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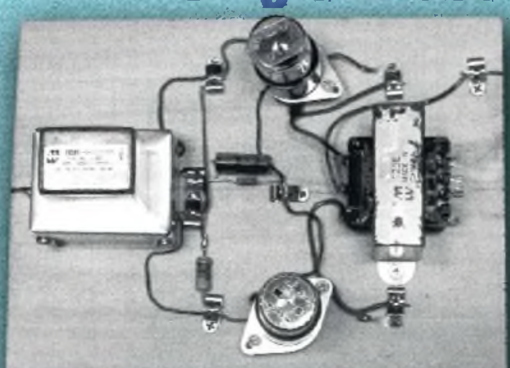

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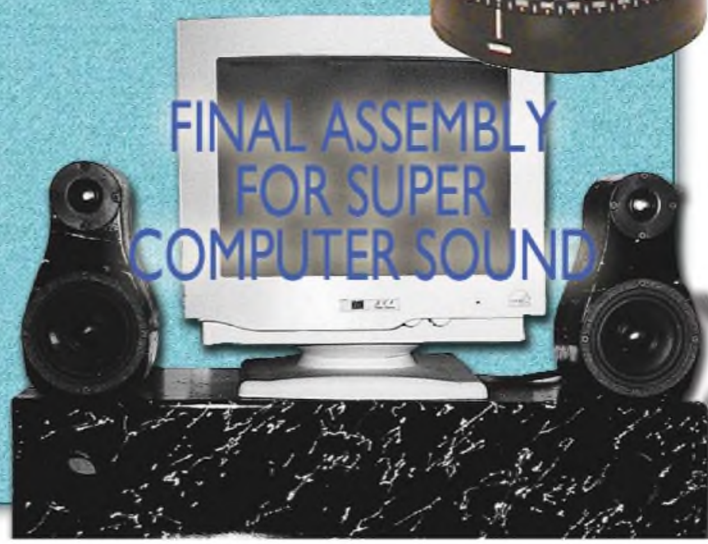

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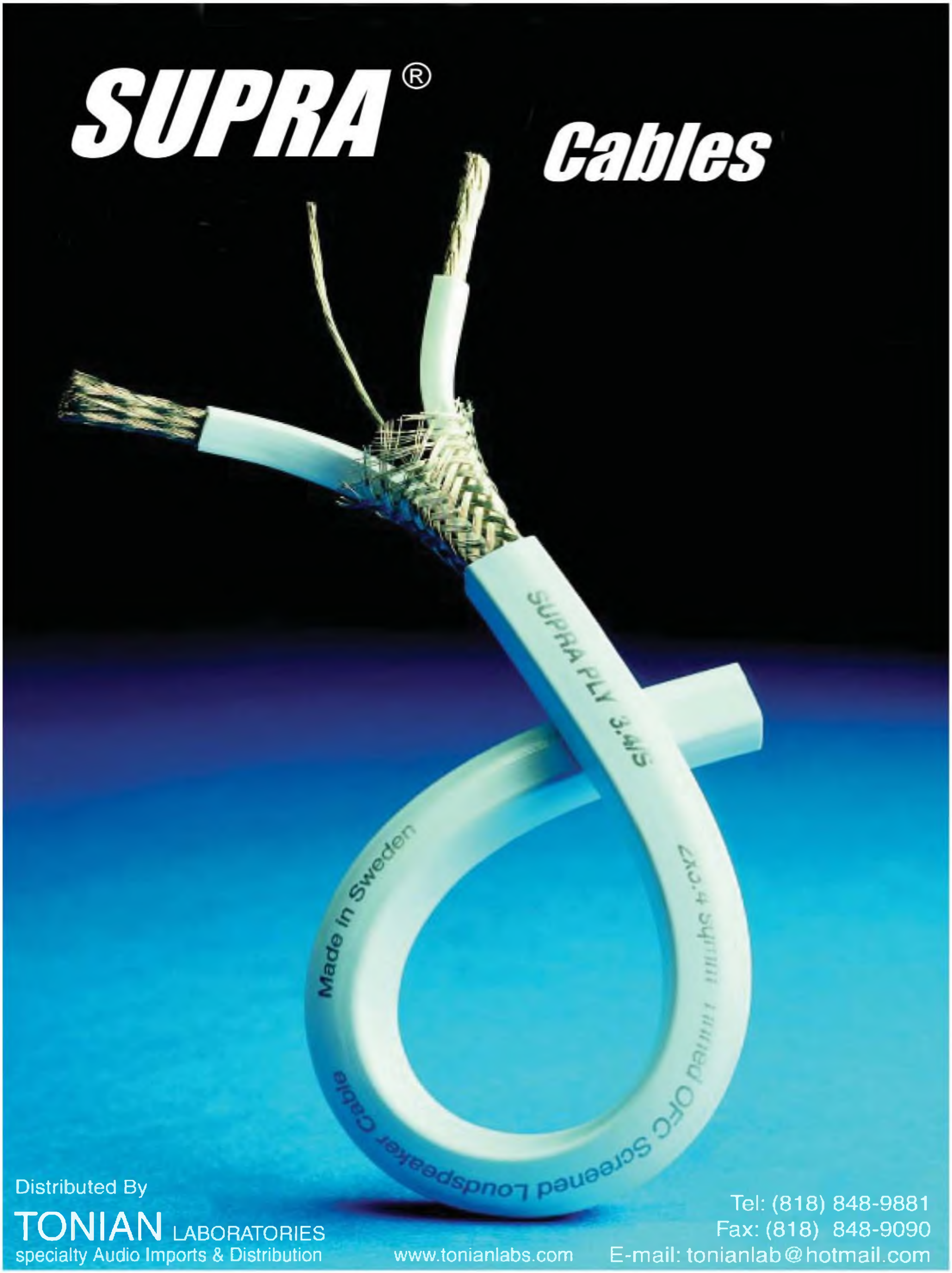
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ALMA TEST CD

ALMA International announces the release of its new Test Signal CD, which was developed to accompany the EIA-426-B standard. The CD contains the calibration and test signals for all the tests defined in the standard and outlines the procedures to make best use of the disk. The CD also has a number of test-signal tracks, including additional higher-frequency pure tones and shaped tone bursts that are not a part of the standard. For more, call 609-799-8440 or go to ALMA's website at www.ALMA.org.

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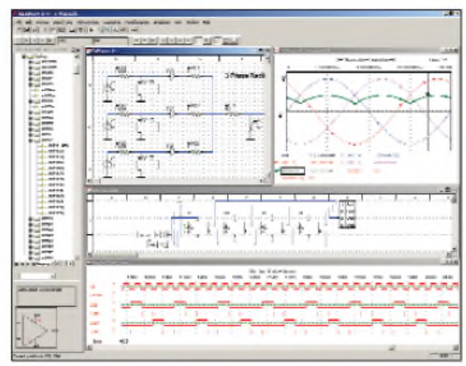
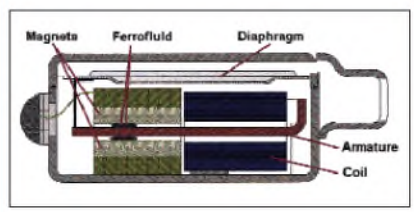


Beige Bag Software introduces B^2 Spice A/D Version 4, which includes the major features required to design and simulate analog, digital, and mixed mode circuits. B^2 Spice's simulation engine includes RF, Fourier, noise, distortion, Monte Carlo, and parameter sweeps. It also includes new features such as PCB export, improved schematic entry, and graphical post processor. B^2 Spice A/D Version 4 also provides support for over 9600 device models. Contact Beige Bag Software, Inc., 279 E. Liberty, Ann Arbor, MI 48104-2120, 734-332-0487, FAX 734-332-0392, Info@beigebag.com, http://www.beigebag.com.

PIANOCRAFT SERIES

Yamaha introduced two additions to its series of PianoCraft micro-component systems, the MCR-E250 and MCR E150. The MCR E250 system consists of the RX-E200 receiver, CDC-E250 three-CD changer, and NX-E200 speakers. The MCR-E150 consists of the CRX-E150 combination receiver/single disc CD player and NX-E150 speakers. PianoCraft systems are available in gold or silver tone metal finishes with piano-style black gloss speakers and piano finish accents on all electronics. For more, contact Yamaha Electronics Corporation, Multimedia Products Division, 6660 Orangethorpe Avenue, Buena Park, CA 90620, 714-522-9011.



SUB-MINIATURE SPEAKER TECHNOLOGY

Emkay Innovative Products announces an important technology advance for its sub-miniature speaker product line. The company—whose sub-miniature speakers are utilized in low-profile headsets and earphones for both communications and professional audio applications—has developed a new damping technique using ferrofluid. The company claims this is an excellent substitute for screen damping and provides the option for dramatic peak reduction. For more, contact Emkay Innovative Products, 1151 Maplewood Drive, Itasca, IL 60143, 630-250-5930, FAX 630-250-5932, website: www. emkayproducts.com.

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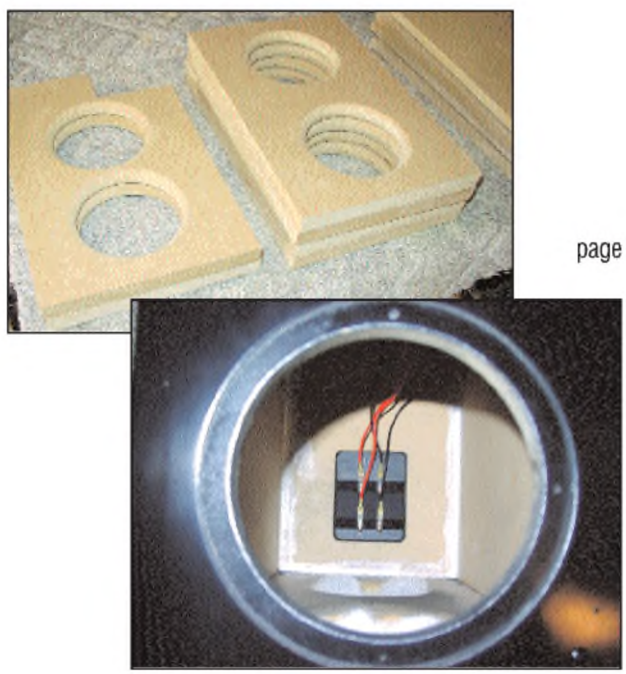

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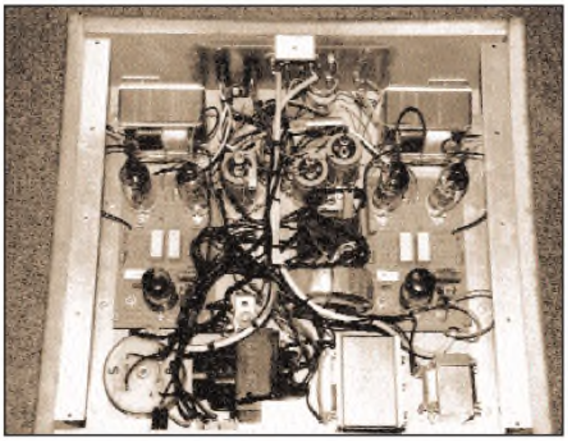

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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NEW FROM NAKAMICHI®

Nakamichi introduces six new mobile power amplifiers, four new mobile subwoofers, and eight new mobile speaker configurations. The PA-4001, PA-1004, PA-1002, PA-8001, and PA-2002 make up the amplifier series; the PA-8001 and PA-2002 feature Nakamichi's Harmonic Time Alignment™ amplifier topology. The SP-W104S, SP-W124S, SP-W1000D, and SP-W1200D make up the new line of mobile subwoofers. Also, there are six coaxial speaker configurations; two are oval models designed for rear deck or in-door applications, while four are round models for use as replacement in-door or in-dash speakers. Two separate speakers, the SP-5200 and SP-6500 round out the new speaker line, which all come with attractive grille covers and installation hardware. Contact Nakamichi at www.nakamichi.com.

KARAT M 30

Canton introduces the Karat M 30 DC three-way shelf speakers, based on the same developmental approach as the larger and more expensive Karat M 50 DC and M 70 DC, using a side-mounted woofer. The M 30 is finished in real beech, cherry, or black ash wood veneers, and is designed to mount on a shelf or pedestal. The 8″ side-mounted woofer provides low frequency reproduction with an overload tolerance comparable to small floor-standing systems, while the speaker presents a narrow face. The 6″ aluminum midrange cone offers clarity, detail, and power handling in the 180–3200Hz frequency range. These speakers are 6¾″ wide, 15⅛″ tall, and 15″ deep. For more, contact Canton Electronics Corp., 1723 Adams Street N.E., Minneapolis, MN 55413, 612-706-9250, FAX 612-706-9255, e-mail: info@cantonusa.com, http://www.cantonusa.com.

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Dan Shapiro, Executive Director, announces MusicLoversAuction.com, a free auction website designed specifically for musicians by musicians. It is comparable to e-Bay, but targets a more specific audience, specializing in buying and selling CDs/albums and musical instruments, and also in auctioning concert tickets, rehearsal space, studio time, bands and DJs for hire, music software, music videos, stereo equipment, group memorabilia, and lessons from pros. Dan Shapiro, Executive Director, MusicLoversAuction.com, One World Trade Center, 8th Floor, Long Beach, CA 90831, 562-983-8068, e-mail: Dan@musicloversauction.com, http://www.MusicLoversAuction.com.

COMPACT 1

D.A.S. Audio introduces the Compact 1 self-powered, three-way, biamplified loudspeaker system. It was designed as a full-range stand-alone unit without need for bass reinforcement, so it is useful in a wide range of portable and permanently installed sound reinforcement applications. The self-powered system incorporates a 1000W "Class D" switching amplifier, which utilizes two 500W ICEpower® amplifier modules manufactured by Bang & Olufsen PowerHouse. The Compact 1's trapezoidal enclosure is made of Wisa® birch plywood, and is protected against abrasion and the elements by multiple coats of catalyzed polyurethane paint. D.A.S. Audio, 1 Enterprise Drive, Old Lyme, CT 06371, 860-434-9190, FAX 860-434-1759.

