might be useful for woofer measurements. However, you can use the line inputs of a PDR (or even the microphones) as the input for an audio spectrum analyzer program as long as you remember the limitations.

SUMMARY

I hope I've given you all the information you need to start recording and editing audio using your PDR (and a computer). If you find digital editing as fascinating as I do, I strongly urge you to read further on the subject. There are many good books on the subject and lots of information on the web (although, unfortunately, some of it isn't accurate—caveat emptor). The manuals that go with Audition (even the free download version) and Audacity are a good place to start.

LINKS

Here are some links you might find useful:

The Zoom H2 product information page:
www.zoom.co.jp/english/products/h2/

The Zoom H2 manual (about halfway down the page):
www.zoom.co.jp/english/download/manual/english.php

The Adobe Audition product information page:
www.adobe.com/products/audition/

The Audacity product information page:
http://audacity.sourceforge.net/

The ImgBurn product page:
www.imgburn.com/

Here's a description about how to burn audio CDs using ImgBurn
(ImgBurn doesn't include much documentation):
http://blog.wolffmyren.com/2008/08/05/imgburn-creating-an-audio-cd/

A FEW SUGGESTIONS FOR PORTABLE RECORDING

Here are a few hints for successful portable recording using your PDR:

1. *Start a session with fresh batteries.* The Zoom H2 will record over four hours on a set of alkaline AA cells (more on high-capacity NiMh cells). I always start a session with new batteries, and then use them up in something else afterwards. (You don't get any conspicuous warnings when batteries die—just a tiny little battery gauge.)
2. *Non-rechargeable batteries are more reliable.* I prefer non-rechargeable cells for important recordings. High-capacity HiMh batteries last very well, and are usually reliable, but occasionally they'll die for no apparent reason. I've never had a set of primary cells do that.
3. *Carry spare batteries.* I always carry a spare set of regular alkaline cells and a pair of expensive one-shot lithium AA cells. (The lithium cells run much longer than alkalines, and have a very long shelf life, so they're great to keep handy for emergencies).
4. *Always carry a few spare SD cards.* They're cheap and you never know when you'll need them.
5. *Always empty your memory chips when you get home, or replace the full ones with empties.* I like being able to grab an SD card from my bag without worrying if it's already full, and the tiny time gauge on many PDRs is difficult to read and easy to misread when you're in the dark or in a hurry. I also almost always use the same size of SD card (4gB) just so I don't need to keep track of different sizes.
6. *Use a card reader.* I always use a card reader to copy my recordings into the computer. This lets me immediately swap an empty card into the recorder, which is then ready for next time. Card readers are also usually much faster than downloading through a cable, and more reliable. Some card readers are much faster than others (download speed is limited by the slower of your USB port, your card, and your reader). I like the Sandisk MicroMate reader, which is not that expensive and it's very fast.
7. *Make sure the SD cards you buy work in your PDR.* Use either one of the brands recommended by your PDR, or a major brand such as Sandisk or Lexar; some brands of cards just don't work well in some recorders. Don't experiment with a new brand or type of card during an important recording. (When they don't work,

sometimes they don't work at all, and you may not find out until too late.)

8. *SD cards are not all the same.* SD cards come in different speeds. Most PDRs don't especially require fast memory, and they won't perform better with faster memory (unless the manual says to use it). The sole exception is if your PDR worked fine at 44/24 but seemed to have trouble at 96/24, which might indicate a data rate issue (I've never seen this happen). Faster cards do, however, copy files to and from your computer a lot faster, and won't cause any sort of problems. (I find it easier to use the same SD cards for my PDR and my camera, which performs better with faster cards.)

DIGITAL RECORDING—PDR VS. PC

Here is a short comparison of the differences between recording digital audio on a PC and on a PDR:

1. *Portability.* A PDR is much more portable than a laptop or desktop computer. Most can also use standard batteries, which is important if you're not near an outlet for your laptop charger.
2. *Quality.* The sound cards that ship with most computers are usually very poor quality, and sometimes even a good sound card will perform poorly due to power supply and environmental issues inside a computer. The audio quality of a PDR is consistently equal to or better than that of the best computer sound card.
3. *Capacity.* Computer hard drives can hold much more than the memory cards used by PDRs, but you can change memory cards in your PDR (and carry spares). This also makes it easier to organize your recordings by separating them onto different memory cards.
4. *Isolation.* Since a PDR runs on batteries, you don't need to worry about ground loops and power supply noise. This makes life much simpler.
5. *Dedication.* Virtually any computer will run more than your recording software. (Even with the most careful configuration, this is almost unavoidable.) Since digital recording is a real-time process, your recording will be ruined if some other program demands attention while you are recording. Even the most powerful modern computer can have problems with this, and using heavy-duty programs such as Adobe Audition makes it worse. Because it is a dedicated hardware device, a PDR doesn't have this problem. This alone is sufficient reason to use a PDR (especially if your computer isn't blazingly fast).