The Infinite Box Concept, Part 1

In the first of two parts, this pair of audio experts explains how a little-known approach to speaker design works and how you can make use of this "infinite box" technique. **By G. R. Koonce and R. O. Wright, Jr.**

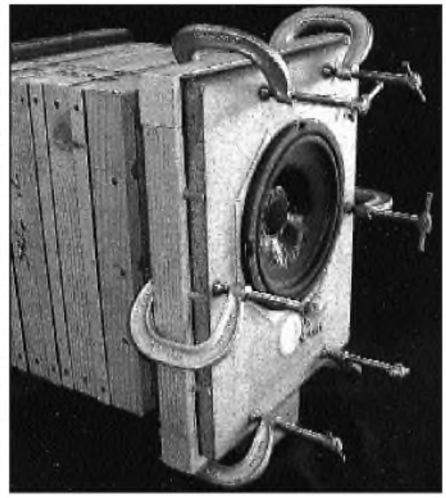

PHOTO 1: Front view of IB type H breadboard with Ref. #139C driver.

The technique described here is not new; however, it has not been well documented. It consists of putting your woofer in an open-back box and filling the rear of the box with dense damping material.

You may think there is no reason to read this article! You may believe this approach yields a low-total-Q closed box, and that the system response would be a second-order high-pass function set by the system total Q and resonance. Wrong! If you're thinking along these lines, you need to read on.

We initially thought the dense damping material did no more than absorb the rear radiation from the driver, so we adopted the name "Infinite Box" (IB). We spent time measuring the system resonance and total Q of the drivers in an IB believing this to define the system response. Initial acoustic testing quickly destroyed this concept.

It will take a bit of theory, but this article covers the history as we know it, how these speakers really work, and how you might design and implement one. It's an interesting story, so join us.

ABOUT THE AUTHORS

G.R. Koonce is an electrical engineer who has enjoyed the hobby of designing and building audio equipment, test gear, and speaker systems. Professionally, he worked for over 30 years producing audio frequency equipment for military use. He is a member of AES and has been a contributing writer to *Speaker Builder* since 1981.

Robert O. Wright, Jr. is a graduate of the University of Tennessee. He is a retired mechanical engineer from the Tennessee Valley Authority and is now the owner of a custom design-engineering firm in Knoxville, TN. It specializes in machine design, pollution control equipment design, sound design, and speaker design. He has been active in the field of audio and HiFi since the early 1950s and he and two others hold a patent on a special speaker-cooling device. He is also a member of the AES.

INFINITE BOX HISTORY

By R. O. Wright, Jr.

I learned about the Owens-Corning (OC) 700 Series Fiberglas® from a local speaker manufacturer's employee in the late 1970s. I was starting to build a set of three-way speakers for my own use and procured drivers and crossovers from him. He suggested I try the sound-damping material that, in his opinion, was superior to all others he had tried.

Taking his suggestion, I procured some of the OC #703 material and ran listening tests comparing it to loose fiberglass used inside the enclosure. The test boxes' sound was a definite but not a dramatic improvement that I heard almost at once. Voices seemed "cleaner" and bass "sharper," so I decided to use the new sound-damping material.

Before experimenting any further with the speakers, I went to a professional sound show in the Nashville, Tenn., area. Everyone talked about what they were doing, and I was no exception. I found out that the OC material was virtually unheard of and thought to be used by only a few of the pro-sound constructors. No one had any firsthand knowledge or data concerning its use.

Returning home, I contacted OC, which sent me a limited amount of basic engineering data on the material, mostly thermal. It was produced to be an insulating material and had somehow found its way into the audio world.

The next year while attending the CES Show in Chicago, I ran across a few people who had used OC 703. Their experiences were limited to a few applications rather than any research work. They spoke about how nice and "clean" speakers sounded when they used OC 703 and how it made the voices of the singers—especially the female voices—sound better.

With my three-way speaker project completed and the listening tests underway, I concurred with their opinion that voices sounded better and that the transient response was quicker than on similar speakers of the day. This was especially true of the high-end recordings with singers and good bass.

When the 1980s arrived, I began to find more data on the use and application of OC 703. At this time I was beginning to form some finite conclusions about the material and its place in the market.

I believe it's obscure because the material is used only in specialized areas of building construction and is not normally offered for sale on the open market. To my knowledge, it is offered only for sale by OC warehouses and distributors, rather than the giant discount warehouses. As a result, the material seems to be relegated to an obscure place in the world of audio construction. No one that I have run across has any idea how it found its way into the audio world.

Due to the sound parameters of OC 703, it lends itself to all types of interesting box designs. During the mid-1980s, I proceeded with the design of a subwoofer system using a pair of 15″ speakers employing the IB concept. Engineering work showed this was indeed possible and would produce a speaker box much smaller than the conventional vented box. The material had superior sound-absorbing qualities even down in the low-frequency range.

I mounted a pair of 15″ speakers in a box with an "over and under" configuration. The box was 48″ high, 28″ wide, and 24″ deep with an open back, hence an IB configuration. I stuffed the higher-density Owens-Corning #705 material into the open back of the box approximately 18″ deep. I chose to limit their audio range to less than 100Hz via electronic crossover and used a high wattage amplifier to power them. This system produced solid bass along with a tight transient response with excellent damping of the cone excursion.

It appears OC developed these high-density damping materials during or just after WWII for use in aircraft. G. R. Koonce (GRK) has drawings by Altec Lansing Corporation that show the use of a "PF-617" damping material inside speaker enclosures and dated in the early 1950s. This material was manufactured at the time by OC, and the best information indicates it had a density of 7–9 lb per cubic foot. Owens-Corning has since superseded it with #707 material, which is only made to order. Later, OC sold the machinery to manufacture PF-617 material to UPF Corp. in California.

Around the millennium's turn, I arranged for GRK to get some of the OC material for tests and use in speaker building. Numerous tests revealed a number of interesting conclusions. This is the basis for this article on the IB concept using OC 700 Series sound-damping material. The OC 700 Series material is also excellent for use in lining enclosure walls to reduce standing waves, which is a more conventional application.

I would like to thank Debbie McGill of Performance Contracting, Inc., of Knoxville, Tenn., for all her help in obtaining data on OC products. Through her efforts, I obtained the number (800-328-7617) for the Owens-Corning Customer Service Center. Call this number to locate the nearest OC distributor in your area.

While documentation on how an IB works seems lacking, many people have built them and reported their superior performance. See, for example, "A Quick Bookshelf Pair" by Lester Mertz (*audioXpress,* February 2001, p. 26). What is generally reported is the "taut" bass and "clean" sound that I had experienced. You also hear that the systems are rather insensitive to what driver you use. This work will give some answers to the questions raised in building this type of system.

WHAT IS AN INFINITE BOX?

Everyone knows that playing a bare woofer produces little bass, as the radiation from the front and rear of the cone cancels. The common cure is putting the woofer in a box to isolate the rear-cone radiation. Another approach, called infinite baffle, is to mount the woofer in a wall so the cone front and rear radiate into different rooms. The infinite baffle approach is effective with the right driver, but was more practical in the monaural days than today's surround-sound era.

The approach presented here started out as simply absorbing the cone's rear-radiation to simulate an infinite baffle. You mount the driver in an open-backed box that contains sound-absorbing material. In practice, this produces something totally different.

First, the damping-material-packed

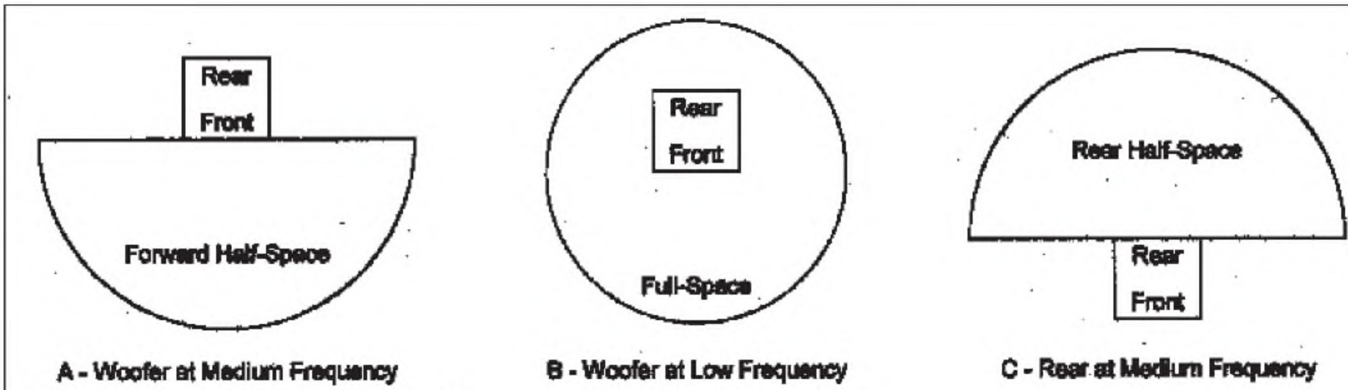

FIGURE 1: Simplified radiation patterns of enclosure.

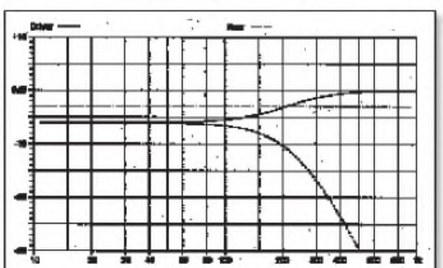

FIGURE 2: 6dB DSL and RDSL for a 13.5″ wide enclosure.

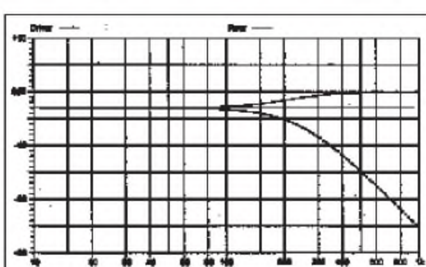

FIGURE 3: 3dB DSL and RDSL for a 13.5″ wide enclosure.

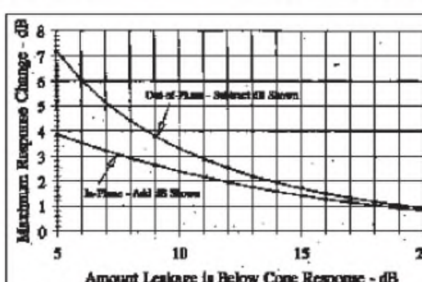

FIGURE 4: System response change versus leakage level.

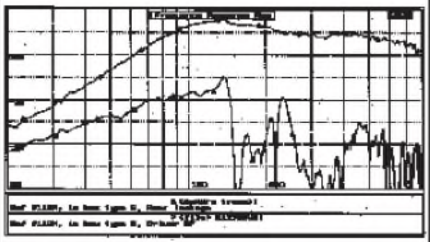

FIGURE 5: Typical cone and rear-leakage responses.

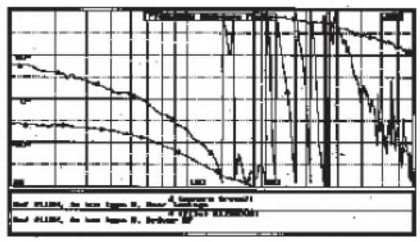

FIGURE 6: Typical phase shifts of responses.

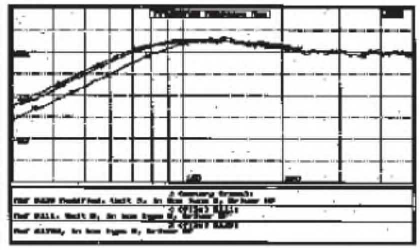

FIGURE 7: Cone responses for three drivers on same IB.

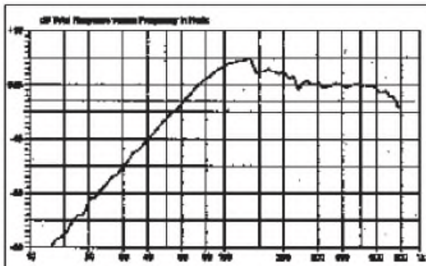

FIGURE 8: System response of IB type B with Ref. #119X driver—no DSL.

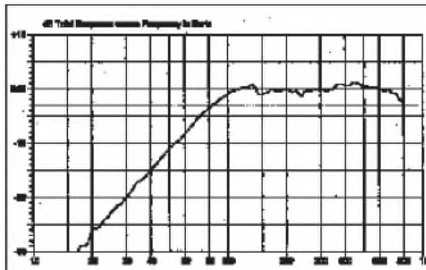

FIGURE 9: System response of IB type B with Ref. #119X driver—6dB DSL.

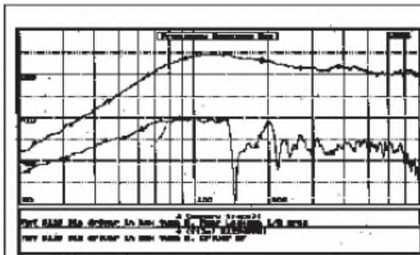

FIGURE 10: Test responses for IB type H with Ref. #119X driver.

box modifies the driver parameters just like other box types. Second, the damping materials lack low-frequency attenuation yielding box "rear leakage" that interacts with the cone's response. We will show that phase shift through the damping material causes the rear leakage to augment the system bass response. With proper design the approach produces a well-damped system with predictable response and acceptable low-frequency cutoff (f_3).

As mentioned previously, the IB is not a new concept. The approach is "said" to be relatively insensitive to driver selection, but we thought there must be a more scientific way to handle IB design. This article reports on learning how to use the IB technique. It shows you how to design an IB and what thickness of damping material is needed along with why you might choose to try this approach. However, we must start with some theory that is important to the IB story.

DIFFRACTION SPREADING LOSS

When you suspend an enclosure well away from all reflecting surfaces, it will display diffraction spreading loss (DSL). Assuming all front-panel-mounted drivers and at medium frequency, the enclosure will radiate only into the forward half-space defined by the front panel (*Fig. 1A*). With falling frequency the front-panel dimensions can't restrict the radiation to the forward half-space.

Below some frequency the enclosure will become omnidirectional, radiating into full-space (*Fig. 1B*). When that occurs, the forward on-axis response will dip 6dB. The way to maintain a flat on-axis response is to use DSL compensation in the system's design.