It's sort of obvious why a PDR makes more sense for recording live events; it may be a little less obvious why it makes sense to use a PDR to digitize audio at home as well. For me, it's a combination of the last two—isolation and dedication. As for isolation, let's just say that computers are not designed by audio engineers. They tend to have noisy grounds, which don't always mix well with audio equipment. Even many laptops generate enough noise to be an issue. Besides all that, just getting the grounds right on one more piece of equipment is a hassle—so why bother?

For me, though, the main reason I prefer a PDR is that it is a *dedicated* hardware device. What I really hate about the "dedication issue" is that it can be sneaky. Digital *editing* is absolutely reliable; if the computer slows down, the process simply takes longer, but the end result is still perfect audio.

Digital *recording*, on the other hand, requires that the computer be ready to accept each sample when the sound card is ready to deliver it. If not, the sound card moves on, the sample is lost, and the program records a blank in its place. (This is called "a dropped sample.") In theory the program can take measures to avoid this; in practice those measures don't always work. The results range from occasional clicks in the audio that sound like record ticks to a stuttering effect that sounds something like a really bad record skip. The worst part of it is that the recording software won't warn you if it happens; you won't know it happened until you hear it later. I consider this risk to be the single bad feature about digital audio.

(Editing consoles avoid this problem by modifying the operating system to give absolute priority to the audio stream, and by other software customizations. You can avoid it by using a very fast computer, configuring it very carefully, choosing your software carefully, and not installing much other software on it. The easiest and most reliable way to avoid it is to use a dedicated recorder—a PDR.) *aX*

audio news

Compiled by Richard Surrette

The USBscope50, from **Saelig Company, Inc.,** combines a very small form factor with the high performance associated with much larger bench-type products. Its modular design allows you to configure a single channel or up to four channels with multiple adapters. Powered from a USB port, the USB-scope50 includes a proprietary synchronization system for tight signal channel-to-channel matching and concurrent triggering. It offers AC/DC coupling, 50MSps 8-bit sampling, and displays measured signals with scope software. For more details, please visit www.saelig.com.

Th ST69, from **Sterling Audio**, is a large diaphragm, multi-pattern tube condenser microphone with a high-pass filter and –10dB pad included as a complete kit. The power supply is only one switch, while pattern selection, featuring cardioid, figure-eight, and omni, is on the rear of the mike. Also included are a double-sided, multi-pattern, hand-assembled capsule, gold evaporated, Mylar diaphragm, custom nickel core output transformer, and SM4 shockmount. To learn more, go to www.sterlingaudio.net.

Aperion Audio introduces the Signature SLIM-stage 30 Soundbar, a speaker system for HDTVs. Designed to produce theater-like sound for flat-panel HDTVs in secondary viewing locations, the SLIMstage 30 includes a 31″ wide powered sound-bar, and utilizes Aperion's Bravus 8A subwoofer, providing 100W RMS of low-frequency power. For more information, visit AperionAudio.com.

The 45 Silver Stereo Amplifier, from **Electra-Print Audio Company**, features full silver wire windings and dual High Z and Low Z impedance input transformers, as well as point-to-point wiring. With a switchable high-pass filter and a no feedback circuit design, the 45 Silver Stereo also includes 21Hz to 21kHz ±1dB frequency response, 1.2% distortion, and 1.7W power. To learn more, go to www.electra-print.com.

Audience has upgraded its Adept Response aR2-TS, aR6-TS, and aR12-TS high-resolution power conditioners. All three models now incorporate Teflon capacitors with mono crystal copper wire leads, and a large ground plane with all wiring secured by welded connectors in a star ground configuration. For more info, go to www.audience-av.com.

Issues about Absolute Polarity are becoming increasingly relevant these days, and you can read two articles about it at http://www.ultrabitplatinum.com/?page_id=88 and http://www.ultrabitplatinum.com/?page_id=312.

New from **Tannoy** (www.tannoy.com), the Prestige Kensington SE speakers have a 10″ dual concentric driver and a paper cone-shaped woofer designed to optimally disperse sound from a 2′ aluminum alloy dome tweeter.