While bass response is subject to personal preference and very room dependent, we believe with floor-standing boxes using a low mounted woofer, DSL compensation of 6dB is excessive. We believe 3–4dB DSL compensation gives better bass balance for such enclosures. For boxes on stands, and used well away from all walls, 6dB DSL compensation might be appropriate. A tall floor-standing tower with a high-mounted woofer might also benefit from 6dB DSL compensation.

IB TESTING PROBLEMS

One of the problems in testing an IB is the DSL. The cone response at the IB's front will see DSL, but what happens to the IB's rear leakage is more interesting. We are listening to this rear leakage from in front of the box. At low frequency the rear leakage will have the same omnidirectional pattern as the cone response.

However, as frequency increases, the rear leakage will be restricted to radiating into the rear half-space (*Fig. 1C*), and the level heard out in front will decrease. We refer to this effect on the rear leakage as reverse diffraction spreading loss (RDSL), which will help "mask" the rear leakage at higher frequencies.

Figure 2 shows plots representing 6dB DSL and RDSL for a 13.5″ box width, and one wavelength wide at about 1kHz. At low frequency both responses are down 6dB on-axis in front of the box. Rising frequency results in the cone response (DSL curve) rising to 0dB as the rear response (RDSL curve) continues to drop. This shows what you hear out in front of the box with no nearby reflecting boundaries.

You can see why mounting the port of a vented box (VB) on the enclosure's rear has a major sonic advantage, as high-frequency port leakage is suppressed by the RDSL. *Figure 3* plots 3dB DSL and RDSL as used for modeling purposes. These curves do not fit any theory; they are simply half the effect the theoretical 6dB DSL produces.

Another problem in determining the IB's response is that the two acoustic outputs are not in the same plane. The cone is on the front panel while the rear leakage occurs at the back of the box. This same problem exists when working with a VB with a rear-mounted port.

Clearly a large anechoic chamber would allow measurement of the IB sys-

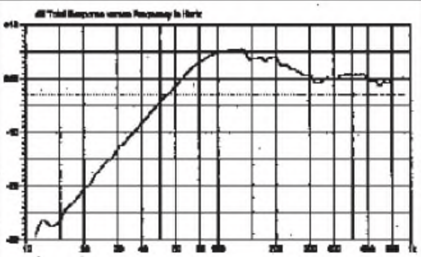

FIGURE 11: System response of IB type H with Ref. #119X driver—no DSL.

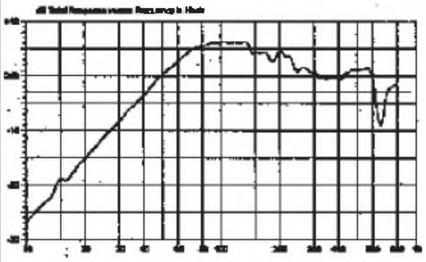

FIGURE 13: System response of IB type H with Ref. #139C driver—no DSL.

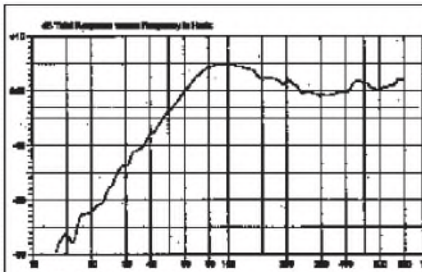

FIGURE 15: System response of IB type J with Ref. #82A driver—no DSL.

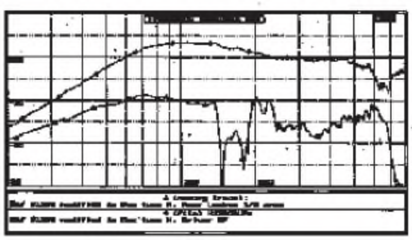

FIGURE 12: Test responses for IB type H with Ref. #139C driver.

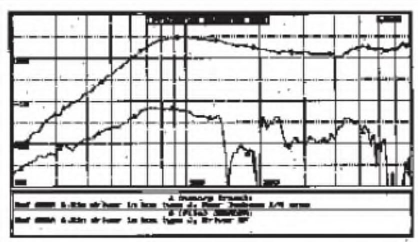

FIGURE 14: Test responses for IB type J with Ref. #82A driver.

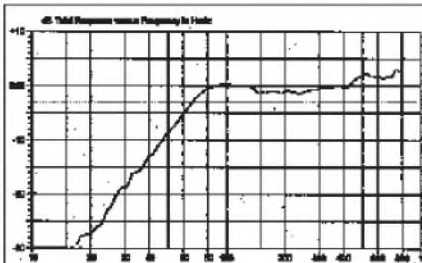

FIGURE 16: System response of IB type J with Ref. #82A driver—6dB DSL.

tem response, including all these effects. But lacking an anechoic chamber, we needed another approach. Burying the box in the back yard is not an option with the IB, and with any box type this approach removes DSL effects. Ground-plane testing also has problems, because when you lay the box against the ground plane it changes the effective box dimensions and thus both the DSL and RDSL effects.

For the homebuilder this leaves near-field testing. We used this along with special software to properly sum the near-field results for the cone and rear-leakage responses. Note that DSL and RDSL do not appear in near-field testing and were handled via the software.

You measure both the cone and rear-leakage responses via the near-field technique. The rear-leakage measurement includes any attenuation and phase shift due to transit from the cone's rear through the box and damping material. To properly sum these two responses for the out-in-front on-axis response, you must do the following at each frequency point:

1) Apply the desired DSL correction to the cone response based on the width of the box.
2) Apply the desired RDSL correction to the rear-leakage response based on the width of the box.
3) Correct the rear-leakage response for the in-air transit time from the back to the front of the box based on the box depth.
4) Correct the rear-leakage response for any radiating area difference between the cone and the rear of the box.
5) Vector-sum the two corrected responses.

This is the approach we used. The program developed allowed selection of summing with no DSL/RDSL correction or with 3dB or 6dB of DSL/RDSL correction. No DSL/RDSL correction allows comparison with other box techniques, because they are normally displayed that way. Generally, design software and most test response plots ignore DSL and are only correct if the enclosure is right against the rear wall.

The results with DSL/RDSL correction may give a more accurate view of the system's true on-axis response. The effects of DSL/RDSL permit you to modify the "bass sound" of speaker systems by varying their placement relative to nearby walls. As most readers have learned, clarity and imaging may suffer with placements that provide maximum bass.

ADVANTAGES OF THE IB

The main advantage of the IB is its tight, fast bass due to the superior damping it possesses compared to other box types. The typical hi-fi woofer is about 1% efficient, leaving the cone well isolated from the electrical side. You can accomplish the application of sufficient electrical-side damping by using an extremely large magnet, which improves the efficiency, but this may result in a driver not producing a reasonable system design. Applying the damping on the acoustic side can avoid this problem, permitting the use of a standard driver with a more reasonable cost.

As discussed, any woofer system not built into—or sitting right against—the rear wall will exhibit DSL. Approaches such as dual-woofers or a dual-voice-coil woofer exist to compensate for DSL. These can do the job but add complexity by requiring additional crossover components. The dual-woofer approach does have the advantage of increased bass power capability.

All you need for DSL compensation is a peak in the bass response and a properly sized box. Most box types produce poor sound quality when built with a response peak because of a high system Q. With its superior damping, the IB does not have this defect, and you can build good-sounding IB systems with reasonable bass peaking for DSL compensation, which is another advantage.

IB BREADBOARD

The important things to learn for the IB are how much box volume and how much damping material to use. Having no hope of a theoretical solution, we used empirical development by testing. We developed a breadboard that allowed varying the box volume ahead of the damping material, called the dead-air (DA) volume, and the amount of damping material used (*Photos 1* and *2*). It consists of a front section holding a test baffle, that mounts the driver via C-clamps with threaded rods out the back.

We built a group of frames (*Photo 3*)

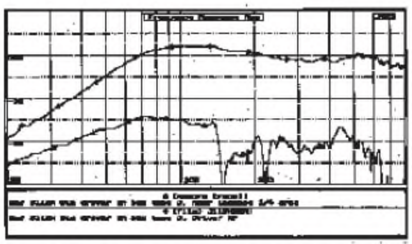

FIGURE 17: Test responses for IB type J with Ref. #119X driver.

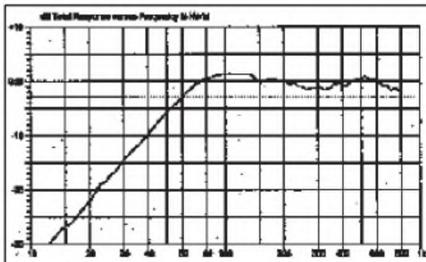

FIGURE 19: System response of IB type J with Ref. #119X driver—3dB DSL.

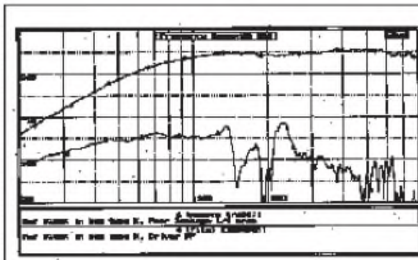

FIGURE 21: Test responses for IB type K with Ref. #109K driver.

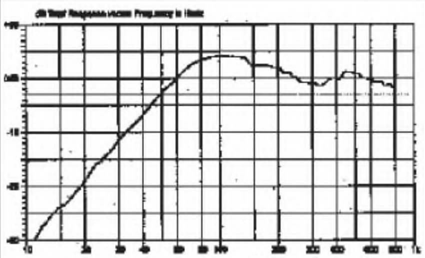

FIGURE 18: System response of IB type J with Ref. #119X driver—no DSL.

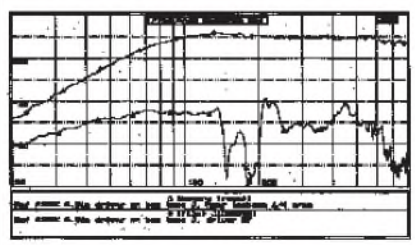

FIGURE 20: Test responses for IB type J with Ref. #109K driver.

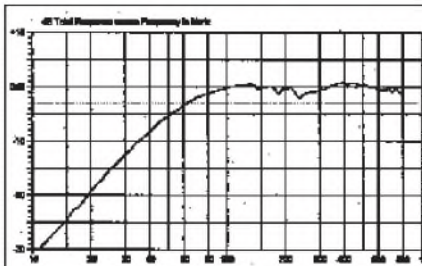

FIGURE 22: System response of IB type K with Ref. #109K driver—no DSL.

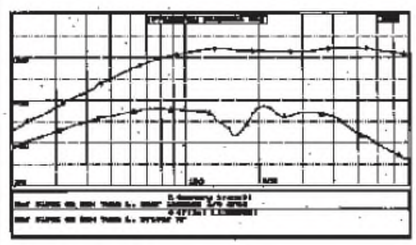

FIGURE 23: Test responses for IB type L with Ref. #109K driver.

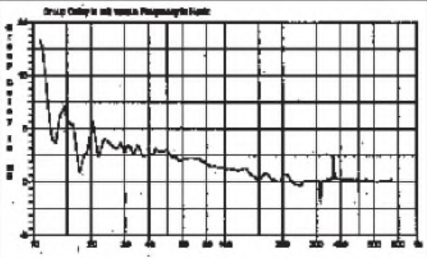

FIGURE 25: Group delay of IB type K with Ref. #109K driver.

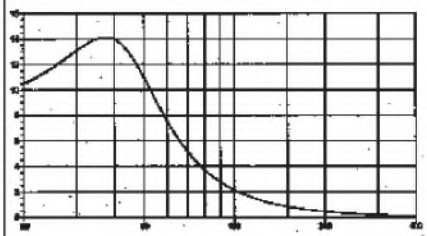

FIGURE 27: Group delay of vented box system with Ref. #109K driver.

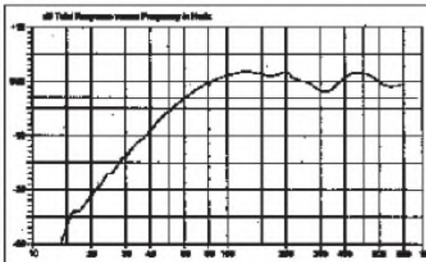

FIGURE 24: System response of IB type L with Ref. #109K driver—no DSL.

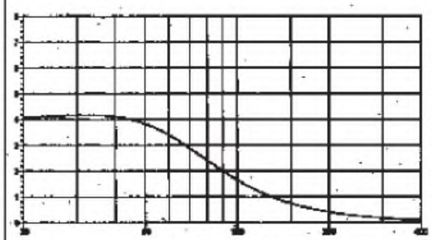

FIGURE 26: Group delay of closed box system with Ref. #109K driver.

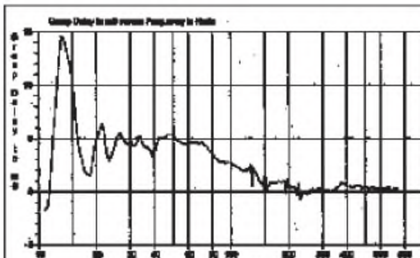

FIGURE 28: Group delay of IB type K with Ref. #119X driver.

that slide onto the rods to allow assembling the desired box. There are open 1 × 2 (0.8″ thick) and 2 × 2 (1.54″ thick) frames used as spacers. There are 2 × 2 frames filled with two 1″-thick layers of Owens-Corning #705 damping material, which has a density of about 6 lb per cubic foot. We secured the damping material to the frame with silicon rubber so it could not move—and with no air space allowed between the two-1″-thick layers. The total damping material is thicker than the frame, requiring a spacer behind each damping frame.

Each of the frames has a foam-tape gasket on the front for sealing. The desired frame arrangement is assembled onto the threaded rods and finally a "back board" spacer is added. We put stiff springs on the rods and tightened the nuts until the springs compressed to a set length. Tightening all rods without the springs had proved impossible with all the foam-tape gaskets.

The assembled breadboard had an internal height of 11.25″ and an internal width of 8″ for an area of 90 in^2. One IB question is, do you want to leave the entire rear of the box open? To establish this, we constructed two reducer plates (*Photos 2* and *3*) that bolt to the "back board" and reduce the rear open-area to half or one-quarter of the 90 in^2 area.

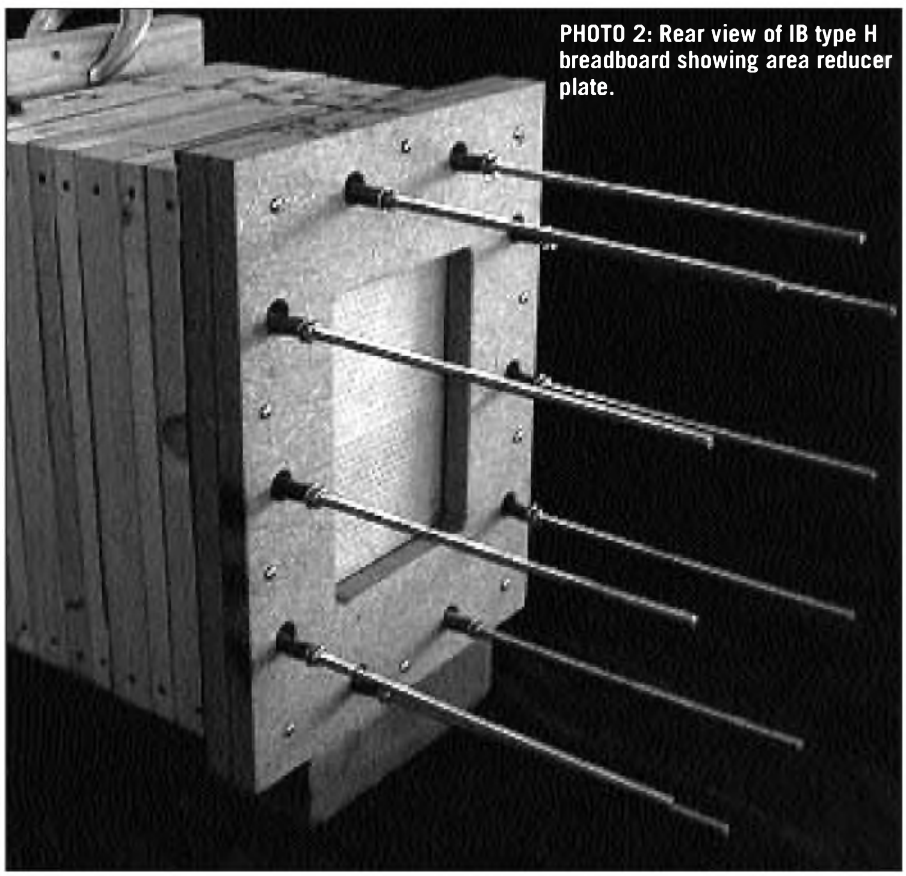

PHOTO 2: Rear view of IB type H breadboard showing area reducer plate.

The minimum DA volume of the front section followed immediately by a damping layer was 480 in^3 (0.278ft^3). We could reduce this DA volume by placing bricks inside the box. The maximum DA volume depended on the number of spacers available and, to keep the possibility of two damping frames available, yielded an upper limit of 1,182 in^3 (about 0.684ft^3).

Note throughout this work that the DA volume is not corrected for any loss due to the driver. The driver is front-mounted onto the baffle, and calculations showed that the volume added by the baffle hole almost matches the driv-

PHOTO 3: Various frames used to assemble the IB breadboard. Top left is the one-quarter area reducer plate; top right is a 2″ damping layer, bottom left is a 1 × 1 spacer; and bottom right is a 2 × 2 spacer.

er's volume. *Table 1* shows parameters for the various breadboard configurations used with each type identified by a letter.

HOW MUCH LEAKAGE SUPPRESSION?

How much suppression of the rear leakage is needed in an IB? This is a trick question! The answer depends on how much you are willing to let the rear leakage modify the system response. *Figure 4* shows plots of the maximum influence the rear leakage can have on the system response versus the amount the leakage is suppressed.

The "in-phase" curves show how much peaking could be added by the leakage being in-phase with the cone response. The "out-of-phase" curve shows how much the system response could be reduced by the leakage being out of phase with the cone response. Typical of things done in dB, the two effects are not equal.

For example, if you decided to allow the rear leakage to add a maximum of about 1dB to the system response, it means you need to suppress the rear leakage about 18dB. This would be over the frequency range from about system f_3 up through the woofer passband. This suppression would limit any rear-leakage-caused dip to about 1.2dB. These values are without any DSL/RDSL effects. The RDSL on the rear leakage will help limit its effect at high frequency.

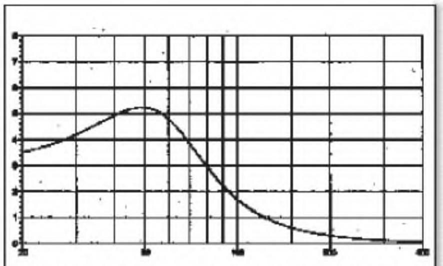

FIGURE 29: Group delay of closed box system with Ref. #119X driver.

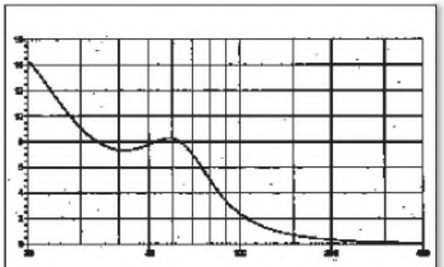

FIGURE 30: Group delay of vented box system with Ref. #119X driver.

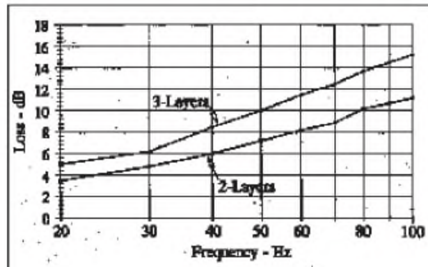

FIGURE 31: Loss through two and three layers of damping material.

Testing showed that the 18dB suppression used in this example exceeds that required for a successful IB system.

INITIAL TESTING

When you mount a driver in the IB, it changes its measured parameters. The driver parameters f_S, Q_{ES}, Q_{MS}, and Q_{TS} are transformed into the system parameters f_X, Q_{EX}, Q_{MX}, and Q_{TX}. The "x" subscript avoids confusion with parameters for a closed-box (CB) system using the standard "c" subscript. With regard to an individual IB, the final subscript is the letter of the IB type identified in *Table 1*.

The driver data was taken via Liberty Instruments' Audiosuite with the driver front-mounted into a baffle, which was then clamped to the IB and the system parameters measured. The system parameters should be considered as approximate.

Audiosuite computes T/S parameters by fitting a curve to the measured impedance. For a baffle-mounted driver or in a lossless CB, this fit is very good. However, Audiosuite was never designed to handle the IB, so the curve fit is not as good.

We selected nine driver types for this work (*Table 2*) to see how a variety of drivers responded on the IB. Later we reduced the driver list based on initial results. To have a low f_S driver, we modified the Ref. #139 driver by the addition of 8g to the moving mass–about a 25% increase. The data in *Table 2*, and used throughout, is for the modified driver.

The system response of an IB is the basic shape set by the cone response modified by the rear-leakage response. While later shown to be false, testing started with the concept that the cone response was determined by the system resonance and total Q. *Table 3* lists the system resonance and Qs for various IB breadboard/drivers combinations.

Study of the results helps in understanding the IB system. *Table 4* shows how the driver (*Table 2*) and system data (*Table 3*) compare. A final "s" subscript refers to baffle data, while other final subscripts refer to the IB type.

System resonance is greatly affected by the DA volume, but is not too sensitive to the number of damping layers. Later acoustic testing verified this point, and DA volume turns out to be the most important design parameter of the IB, as with other box types.

Compared to system resonance, electrical Q is a less sensitive function of DA volume. It also varies with the number of damping layers, rising with more layers. System electrical Q ranges from 0.76 up to 1.81 for the driver/IB combinations in *Table 3*.

The mechanical Q is not easy to catalog. It tends to increase with increasing DA volume and with more damping layers. System mechanical Q ranges from

TABLE 1
PARAMETERS FOR VARIOUS BREADBOARD IB CONFIGURATIONS

TYPE	DEAD-AIR[1]	LAYERS[2]	GAP[3]	REAR AREA[4]	DEPTH[5]	NOTES
A		3	1	Full	14.6	Note 6
B	480	3	1	Full	14.6	
C	480	3	1	½	15.4	
D	480	3	1	¼	15.4	
E	480	2	1	Full	11.4	
F	480	2	1	½	12.2	
G	480	2	1	¼	12.2	
H	824	2	⁄8	½	15.6	
I	824	2	⁄8	½	15.6	Note 7
J	1,182	2	⁄8	¼	19.1	
K	1,182	2	⁄8	¼	19.1	Note 8
L	1,182	1		¼	19.1	Note 8

Notes:
1—Dead-air volume in cubic inches.
2—Number of damping layers. Each layer is 2″-thick #705 material.
3—Approximate air gap between the damping layers in inches.
4—The amount of 90 in^2 box area open at rear.
5—Total box depth from front panel to rear of IB in inches.
6—For 6½″ and 8″ driver the dead-air volume was 345 in^3. For the 6 × 9″ driver it was 412.5 in^3. Set with bricks.
7—This IB type had nominal 3″ fiberglass layer at back of dead-air volume against front damping layer.
8—This IB type has the walls of the dead-air volume lined with nominal 1½″ fiberglass.

TABLE 2
DRIVERS USED IN TESTING

DRIVER	f_S	Q_{ES}	Q_{MS}	Q_{TS}	V_{AS}[1]	NOTES
Ref. #81B	75.8	1.07	3.93	0.84	615	6½″
Ref. #82A	60.8	1.38	4.53	1.07	850	6½″
Ref. #109K	45.6	0.57	2.28	0.46	1,514	6½″
Ref. #111D	50.1	1.06	3.63	0.81	726	6½″
Ref. #102A	43.3	0.32	2.94	0.29	3,024	8″
Ref. #119X	30.1	0.59	2.37	0.45	4,804	8″
Ref. #139C	27.6	0.59	2.96	0.48	3,992	Modified 8″
Ref. #141A	61.1	1.51	7.89	1.27	3,145	8″, whizzer
6x9B	75.8	1.03	4.22	0.84	1,192	Whizzer

Notes: 1—V_{AS} shown in cubic inches.

1.20 to 2.73 for the shown driver/IB combinations. In all cases the system mechanical Q is lower than that of the driver because of the loss introduced by the damping material.

For high-Q_{TS} drivers the IB system total Q falls in a range not far from Q_{TS} and may be below Q_{TS}. For lower-Q_{TS} drivers the system total Q is higher than Q_{TS} by design to get an acceptable response. The system total Q varies little with DA volume, but is sensitive to the number of damping layers. Over a range of DA volumes and damping-layer numbers, the total system Q stays

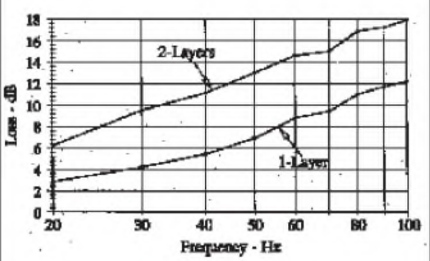

FIGURE 32: Loss through one and two layers of damping material.

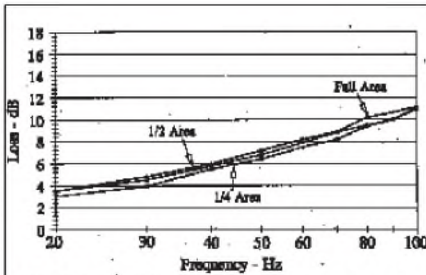

FIGURE 34: Effect of open rear area on loss through damping material.

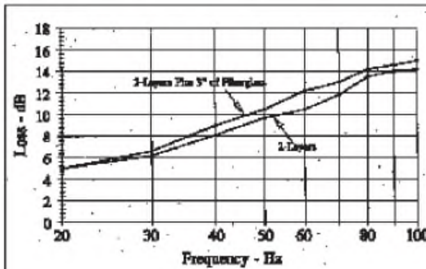

FIGURE 36: Effect of 3″ fiberglass layer on loss through damping material.

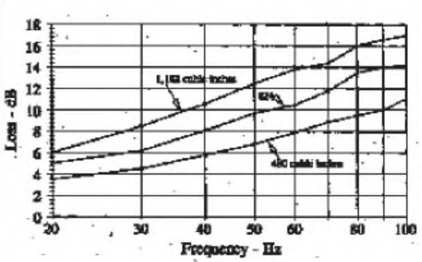

FIGURE 33: Effect of dead-air volume on loss through damping material.

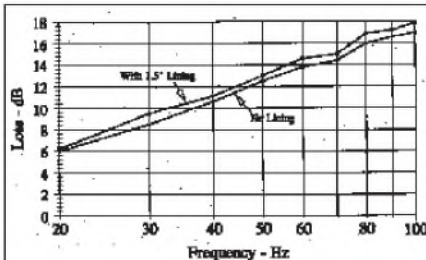

FIGURE 35: Effect of fiberglass lining on loss through damping material.

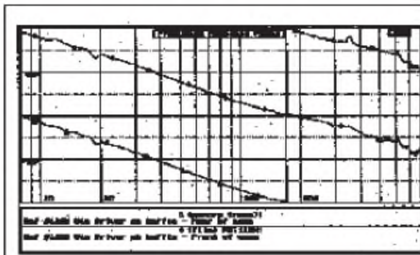
FIGURE 37: Phase difference between front and rear of driver cone.

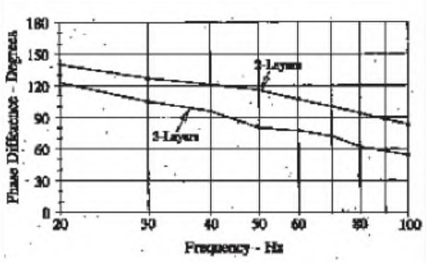

FIGURE 38: Phase difference through two and three layers of damping material.

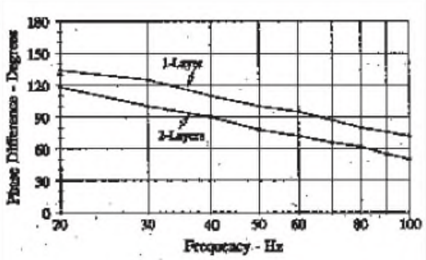

FIGURE 39: Phase difference through one and two layers of damping material.