Allowing installers to quickly and easily produce custom-length HDMI assemblies of up to 39′ at resolutions of 1080p, 12-bit deep color, **BTX**'s Field-Terminatable HDMI Connector resolves the issue of fitting connectors through conduit and eliminates excess cable behind the rack or inside a wall (www.btx.com). The connector is compatible with HDMI 1.4 and is terminated through the insulation displacement connection process.

audio news

Parts Express is now providing on-site speaker repair and reconing services. The company offers expert-level restoration qualifications, and service is performed at their warehouse location. Using the Parts Express speaker repair lookup wizard, look up the speaker's manufacturer and model number to obtain an instant repair estimate. Then follow the instructions to submit your repair order, and prepare to ship your speaker. For more information please contact RobG@parts-express.com.

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letters

VERSUS

In regard to the editorial, "Amplifier Differences and Lack of Tests," (July '09, p. 5), bravo. This is a subject that deserves a *lot* of attention and gets very little. Perhaps this rates a column in every issue.

We certainly *do* need to do a lot more science because, when we fail to do so, the mystical claptrap creeps in to fill the gap. I think a lot of the problems arise because of a lack of understanding of psychoacoustics, or how our brains process sound. By this I mean both that audiophiles aren't aware of the current science, and that the current science is itself incomplete. (Truthfully, I don't think it's so much an AES conspiracy as simple lack of interest in the details by the public combined with a lack of ability to understand the information that does exist.)

As for tube amps, and why they sound better: We always assume that input is additive in what we listen to, but this is far from true. We forget masking, which is a major factor in what we hear and perceive. Most people agree that second harmonics sound nice, or at least a lot less annoying than third harmonics. It is also common knowledge that tube amplifiers usually produce quite a lot of second harmonic distortion.

The mystical theory is that there is some magical property that makes tube amplifiers sound good in spite of these high levels of distortion and that this tubey goodness overshadows the high distortion. I suggest that the exact reverse is true and that the *only* benefit is the distortion. (I am not, however, saying that tube lovers like just that second harmonic distortion.)

Depending on the frequencies involved, it is possible that the relatively benign second harmonic distortion is actually masking more objectionable third harmonic or other distortion. If so, then the other distortions won't measure any lower, but they will be perceived as being lower.

This is not voodoo. It is the science of psychoacoustics. Likewise, adding nonobjectionable noise can sometimes mask objectionable distortions as well. (This is how dithering and noise shaping work.)

Here's my theory about your experiences with harmonic structure: Let's start from two assumptions. The first is that, due to a combination of choice and necessity, a lot of information is omitted in commercial recordings. Modern techniques such as multiple close microphones sometimes require that bandwidth be limited to prevent interactions, and modern audiences simply favor clean sound.

The second is that tube amplifiers, especially triode amplifiers with little or no feedback, produce lots of relatively benign second harmonic distortion. The total perceived sound of the recording is a complex combination of lots of sounds, many of which mask others to some degree. In a live situation, there is a lot of harmonic content and various types of noise.

If you artificially reduce the harmonic content and noise, the result is that you will emphasize other content that what you reduced was previously masking, making it appear to be over-emphasized. My guess is that those drum overtones seem to be emphasized because some other sounds (maybe harmonics) which normally mask them are missing.

Damping the drum, or lowering those levels, will correct the imbalance to a degree, but the proper correction is to avoid creating the problem in the first place. I suggest that playing the recording through that tube amp added a lot of second harmonic distortion.

This distortion, by itself, is relatively innocuous, and it served to reproduce the masking effect that was originally present from whatever harmonic content and noise was omitted from the original recording. Because the imbalance is annoying and disconcerting, and the distortion is innocuous, the net result is an improvement. (If I'm right, then the Gainclone was *not* over-emphasizing anything, and would measure flat.) Also, if I'm right, then the best solution is to avoid the problem in the recording itself.

(I haven't done much recording live—very little, in fact—but I must comment that un-engineered live recordings, whatever their shortcomings, often do sound an awful lot like the original performance; often much more so than professionally engineered ones.)

This theory could be tested pretty easily: Take one of those recordings that you (and probably I) agree sounds better being played through that triode amplifier. Record the output of the triode amplifier digitally using a good A-to-D converter. Now play it back through the Gainclone. Does it sound the same or not?

Or you could connect the output of the triode amplifier (with the speakers for loads and through attenuators) to the input of the Gainclone and then see whether the improved sound is faithfully amplified by the Gainclone (without any unnatural emphasis). (If so, then the difference is strictly a matter of the triode adding something.) You could compare the outputs of both using a pair of headphones with attenuators to be entirely fair.