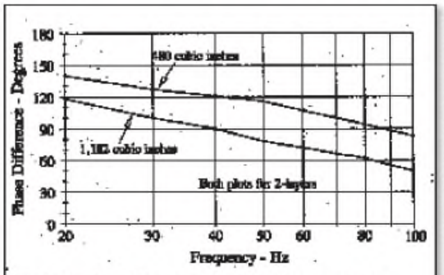

FIGURE 40: Effect of dead-air volume on phase difference through damping material.

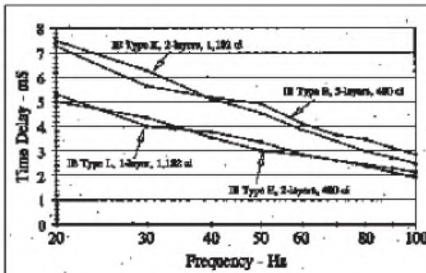

FIGURE 41: Time delay through damping layers versus frequency.

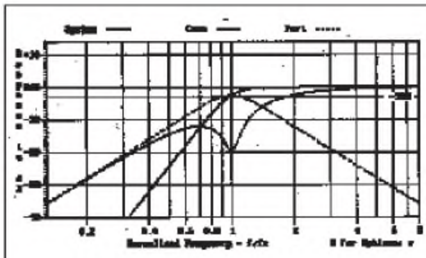

FIGURE 42: Cone, port, and system responses for B4 vented box.

reasonably low. A low total Q with a closed box would mean little peaking in the response; acoustic testing showed this is not true with the IB.

BASIC ACOUSTIC RESPONSE SHAPES

We started testing with low DA volumes in systems that produced rather high f_3 values. Once a valid design approach was developed, we could select what DA volume to use, and later we will concentrate on that testing. *Figure 5* shows the measured cone and rear-leakage response magnitudes for the Ref. #119X 8″ driver on IB type B. Ignore the wiggles in all test curves at low frequency due to noise interfering with the testing. For these typical results, with a low DA volume of 480 in^3, note the following:

1. The cone response resembles that of a CB, not that of a VB. It shows no dip as occurs with the VB near the box-tuned frequency. The low-frequency slope of this response is about 12dB/octave, just as it is for a CB.
2. The rear leakage shows a slope of about 6dB/octave up to about 100Hz because the loss through the damping material increases with frequency at about 6dB/octave. The loss through the damping material is indicated by the difference between the cone and rear-leakage plots. This loss varies with numerous parameters, as developed later.
3. From about 150Hz up, the rear leakage shows major dips and other roughness. We have not identified the cause of the two major dips in the range 150Hz–220Hz. These dips are not as bad with only one damping layer (2″ total thickness). The dip frequencies do not match any DA volume resonance mode we could develop and may be caused by some damping-material property.

Figure 6 shows the measured cone and rear-leakage response phase shifts for the same driver/IB combination. The driving function for the rear leakage is the back of the cone, thus it is always out of phase with the cone response. The cone and rear-leakage responses do not stay 180° apart with rising frequency.

A major phase shift is caused by the time delay in traveling from the front to the rear of the box, thus going through the damping material. By 100Hz the rear leakage is getting close to being in-phase with the cone and will boost the system response. By about 150Hz the two outputs are in phase, meaning a phase shift of 180° in traveling through the box.

A phase shift of 180° at 150Hz matches the time delay caused by traveling 45.1″ in air. This is a greater distance than the 14.6″ box depth; clearly the damping material produces the major time delay. Traveling 45.1″ in air takes about 3.3ms, while 14.6″ in air takes about 1.08ms, so about 2.26ms delay is due to the damping layers at 150Hz. Above 150Hz the phase shifts become rather complex so the rear leakage will tend to add ripple to the system response, as it sometimes adds and other times subtracts from the cone response.

IB TOTAL ACOUSTIC RESPONSE

Figure 7 shows cone responses for three drivers on IB type B with its 480 in^3 DA volume. Even though these systems have a low total Q—from 0.70 to 0.92—the cone responses show considerable and nearly identical peaking. The frequency scale changes, but the basic shapes are the same.

The IB has a reputation for being relatively independent of driver choice, and *Fig. 7* indicates this may be true. A low DA volume produces a system with a high f_3, which is predicted by the system resonance values that range from 61.5–80.9Hz for these drivers in this IB.

Figures 5 and *6* showed the magnitude and phase-shift responses for the Ref. #119 8″ driver in IB type B. Developing the system response for this combination involves summing the cone and the rear-leakage responses, taking into account the numerous parameters developed earlier. *Figure 8* shows the system response with no DSL correction, showing a healthy peak of some 5dB and a bit of ripple even though some smoothing has been applied to the plot. Note f_3 is about 62Hz, which is a bit high for an 8″ driver system.

Figure 9 shows the same system response with 6dB DSL correction applied to the cone response and 6dB of RDSL correction applied to the rear leakage before summation, based on the breadboard's 13″ front-panel (FP) width. For the remainder of this work we will refer to "DSL correction," but keep in mind that this means DSL correction on the cone response and matching RDSL correction on the rear-leakage response. The system response shows lower ripple and is an amazingly flat on-axis response for a system that would sit on stands away from all reflecting surfaces. The system f_3 has risen to about 83Hz. This is typical; the on-axis response of the system with DSL correction included will show an f_3 higher than the system response without DSL.

Remember, DSL is not something that you can elect to have or not have. It is a fact of life for systems that are not built-into or backed-up-to the rear wall. The DSL is the reason a speaker system placed out from the walls does not "sound" flat down to the f_3 value predicted by design software and shown in testing done by near-field measurement.

The plots without DSL are shown for comparison with other box types because that is how most boxes are designed. Clearly, the IB can provide DSL compensation while still maintaining a low total system Q. The ripples in the system response are nothing compared to what your room will do to the bass response!

TABLE 3
T/S PARAMETERS IN VARIOUS IB TYPES

DRIVER	f_X	Q_{EX}	Q_{MX}	Q_{TX}	IB TYPE	DA—in³	LAYERS
Ref. #109K	59.2	0.82	2.06	0.59	K	1,182	2
	59.3	0.76	1.61	0.51	L	1,182	1
Ref. #111D	71.1	1.32	1.81	0.77	A	345	3
	69.5	1.44	2.22	0.87	B	480	3
	69.2	1.24	1.78	0.73	E	480	2
	60.3	1.31	2.73	0.89	K	1,182	2
	60.4	1.23	2.14	0.78	L	1,182	1
Ref. #119X	89.3	1.74	1.53	0.82	A	345	3
	80.9	1.81	1.92	0.92	B	480	3
	81.6	1.61	1.56	0.79	E	480	2
	58.3	1.40	2.13	0.85	K	1,182	2
	59.8	1.29	1.50	0.69	L	1,182	1
Ref. #139C	61.5	1.27	1.56	0.70	B	480	3
	60.1	1.16	1.20	0.59	E	480	2
	47.8	1.09	2.07	0.71	K	1,182	2
	46.1	0.99	1.32	0.57	L	1,182	1
6x9B	95.3	1.35	2.55	0.88	L	1,182	1

PREDICTING THE IB RESPONSE

The preceding work has shown that the IB system response is not the same as that of a CB system with matching system resonance and total Q. Several attempts to develop an IB-system-design approach also failed. Finally, we developed an approach using a CB prototype that passed testing. This approach will give a prediction of f_3 for the IB system and the proper DA volume to use.

You design a lossless CB prototype for

the driver with a Q_{TC} at a desired value to define a box volume, V_B. You set the DA volume to 0.74 times V_B. The IB-system f_3, without DSL correction, can be predicted from the CB prototype f_3 value.

For CB-prototype designs with Q_{TC} near one, you will get a peaked response with f_3 about the same as the CB-prototype value. Designs based on a CB prototype with Q_{TC} below unity give a flatter system response with f_3 below the CB-prototype f_3 value. We have no rule to define how much f_3 will be below the CB-prototype design value, but it does not move very quickly with the Q_{TC} value used.

Table 5 shows how this works for drivers in a lossless CB-prototype design with Q_{TC} near unity. We added the Ref. #82A driver to see how a high-Q_{TS} driver behaves on the IB and requires a high design Q_{TC}. The CB-prototype responses would show peaks of 1.25–2.7dB. The box volume (V_B) for the CB-prototype design is multiplied by 0.74 to give the design DA volume.

Testing these results required configuring the IB breadboard with the specified DA volume and testing the system. The breadboard can implement only certain DA volumes. We modified the breadboard to a DA volume of 824 in^3 (IB type H), which is close to the proper value for testing the Ref. #119X and Ref. #139C 8″ drivers.

The measured cone and rear-leakage responses (*Fig. 10*) and the total system response without DSL (*Fig. 11*) are shown for the Ref. #119X driver. *Figures 12* and *13* show the same plots for the Ref. #139C driver. *Table 6* compares the f_3 from these system responses with that predicted by the CB prototype. Agreement is good, keeping in mind the system f_3 value is being read from a plot.

Now, the two 6.5″ drivers wanted dead-air volumes of 1,047 and 1,758 in^3. The maximum the breadboard can do and still have two damping layers is 1,182 in^3. We decided to test all four drivers at this DA volume, which is equivalent to a CB-prototype V_B of 1,596 in^3. *Table 7* shows the results for CB-prototype designs in this V_B. This IB type is J, which has two-2″ layers of damping material.

Only the plots for two drivers are shown. *Figure 14* shows the test results for the Ref. #82A driver, and *Figs. 15* and *16* show the system response without and with 6dB of DSL correction, respectively. Even though this driver has a very high Q_{TS} (1.07), it shows about the same peaking without DSL, as in the earlier plots. This produces a response that would give 6dB of DSL compensation (*Fig. 16*). Note, however, that this DA volume means a physically big box for a 6.5″ driver.

TABLE 4
PARAMETERS FROM *TABLES* 2 AND 3 GATHERED BY TYPE

DRIVER	f_S	f_A	f_B	f_E	f_K	f_L
Ref. #109K	45.6				59.2	59.3
Ref. #111D	50.1	71.1	69.5	69.2	60.3	60.3
Ref. #119X	30.1	89.3	80.9	81.6	58.3	59.8
Ref. #139C	27.6		61.5	60.1	47.8	46.1
6x9B	75.8					95.3
	Q_{ES}	Q_{EA}	Q_{EB}	Q_{EE}	Q_{EK}	Q_{EL}
Ref. #109K	0.57				0.82	0.76
Ref. #111D	1.06	1.32	1.44	1.24	1.31	1.23
Ref. #119X	0.59	1.74	1.81	1.61	1.40	1.29
Ref. #139C	0.59		1.27	1.16	1.09	0.99
6x9B	1.03					1.35
	Q_{MS}	Q_{MA}	Q_{MB}	Q_{ME}	Q_{MK}	Q_{ML}
Ref. #109K	2.28				2.06	1.61
Ref. #111D	3.63	1.81	2.22	1.78	2.73	2.14
Ref. #119X	2.37	1.53	1.92	1.56	2.13	1.50
Ref. #139C	2.96		1.56	1.20	2.07	1.32
6x9B	4.22					2.55
	Q_{TS}	Q_{TA}	Q_{TB}	Q_{TE}	Q_{TK}	Q_{TL}
Ref. #109K	0.46				0.59	0.51
Ref. #111D	0.81	0.77	0.87	0.73	0.89	0.78
Ref. #119X	0.45	0.82	0.92	0.79	0.85	0.69
Ref. #139C	0.48		0.70	0.59	0.71	0.57
6x9B	0.84					0.88
Layers		3	3	2	2	1
DA—in^3		345	480	480	1,182	1,182

Note: Final subscript of "s" means baffle data. Other final subscript indicates IB type.

TABLE 5
PROTOTYPE CLOSED-BOX DESIGNS

DRIVER	Q_{TC}	f_3	dB PEAK	V_B—in^3	DA—in^3
Ref. #82A	1.25	51.9	2.7	2,376	1,758
Ref. #111D	1	48.5	1.25	1,415	1,047
Ref. #119X	1	52.1	1.25	1,248	924
Ref. #139C	1	45.2	1.25	1,196	885

Note: Q_{TC}, f_3, peak, and V_B refer to CB prototype and DA to corresponding IB.

Figures 17–19 show the same information for the 8″ Ref. #119X driver, except that 3dB DSL correction is shown in *Fig. 19*. In our opinion, IB type J with this driver would thus be appropriate for a floor-standing system. *Table 8* compares the system f_3 values for IB type J with no DSL correction with those of the CB prototypes for all four drivers. Again, the agreement is good.

We attempted to suppress the rear-leakage dips in the frequency range 150–220Hz by lining the bare walls of the DA volume with 1½″ nominal thick fiberglass. This converts IB type J into IB type K, which still has 1,182 in^3 DA volume and two-2″ damping layers. We tested the Ref. #109K 6.5″ driver, with a moderate Q_{TS} = 0.46, in both these IB types. The CB prototype for this driver with V_B = 1,596 in^3 (DA volume/0.74) indicates a total system Q_{TC} of 0.64 and an f_3 of 70.7Hz. What would the system response be with this low total-Q design?

Figures 20 and *21* show the measured responses for the driver in the two IB types, showing cone responses without peaking. Note also the fiberglass lining had almost no effect on the rear-leakage dips in the 150–220Hz range, but greatly reduced the rear leakage above 300Hz. Clearly, lining the DA volume with fiberglass is worthwhile. (Later testing showed it also reduced the cone response peaking.)

Figure 22 shows the system response for the Ref. #109K driver on IB type K with no DSL correction, again with no peaking. The f_3 value from *Fig. 22* is about 62Hz, which is below the CB-prototype-predicted 70.7Hz. When you design with a CB prototype for a Q_{TC} well below unity, the IB system f_3 will be below that predicted by the prototype.

We converted IB type K to IB type L

TABLE 6
RESULTS FOR f_3 WITH IB TYPE H

DRIVER	f_3 FOR IB TYPE H	f_3 FOR CB PROTOTYPE
Ref. #119X	55	52.1
Ref. #139C	42	45.2

TABLE 7
PROTOTYPE CLOSED-BOX DESIGNS FOR V_B = 1,596 in^3

DRIVER	f_3	Q_{TC}	dB-PEAK	ALPHA
Ref. #82A	54.2	1.32	3.1	0.498
Ref. #111D	47.9	0.98	1.1	0.455
Ref. #119X	49.7	0.91	0.7	3.01
Ref. #139C	42.9	0.90	0.7	2.50

TABLE 8
COMPARISON OF SYSTEM f_3 WITH CB PROTOTYPE FOR IB TYPE J

DRIVER	f_3 FOR IB TYPE J	f_3 FOR CB PROTOTYPE
Ref. #82A	52	54.2
Ref. #111D	46	47.9
Ref. #119X	49	49.7
Ref. #139C	40	42.9

by removing the last damping layer and its spacer. Thus, IB type L has a DA volume of 1,182 in^3 with a single 2″ layer of damping material. *Figure 23* shows the measured responses for the Ref. #109K 6.5″ driver in this box. While the overall rear-leakage level rises as expected, the nasty dips in the 150–220Hz range are reduced.

This supports the possibility that something about the damping material is causing the nasty dips. Perhaps using a variety of damping material densities would reduce these dips, but we did not test this in our work. *Figure 24* shows the system response with no DSL correction; while f_3 is unchanged it shows deeper rippling than evidenced in *Fig. 22* for two damping layers.

SYSTEM GROUP DELAY

We don't have the ability to compute or measure the transient response of the IB. However, we can compute the group delay, which is an indicator of the transient response. You want a group delay that is low and constant. We will compare the group delay for two drivers in the IB with the same drivers in CB and VB designs using the box volume of the CB prototype. Thus, the DA volume of the IB is 74% of the volume used for the other boxes.

On the IB group delay plots, which are based on measurement, you want to ignore anything below about 30Hz, because noise is contaminating the data. This is no problem because 30Hz is well below the system passband. Also, any sharp triangular spikes at high frequencies are anomalies caused by the software trying to sort out the wild phase variations of the rear-leakage response.

Figure 25 shows the group delay for the Ref. #109K 6.5″ driver on IB type K. Above 30Hz, the worst case is about 5.5ms, and it decays smoothly with rising frequency without major peaking. *Figure 26* is the theoretical group delay for this driver in a 1,596 in^3 CB; i.e., the CB prototype. This shows the smooth group delay typical of a CB and has a maximum of slightly over 4ms.

The vented-box group delay in the same box size (*Fig. 27*) shows the typical shape for this box type with a peak of about 14ms around 40Hz. The IB measures to have about the same group-delay function as the CB prototype.

Figure 28 shows the group delay for the Ref. #119X 8″ driver on IB type K with a worst case—above 30Hz—of about 6ms and smooth decay with no major peaks. *Figure 29* shows the 1,596 in^3 CB prototype's group delay, which almost matches that for the IB. The group delay in a VB of the same size (*Fig. 30*) shows an unusual shape with larger maximum group delay. This is a small VB for an 8″ driver; if used in a bigger VB it would have a better response but much more group delay.

The general result is that the group delay of an IB is very nearly the same as the CB prototype used to design the IB. It is expected that the IB will hold the same advantage in transient response over a VB that is enjoyed by the CB.

LOSS AND PHASE SHIFT THROUGH DAMPING MATERIAL

The loss (attenuation) and phase shift through the damping material depend on more than just the number of layers used. All of the following will affect them:

1) Number of damping layers—1, 2, or 3.
2) Dead-air volume—345 to 1,182 in^3.
3) Rear area that is open—full, half, or one-quarter box area.
4) Fiberglass lining on dead-air volume walls—1.5″ or none.
5) Fiberglass layer ahead of damping layers—3″ or none.
6) Air gaps between damping layers—⅞ or 1″.
7) Type of damping material used (not tested in this work).

We tested the Ref. #119X 8″ driver in more IB types than any other driver. We examined all of the plots for this driver and recorded the loss and phase shift for various IB types that varied only in one parameter. Note that we do not have sufficient data to draw any conclusion on the effect of the air gap between damping layers, so we ignored this variable. Also, all testing was with Owens-Corning #705.

The loss through the damping material was obtained as follows. We examined a plot for the proper conditions and recorded the difference between the cone and rear-leakage response magnitudes for each frequency. These are the plotted values.

Figure 31 shows the difference in loss for two and three layers of damping material with other parameters fixed. *Figure 32* shows the results for one and two layers and cannot be compared to *Fig. 31* because other conditions are different. Note that two layers in *Fig. 32* just about accomplish the 18dB suppression in our earlier example, but only above 100Hz. The slopes of all these curves are about 6dB/octave and they all indicate the loss goes to 0dB as frequency approaches zero.

Figure 33 shows the effect of DA volume on loss through the damping material, and it is considerable. A low DA volume might require more damping layers than a larger box. The effect on damping loss of the rear area that is open is shown in *Fig. 34*. Within the accuracy of these tests, we would say no effect is indicated. However, examination of the test plots shows that reduced rear area does lower the rear-leakage level at high frequency and is thus worthwhile.

Figure 35 shows the effect of a fiberglass lining in the DA volume on damping-material loss. The effect shown is minor, but examination of the test plots shows the high-frequency rear leakage reduced by this lining, so it should be used. The effect of adding a 3″ thick fiberglass layer ahead of two damping layers is shown in *Fig. 36*. There is some improvement in rear-leakage reduction as frequency rises.

We also read the phase-shift difference between the cone and rear-leakage responses from plots that had all but one parameter constant. This is not the same as the phase shift through the damping material. Common sense says the rear of the cone is 180° out of phase with the cone response over our frequency range of interest.

A driver's declining magnitude response limits low-frequency measurement, but *Fig. 37* verifies that the two cone faces do remain out of phase over 8Hz–800Hz as expected. Thus the phase shift in traveling from the rear of the cone to the back of the box is obtained by subtracting the phase difference values in *Figs. 38–40* from 180°. Most of this phase shift is due to the damping layers, the remainder to the in-air transit through the DA volume of the box.

Figure 38 compares the phase difference for two and three layers of damping material. More damping layers cause the phase difference to approach zero more quickly; i.e., the two box outputs tend to be more in-phase at any frequency. *Figure 39* shows the same result for one and two layers, but again can't be compared to *Fig. 38*. *Figure 40* shows, as with loss, the DA volume has a major effect on the phase shift through the damping material. More DA volume exhibits less phase difference, meaning increased phase shift through the damping layers aided somewhat by the breadboard depth increase.

As noted, if you subtract the phase difference values shown in *Figs. 38* and *39* from 180°, you get the phase shift from the cone's rear to the back of the box. Converting this to time delay and subtracting the time delay traveling through the DA volume yields the time delay through the damping layers (*Fig. 41*). Measured steady state, this time delay varies with frequency.

These results show that building with the proper amount of DA volume uses the damping material in a more effective manner than simply pushing it right up against the rear of the driver's magnet structure. For example, if you built an IB with an 8″ thickness of damping material pushed right up against the driver magnet, and then removed the front 4″ of the damping material, this might produce a better-sounding system with a lower f_3 and without any change in box size.

TEMPER YOUR PREJUDICES

Some people can't stand to be in a room where a CB system with a Q_{TC} much above unity is playing music. The CB can sound terrible when designed with a high Q_{TC}. You must avoid thinking about the IB as though it were a CB. Testing has shown that even when the system total Q is well below unity the IB response can peak. Thus it is possible to build IB systems with an acceptable response peak for whatever purpose. When working with the IB you must get past your experience that a peaked response means bad sound.

FREE BASS?

Everyone is aware that in theory the VB

has a 3dB advantage over the CB because it uses both sides of the cone. This fact is many times misunderstood. If you use a given driver first in a CB and then in a VB, it will show the same efficiency. In such a case the 3dB advantage of the VB must be taken in some other form such as a lower system f_3 and/or a smaller box size. In reality, the VB never really uses the outputs from both sides of the cone at full amplitude at the same time, which is something that could increase efficiency.

Figure 42 shows the cone, port, and system responses for a B4-aligned VB. The VB enclosure is a tuned structure where the port makes the major contribution to the system output over a narrow frequency range. Throughout this range the cone response is suppressed. Thus, the flat-response VB tends to use one side of the cone or the other, but never both at once in a way that would increase efficiency.

The IB is different. First it is not a tuned system. The rear leakage can contribute over a wide bandwidth, for better or worse. If you build an open-back enclosure without damping material, the rear output from the cone tends to reduce the system output because it is out of phase with the cone's front output.

We have learned that the damping material produces a phase shift, so in the bass region the rear leakage does not tend to cancel but instead reinforce the cone response. Less damping material gives a higher rear leakage but less phase shift, so the leakage does not reinforce the cone response as effectively. More damping material yields two signals more in-phase but reduced rear-leakage amplitude.

Again, there is no free lunch, but some damping material thickness should produce the maximum amount of bass boost by having the right combination of leakage and phase shift. Note that this point may also produce a very rippled response. We did no testing to establish such an optimum point.

Does the boost in bass response with the IB represent free bass? Sometimes it does. Peaking coming from the cone response is not free, but comes at the expense of increased cone excursion just as with a CB. However, peaking due to the rear leakage adding to the cone response is "free." It comes without added cone excursion or increased driver power dissipation.

In theory, the absolute maximum SPL boost you could get from the rear leakage is 6dB, as correlative acoustic signals add as pressure and not as power. In practice, the fact that the two signals are not fully in-phase and the leakage is suppressed limits any such boost to a much lower value.

One damping layer on IB type L suppressed the rear leakage by 12dB at 100Hz. At this frequency the phase difference between the two outputs was about 73°, giving a bass boost of about 0.83dB. This is not a lot, but it is free. The value for 80Hz comes out to be about 0.71dB boost, but by 60Hz the two signals are more than 90° apart, so the rear leakage would cause a cut of about 0.45dB. Most of the peaking in an IB's response is due to the cone response shape and is not "free."

Part 2 offers the listening results and shows how you can build this IB concept for your own designs. ❖

A Beginner's Push-Pull or Single-Ended Amp

This versatile amp design lets you sample both push-pull and single-ended sound, just by replacing a few components. **By Larry Lisle**

There are two basic commonly used kinds of high-fidelity tube amplifiers: the single-ended type in which one tube drives the output transformer, and the push-pull circuit with two tubes connected to opposite ends of the output transformer. Forests of trees have been sacrificed to make paper for articles on which of these arrangements sounds best. The truth is that either configuration can sound great, but they are different.

How can a beginner decide which to build? That's the point of this article. It's about an amplifier that you can operate in either mode using mostly the same components and with about the same input requirements (about two-tenths of a volt for a comfortable listening level) to eliminate or control as many variables as possible. You can then decide by listening which sound you like and build a bigger amp accordingly, or just enjoy the music!

PUSH-PULL

Let's look at the push-pull version first. *Figure 1* shows a circuit that hasn't changed much since Colpitts invented it almost a century ago. A key component is the input transformer, a Hammond model 808. This is a line-to-grid step-up transformer of broadcast quality. The specifications say it's flat from 50 to 15,000Hz, and the example I used in the prototype is even better than that. (Hammond manufactures an extensive line of all kinds of transformers that aren't normally stocked by dealers such as Antique Electronic Supply, but AES can order them for you and get you a free catalog.)

The transformer steps up the voltage from an input device, such as a CD player, and supplies equal but out-of-phase voltage to the grids of the two tubes, V1 and V2. The 51,000Ω (or 51kΩ) resistor acts as a load for the transformer. The tubes are twin-triode 6SN7s.

In this circuit the two triodes in each glass envelope are connected in parallel (that is, plate-to-plate, grid-to-grid, cathode-to-cathode) to lower the plate resistance of each tube to about 3,500Ω. The 220Ω resistor in the cathode circuit provides bias by raising the cathodes about 4V above ground potential for direct current while the 470µF capacitor keeps the cathodes at ground potential for alternating current. Some designers omit the cathode capacitor as unnecessary, but I've found that using a capacitor across the cathode resistor helps eliminate the odd-numbered harmonics and also helps keep hum from the alternating current in the filament from affecting the cathode.

Many designs use direct current on the filaments of audio tubes. In addition, many have provisions for balancing the two sides of the circuit with a potentiometer connected between the two cathodes. I haven't used either of these for two reasons.

First, they didn't prove to be necessary: hum and noise in the prototype is down over 65dB and distortion is low at quiet listening levels. The second reason is that this is a beginner's amplifier. There's always the temptation to add this or that little extra until the circuit looks so complicated that a beginner won't even try to build it.

The output transformer is a Hammond 125-E. This isn't, strictly speaking, a high-fidelity transformer, but it's a very good one. It is also one of the few transformers that you can use for either a single-ended or push-pull circuit and has provision for multiple output im-

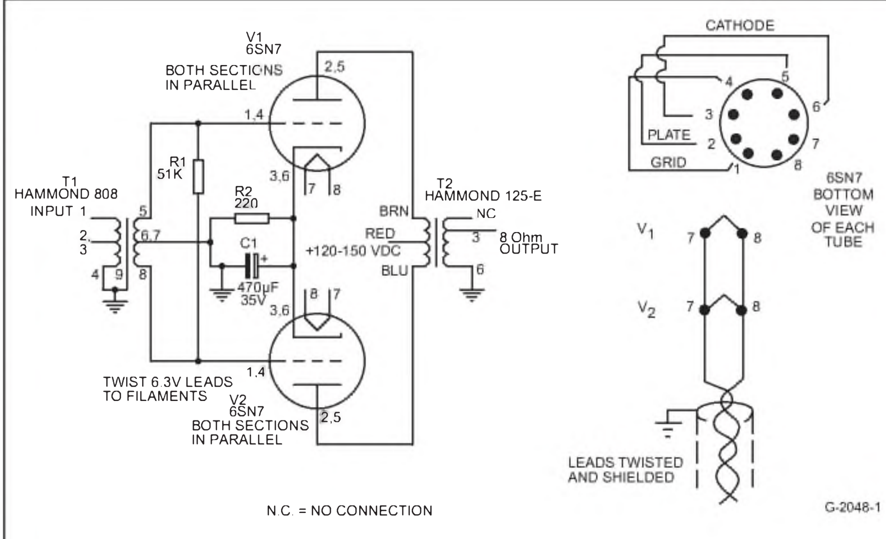

FIGURE 1: The circuit of the push-pull version. The input voltage from a CD player, for instance, is stepped up by the broadcast-quality Hammond transformer and applied to the grids of the 6SN7 tubes. The output transformer is a Hammond 125-E with multiple impedance taps.