As for your suggestion about tube versus solid-state amplifiers and speakers (and dull and lifeless sound): I think you're correct there, at least to a degree, but it depends on the speakers. Most solid-state amplifiers do provide high damping factors (including the Gainclone).

However, most speakers designed to be connected to that type of amplifier are designed to expect precisely that high damping factor. Most tube amplifiers have rather low damping factors and tend to be sensitive to load inductance and capacitance.

Therefore, any speaker that is designed to run well on a low damping factor tube amp may well sound overdamped on a solid-state one and, likewise, any speaker designed to run well on a solid-state amplifier may sound muddy and underdamped on a tube one. (I am inclined, however, to suspect that the Gainclone has good transient and frequency response.)

You can easily test this by reducing the damping factor of the Gainclone by simply putting a resistor in series with each speaker. If they're 8Ω speakers, a 1Ω resistor will reduce the damping factor to approximately 8 and not change much else. (The damping factor is calculated as equal to the impedance of the speaker divided by the total source impedance.)

In this case, the source impedance is the sum of the output impedance of the amp, the resistance of the speaker wire, and any resistors you add in between. For this test, the resistor will dominate the others. For a few minutes at low levels a cheap few-watt resistor should survive (watch for smoke). Use metal film or carbon—avoid wirewound power resistors because most of them are very inductive (or get non-inductive ones). Of course, damping factor varies with frequency especially on tube equipment, but this should give you a pretty good idea if you're looking in the right direction.

Here's another interesting mystery with my theory of the explanation: I have a pair of speakers with large, heavy woofers. My friends and I agree that they sound like the bass goes lower when driven by certain amplifiers than by others even though the specifications of the amplifiers seem to suggest that they are equal. The amplifiers that sound better all have oversized power supplies and multiple paralleled output devices.

Being someone who does *not* believe in magic, I am sure that the difference can be measured, but it doesn't seem to be connected to the common measurements. My theory on this is that those massive woofers require a high damping factor to produce clean bass. When we normally measure damping factor, we do so statically by measuring the output impedance of the amplifier. I am not aware, however, of anyone measuring the overall dynamic output impedance under high-current conditions. Perhaps the big power supplies and paralleled output devices in the better amplifiers allow them to maintain a low output impedance while sinking large current spikes coming back from the speakers, and so maintain their damping factor under difficult loads, while the not-as-good amplifiers lose damping when presented with difficult loads.

One more mystery to consider: One of the mystical claims I often hear is that certain pieces of equipment do better at retrieving ambience while others clip off the tails of transients. They actually claim that, for example, the ambient tail on a bell sound will persist longer on some amplifiers than others. Clearly, the implication is that certain pieces of equipment actually fail to reproduce certain input signals, which makes no sense when said of a linear amplifier or preamp. (This could actually occur with some compression techniques where low-level content actually may be discarded under some conditions.)

Now, think about this psychoacoustically and assume that the sounds aren't missing, but that the listener sometimes doesn't hear or notice them. This provides all sorts of possible answers and explanations:

1. It could be that amplifiers with high noise floors appear to clip transients sooner because, as they decay, they

disappear more quickly below the noise floor. It could even be that the specific characteristics of the noise floor on some amplifiers make it more or less likely that the noise will mask the signal sooner or at higher levels.

2. It could be that small differences in amplifier power are to blame (in one case, a 6W per channel amp is claimed to clip transients more than a virtually identical, but 12W per channel sibling). Since you can play the more powerful amp a bit louder but probably not noticeably louder, it could be that you can hear the transient decays longer from it before they disappear below the ambient noise floor but fail to hear the difference in level.

One last mystery to invite comments: I don't see many reports from people who claim to prefer the sound of vinyl and who have carefully recorded their vinyl, using very high quality A-to-D converters and good bit rates, and compared the result to the original. (The few that I have seen suggest that the sound of vinyl is perfectly preserved on a digital recording.) If the vinyl sound stays, then it is simply a sum of the distortions and equalizing requirements of vinyl.

If that were true, then it implies that a vinylizer processor could be made that would make CDs sound like vinyl, among other interesting things. I sort of suspect that this is one of those questions which most concerned people would prefer *not* to see answered definitively.

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Keith Levkoff ("Personal Digital Recorders, Part 3") is a self-employed computer product analyst and technical writer who has been an audio hobbyist for longer than he cares to admit. He has spent most of his life working in the electronics and computer industries, and has held positions ranging from Electronic Assembler and Prototype Technician to Production Engineer and Marketing Engineer. Keith is interested in speaker design, all sorts of amplifiers, and, of course, digital audio. His other hobbies include digital photography and computers.


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