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Jensen JE-34K-DX	3	47	20Hz~20kHz	550	
Peerless 4722	38	50	20Hz~20kHz	300	

**Air Economy

■Speaker

Model	Specifications					Price* (US $)	Postage** (US $)			
	D(cm)	Ω	Response	db	w		I	II	III	IV
Diatone P-610MB	16	8	45Hz~20kHz	90	7	360	30	40	50	66
Fostex FE208Σ	20	8	45Hz~20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz~20kHz	94	80	236	42	50	73	98
Onken OS5000T	—	8	7kHz~25kHz	105	2.5	4,000	70	84	133	181
ALE 1710 Tweeter	8	16	6kHz~	118	10	3,380	85	110	170	230

*Price is for a pair **Air Economy

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SRS-3030 Classic System II	
SRS-2020 Basic System II	
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■TANGO TRANS(28 models are available now)

Model	Specifications				Price (US $)	Postage** (US $)				
	W	Pri.Imp(kΩ)	Freq Response	Application		I	II	III	IV	
XE-20S (SE OPT)	20	2.5 , 3.5 , 5	20Hz~90kHz	300B,50,2A3	396	47	56	84	113	Price is for a Pair
U-808 (SE OPT)	25	2 , 2.5 , 3.5, 5	20Hz~65kHz	6L6,50,2A3	242	42	50	73	98	
XE-60-5 (PP OPT)	60	5	4Hz~80kHz	300B,KT-88,EL34	620	62	74	115	156	
FX-40-5 (PP OPT)	40	5	4Hz~80kHz	2A3,EL34,6L6	320	47	56	84	113	
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz~100kHz	300B,50,PX-25	620	62	74	115	156	
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	1.0	30Hz~50kHz	211,845	620	62	74	115	156	
NC-14 (Interstage)	—	[1+1 : 1+1] 5	25Hz~40kHz	[30mA] 6V6 (T)	264	30	40	50	70	
NC-16 (Interstage)	—	[1+1 : 2+2] 7	25Hz~20kHz	[15mA] 6SN7	264	30	40	50	70	

** Air Economy

■TAMURA TRANS(All models are available)

F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	740	60	70	110	145	Price is for a Pair
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	760	60	70	110	145	
F-2013	40	10	20Hz~50kHz	211,242	730	70	84	133	181	
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160	

** Air Economy

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pedances. This feature lets you experiment with the ratio of tube plate resistance to load impedance.

The usual practice is to make the load impedance twice the plate resistance with a single-ended circuit and double that for a push-pull circuit. However, I've built very nice-sounding amplifiers with the load impedance much higher and much lower than these guidelines. The 125-E lets you play around with this dimension. The output connections shown in *Figs. 1* and *2* are only a starting point; don't be afraid to experiment!

SINGLE-ENDED

It takes only a few changes to go from the push-pull circuit in *Fig. 1* to the single-ended circuit in *Fig. 2*. Be sure the power supply is unplugged from the wall outlet and all capacitors are discharged before touching any of the wiring on the amplifier!

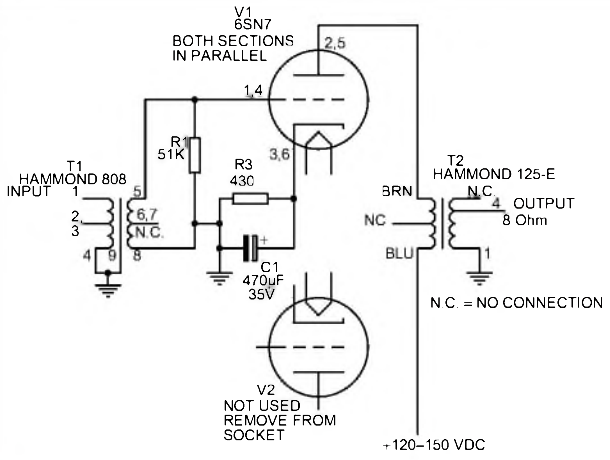

FIGURE 2: The single-ended version of the amplifier is even simpler than the push-pull. It's easy to change from one to the other, but be sure to unplug the power supply and short all capacitors first!

Disconnect the center-tap of the secondary of the input transformer from ground, and ground the lower end. Change the cathode resistor from 220Ω to 430Ω or to two 220Ω resistors in series. Because the current from only one tube is now going through the resistor the value must be increased to keep the voltage drop and consequently the bias the same at about 4V.

Change the connection from the power supply from the center-tap of the output transformer primary to the end connected to the plate of V2 and remove V2 from its socket. Finally, use the correct taps on the output transformer. Changing from one mode to the other takes only a couple of minutes, but double-check your work against *Fig. 2* before plugging in the power supply.

CONSTRUCTION

I built the amplifier on a 7″ × 10″ board, and you can see much of the wiring from *Photos 1* and *2*. The input transformer is mounted on a small aluminum box and requires a hole about ⅞ of an inch in diameter. The easiest way to make the hole is with a socket punch. If you don't have one, you can simply drill small holes in a circle and then knock out the center with a chisel. The rough edges, which you can smooth with a file, won't show because they're hidden by the transformer's case.

The tube sockets are partly wired before attaching them to the board with screws. Here are some suggestions to get you started:

Connect a green wire to pin 7 and pin 8 of each socket for the filament. Twist the wires together to reduce the hum field. You can slip a piece of grounded braided copper shield over the filament wiring as shown in the photos if hum is a problem, but I found it wasn't necessary in the prototype.

Connect a red wire from pin 2 to pin 5 of each socket for the plates, and connect a longer red wire to one of the pins. Connect a black wire from pin 3 to pin 6 for the cathodes, then connect a longer black wire to one of the pins. Connect a blue wire from pin 1 to pin 4 for the grids, and connect a longer blue wire to one of the pins. Now screw the sockets to the board, using a piece of "spaghetti" (insulated plastic tubing) around the screws to prevent any shorts.

Most connections are made with solderless Fahnestock clips, as shown in the photos. The one to the right of the input transformer is fastened with a bolt that goes all the way through the board to another Fahnestock clip underneath. This is the central ground connection, to which all ground connections are directly wired.

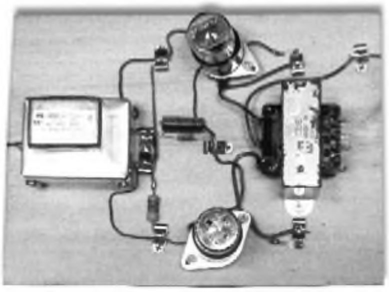

PHOTO 1: This simple amplifier gives you a very good idea of what a push-pull tube amplifier sounds like. It's simple to build and sounds great!

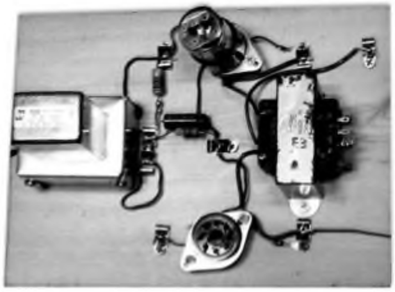

PHOTO 2: It takes only a few minutes to change the amplifier to a single-ended configuration. It also sounds great, but different.

PARTS LIST

C1	470µF electrolytic capacitor, 35WV DC
R1	51,000Ω resistor, 1 or 2W
R2	220Ω resistor, 1 or 2W
R3	430Ω resistor or two 220Ω resistors in series, 1 or 2W
T1	Hammond 808 line to grid transformer
T2	Hammond 125-E output transformer
V1, V2	6SN7 tubes with any suffix, eg. GTA, etc. but both tubes should be the same

Miscellaneous: Board 7″ x 10″, 4 rubber mounting feet, 2 octal sockets, Fahnestock clips, number 20 or 22 solid hook-up wire in various colors, small aluminum project box, screws, and so on. You can obtain most parts from Antique Electronic Supply, 6221 South Maple Ave., Tempe, AZ 85283, (480) 820-5411, www.tubesandmore.com.

The primary of the output transformer is connected to the appropriate clips. I don't like to cut transformer leads in experimental amplifiers, but you can if you so choose. You can add input and output connectors to match your existing equipment, or you can make direct-soldered connections.

You can use any power supply capable of putting out 120–150V at 30mA (but not a transformerless "AC-DC" supply) and 6.3V at 1.2A. I suggest it not be on the same board as the amplifier to prevent hash or hum problems and to be available for something else later.

I have been very pleased with this little amplifier. It sounds very nice in either configuration and is very representative of the push-pull or single-ended sound. It will give you a much better idea of what they are like than any book or article. ❖

Choosing and Using Electronic Parts: A Survival Guide, Part 3

Here's the final section in our series on how parts function and how to choose and maintain them. **By Charles Hansen**

TRANSFORMERS AND INDUCTORS

The two main categories of inductors are air-core and iron-core. Air-core inductors contain no magnetic iron. The only loss in an air-core inductor is due to the copper resistance. Iron-core inductors have cores made of various alloys of iron and/or other materials that give it the required characteristics.

In addition to the copper winding losses, iron-core inductors also produce losses associated with the iron itself. Eddy-current losses are caused by the varying magnetic field inducing a current to flow in the core. Hysteresis losses are caused by the nonlinear manner in which the iron core becomes magnetized and by the residual magnetism left in the core. Both eddy-current losses and hysteresis losses are considered to be real power losses[5,6,7].

The primary failure mode for wound devices such as transformers and inductors is an open winding, and the likelihood increases directly with the decrease in wire diameter. Both the insulation and the wire itself are more fragile and prone to damage during winding with smaller gauges.

Another failure mode is a winding short. If the device is exposed to excess current, both the wire insulation and interwinding insulation may fail. A shorted turn causes heavy current to flow and may cause the wire to fuse and open. A short can also occur to the core.

If the primary winding in a power transformer shorts to the core, and the transformer is bolted to the chassis, AC line voltage can end up on the chassis if there is no third-wire safety ground. If the equipment has a safety ground, the line fuse should safely open, clearing the fault. This is made less likely because transformers are usually made with the primary winding on the outside to maximize coupling to the secondary.

In older inductors or transformers, rust or corrosion may form on the laminations, causing an increase in eddy-current losses. This could change the magnetic properties enough to affect operation.

The failure rates versus temperature for audio and power transformers and chokes is shown in *Fig. 11*. There is no curve for failure rates versus voltage or current—this is determined by calculating the winding temperature rise and applying the temperature factor to the base failure rate.

VACUUM TUBES

Vacuum tubes are not listed in TR-332. Section 7 of MIL-HDBK-217 lists the failure rate (λ_b) for receiving tubes as 5 failures/10^6 hours (200,000 hrs MTBF), and 10 failures/10^6 hours (100,000 hrs MTBF) for rectifier tubes. These failure rates include random failures and wearout, and there is an additional π_L learning factor that penalizes tube types newly introduced in the last three years. Even if you never apply any plate voltage, tubes will still fail once the heater/filament opens or its emissions fall below a useful level, but these MTBF numbers should be welcome news to tube equipment owners who leave their components on at all times.

ELECTROMECHANICAL RELAYS AND SWITCHES

When a relay coil is energized, the contacts do not transfer immediately. A delay of 1μs to 10ms is required in order for action to take place. The magnetic field in the coil exerts a mechanical force on the movable armature and causes the contacts to make or break the circuit in which the relay is connected.

Time-varying magnetic and mechanical forces, as well as armature inertia, allow relay contacts to "bounce" one or more times when they open and close. In addition, if the contacts close into a low resistance, high current flow can cause a back-emf to be generated that tries to force the contacts open. Some over-travel is provided in the armature system to minimize rebound and provide for some contact wiping, but contact bounce is largely unavoidable.

When the circuit to an inductive load is opened, most of the energy stored in the load will be dissipated through arcing at the contacts. Design engineers must select contacts large enough so

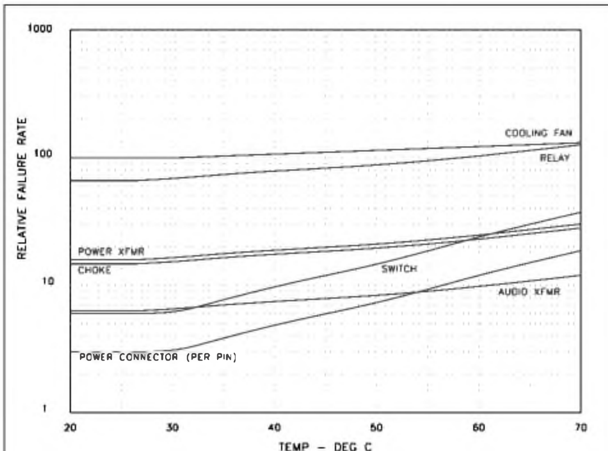

FIGURE 11: Miscellaneous item failure rate versus temperature.

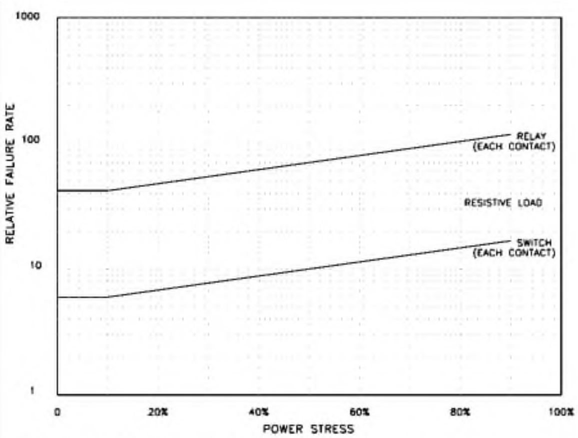

FIGURE 12: Relay and switch failure rates versus contact current.

that deterioration from destructive melting is minimized, but not so large that the current density falls below the critical level needed to ensure low contact resistance. A slight fusing of the contact surface after each operation is actually desirable to remove oxidation and maintain a fresh metal contact surface.

For "dry" circuits (very low voltage and current) it is necessary to use precious metal contacts that do not oxidize. If these contacts are ever exposed to high current, the arcing will destroy the surface plating, which cannot again be used in dry applications.

TABLE 3
SOLID-STATE POWER AMPLIFIER PARTS COUNT

QTY	PART TYPE
12	Transistor, bipolar <0.6W
6	Transistor, bipolar 0.6–6W
16	Transistor, bipolar >6W
9	Diode Si<1A
10	Diode Si, 1–20A
8	Diode, Si>20A (Bridge = 4 diodes)
3	LED
3	Zener, 0.6–1.5W
44	Resistor, metal film <1MΩ
18	Resistor, carbon film <1MΩ
20	Resistor, WW, leaded
1	Resistor, variable, film, <200kΩ (dual)
2	Resistor, variable, cermet trim, <200kΩ
2	Thermistor
22	Cap, plastic, metallized
8	Cap, ceramic
10	Cap, alum <400µF, axial
4	Cap, alum 400–12,000µF, axial
1	Transformer, power
6	Connector, power and speaker
2	Connector, coax
8	Connector, PC board, signal
2	Switch—toggle, PB, or slide
1	Relay
5	Fuse <30A

Switch contacts have all the characteristics of relay contacts. In the direct-acting contacts of slide and rotary switches, the speed of operation is that of the actuating lever or shaft. Toggle switches use indirect actuators to increase the contact speed, so they will bounce when opening and closing.

The failure rates versus temperature for relays and switches is shown in *Fig. 11*. The failure rates versus contact current for relays and switches (per contact set) is shown in *Fig. 12*.

CONNECTORS

The failure rates versus temperature for connectors (per pin) is shown in *Fig. 11*. While all plug-in connectors are given the same failure rate in the reliability specifications, the contact material is important, especially for low level (dry) signals. Copper and tin-plated contacts will eventually oxidize and cause increased noise, or a high resistance or open circuit.

Even silver, the best electrical conductor, is not immune to this problem. Noble metals such as gold or rhodium provide much better long-term performance. For those of you with optical connectors in your digital audio components, their λ_{SS} base failure rate according to TR-332 is 100 versus only 0.5 for electrical coax connectors.

SO WHAT CAN WE DO ABOUT IT?

If you design electronic audio equipment, the choice of stress derating and temperature are the two primary variables you have control over that will affect the reliability of your design.

PARTS DERATING

Everyone knows that you should never exceed the maximum specified rating (voltage, current, wattage, or temperature level) of a component. However, for long and reliable life, a part should never even be operated at its maximum specified rating. The ratio of the actual applied level to the maximum rated level is called the derating factor. In the absence of specific failure rates, a good rule of thumb is that every 20% reduction below the maximum rated electrical stress level will double the life of a component.

The same rule of thumb says that every 10°C reduction in temperature will double the life of a component. Refer to the temperature derating curve in *Fig. 13*. In general, the highest temperature that a part should be exposed to (including self-heating)—the design limit temperature (Tdes)—can be derived from the following formula:

$$\text{Tdes} = \text{Tlim} - \frac{(\text{Tlim} - \text{Tder})^{*}\ \text{Applied Stress}}{\text{DeratedMaxStress}}$$

There is a logarithmic relationship between stress and failure rate, so it pays to be conservative. If you wish to be unusually conservative, never apply more than half the rated voltage, current, or wattage to any component, but this may result in excessive component costs.

Table 2 (part 2) lists derating factors that are a bit more practical[8]. To apply these factors, multiply the maximum rating of the part by the derating factor. For instance, let's say a ceramic capacitor is rated for 50V. You should not apply more than 40V across the cap (50 × 0.80). The applied voltage is always the DC level plus the peak of any AC voltage ripple that may be riding on top.

Where more than one factor is given, you should not exceed any of the factors. For instance, the voltage and current deratings for an IC regulator are 0.80, which would lead to a power derating of 0.64. However, the power derating limit should not exceed 0.50 of ratings for these components, so either voltage or current will need to be less than 0.80 its rating.

TEMPERATURE

The relationship between reliability and temperature is an inverse exponential:

$$R = e \frac{T}{MTBF}$$

where R is reliability and T is temperature.

Ambient temperature is the most

important factor for long life expectancy and high MTBF. Removing heat is very important to increase reliability. Heatsinks (conduction) and fan cooling (convection) are more effective than simple heat radiation. This is because heat removal by conduction and convection are directly related to the thermal conductivity and temperature, while heat removal by radiation is exponential to the fourth power of temperature.

Compact designs make temperature control more difficult. Heat-producing components are forced closer to heat-sensitive components, and the internal temperature rise is greater for a given power dissipation because the heat-radiating surface area is smaller.

DESIGN TOOLS

One way to increase MTBF is to use fewer components. You can achieve this by minimizing the number of protection devices, which results in a higher calculated MTBF, but may produce a worse real-world reliability due to subsequent damage after a power component fails. In order to evaluate these cascading effects, reliability engineers use additional design tools.

One of the most important is the failure mode and effect analysis—FMEA (see sidebar in Part 1). The FMEA takes each mode of failure for a given component (short, open, degradation, value shift, and so on) and then determines the effect on the overall equipment. A failure in one device can propagate and lead to subsequent failures in other components.

An FMEA is complementary to the design process. Whenever an undesirable failure effect is found, the engineers will evaluate the compensating conditions, and perhaps change the design to make the failure effect more benign.

For example, assume an aluminum electrolytic power-supply capacitor fails short. The resulting high current could then cause the rectifier bridge or power transformer to fail, or the capacitor to explode. By including properly sized protection devices in the design, the effect of the power failure is limited to an open fuse or circuit breaker, even though the MTBF is decreased a bit. All the other components in the power supply are protected from damage.

There are also situations in which

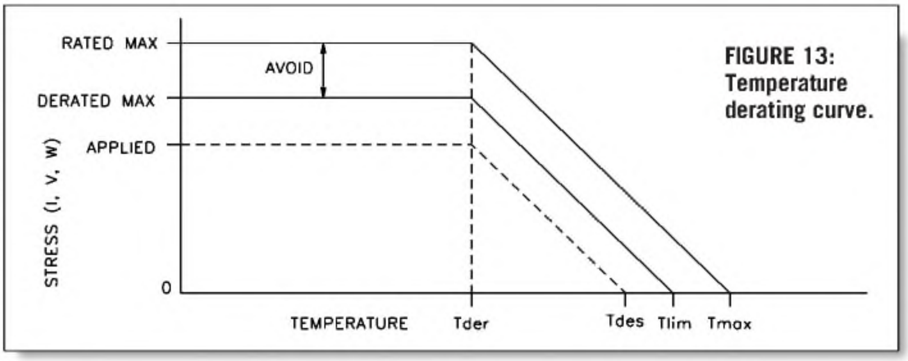

FIGURE 13: Temperature derating curve.

TABLE 4
AMPLIFIER COMPONENT FAILURE RATES, TR-332 PARTS COUNT METHOD

PART TYPE	TYPE FR	QTY	TYPE FR TOTALS
Relay	210	1	210
Resistor, variable, film, <200k (dual = 2)	105	2	210
Resistor, variable, cermet trimmer, <200k	75	2	150
Cap, alum, 400–12,000µF, axial	75	4	300
Transformer, power	57	1	57
Cap, alum <400µF, axial	45	10	450
Transistor, bipolar >6W	30	16	480
Resistor, WW, leaded	30	20	600
Thermistor	30	2	60
Switch, toggle, PB, slide	30	2	60
Diode, Si>20A (Bridge = 4 diodes)	27	8	216
Transistor, bipolar, 0.6–6W	18	6	108
Diode, Si, 1–20A	18	10	180
Zener, 0.6–1.5W	18	3	54
Connector, power (per pin)	15	6	90
Fuse <30A	15	5	75
Transistor, bipolar <0.6W	12	12	144
Diode, Si<1A	9	9	81
LED	9	3	27
Cap, plastic, metallized	3	22	66
Cap, ceramic	3	8	24
Resistor, metal film <1M	1.5	44	66
Resistor, carbon film <1M	1.5	18	27
Connector, coax	1.5	2	3
Connector, PC board, signal (per pin)	0.6	8	4.8

operating procedures can increase the adverse effect of a failure mode. For instance, in order to maximize the life of vacuum tubes and prevent cathode stripping, it is advisable to delay the application of B+ voltages until after the tube heaters warm up. Some manufacturers provide automatic time-delay relays for this function. Others provide separate power and B+ switches that require the equipment owner to manually delay the B+ application. The second method is more reliable from a component failure rate standpoint, but it is also prone to inadvertent "operator error."

MTBF CALCULATION EXAMPLE

Let's take a hypothetical solid-state power amplifier and determine its MTBF in a benign home audio system. It is a 75W stereo amplifier with two paralleled pairs of bipolar output transistors, a protection relay, and an input volume control. The parts breakdown is shown in *Table 3*.

The quickest way to make a reliability estimate is by using the Parts Count method from TR-332, in which you assume an operating temperature of 40°C and electrical stress on all parts to be 50% (i.e., a 100V cap has 50V applied, and so forth). Also use Device Quality Level I nonhermetic parts, to avoid the considerable penalty for Quality Level 0 "remanufactured, reworked, salvaged, or gray-market components."

Using all the failure rates, π factors, and part quantities, you find $\lambda_{\text{amplifier}}$ to be 3742.8 failures/10^9 hours, or an MTBF of over 260,000 hours. That's over 30 years of continuous 24/7 operation, but it presumes that more than 7,000 hours of operating time have already elapsed.

Even if you factor in the first year multiplier to account for early failures, the MTBF is still a healthy 66,795 hours (over nine years of continuous operation). That could explain why the dreadful-sounding receiver I used in college is still working. Your garage door remote control, with its low on-time duty cycle, could last "forever."

Environment is everything. If you took your amplifier from its controlled home environment and put it in orbit in a communications satellite, the MTBF would be 1⁄15 that of your home, or 5089 hours—just under seven months. The *MIL Handbook* has a similar factor for low-earth orbit.

If your amplifier were shot from a cannon, the component MTBF would drop to 238 hours. (The actual amplifier life would, of course, be microseconds given its mechanical packaging.) That gives you a real appreciation for the difficulties that engineers and scientists faced while developing WWII antiaircraft proximity fuses using fragile vacuum tubes.

I also calculated the failure rate for the hypothetical amplifier using the MIL-HDBK-217F Parts Count method, with the same 50% electrical stress and 40°C operating temperature. The MTBF was 59,592 hours, or more than seven years of continuous operation. The *MIL Handbook* penalizes aluminum capacitors and the protective relay much more heavily than the TR-332 does.

Ironically, the fuses and the protection relay and its sensing circuitry, all designed to save your speakers in the event of an amplifier failure, causes the MTBF of the amplifier to drop by almost 12%!

Now, look at the parts with the highest failure rate and see what effect changes in electrical stress and operating temperature have on the failure rates of these five parts and the overall amplifier MTBF. *Table 4* is sorted by Type Failure Rate (2nd column), while the 4th column is the Total Type FR, or Type FR multiplied by quantity.

You can see that devices with moving parts (relay and pots) have the highest failure rates. Next are the large aluminum capacitors. The power transformer also has a fairly high λ. I'm not sure why TR-332 does this—it is not penalized as heavily in the *MIL Handbook*. This list gives you a feel for the first failures you can expect as the amplifier operating hours increase.

Finally, I ran the TR-332 MTBF calculations using the same 40°C average operating temperature, but with actual electrical parts stresses, and calculated case temperatures in the power devices. The MTBF (factored for early failures) dropped from 76,337 hrs to 69,691 hrs, or 9% lower. The biggest penalty was due to the temperature rise in the output transistors and the higher voltage stress on the 63V DC rated aluminum reservoir filter capacitors (55V DC applied versus 31.5V DC for the 50% stress factor in the Parts Count method). The top six failure rate parts are now the relay, the large aluminum capacitors, the smaller aluminum capacitors, the dual volume control pot, the power transformer, and the trimmers.

I hope this article will give you a feel for the design versus cost decisions that engineers must make to bring a product to market. ❖

REFERENCES

5. "All About Toroidal Transformers," C. Hansen, *Popular Electronics*, May 1996, pp. 53–56, 78.

6. "Berning ZH270 Power Amp, Part 1," C. Hansen, *Glass Audio* 1/00, pp. 23–30.

7. "Power Transformers for Audio Equipment," P. Millett, *audioXpress* June '01, pp. 14–19.

8. *The Joy of Audio Electronics*, Chapter 3, C. Hansen, Audio Amateur Press, Old Colony Sound Lab, www.audioXpress.com.

Building the Adire KIT281 Speaker

An easy speaker project that delivers great performance for the money. **By Pete Millett**

PHOTO 1: A finished KIT281 speaker.

I really like to design and build hi-fi tube amplifiers. Because of this, I find myself with more amplifiers around the house than speakers. What to do?

Why, build some speakers, of course!

My requirements for speakers are probably not unlike those of many people. I wanted something that sounded good, especially with lower-powered tube amplifiers. I didn't want to spend more than around \$500 for the drivers and crossovers. And I wanted something with enough bass so as not to require a subwoofer.

I found lots of designs that almost fit my wish list: tons of MTM designs—both bookshelf and tower—with 5″ woofers (and some with 6″). But almost all of them looked as though they lacked bass, or used expensive drivers. Because this was to be my first speaker-building attempt, I didn't want to use drivers that were too expensive, in case I messed up something.

ENTER THE ADIRE AUDIO KIT281

In my search, I stumbled across Adire Audio on the internet. Adire (formerly Avatar Audio) sells both finished speakers and a number of speaker kits, as well as raw drivers and subwoofer amps. Their KIT281—at least on paper—seemed to fit the bill exactly: a pair of 8″ woofers to provide good bass extension, efficiency specified at 92dB/1W (not quite 2A3 material, but OK for my 6–30W tube amps), and, best of all, \$339 per pair for all the parts, including drivers, hardware, and assembled crossovers (*Photo 1*)!

The only option (which I chose to include in my speakers) is a flared port kit, which adds another \$38 to the kit price. This kit alternative provides a port tube with a wide flare on both ends, which I believe reduces turbulence at the port ends, and looks really nice as well.

The KIT281 is an MTM design that uses two of Adire's proprietary AV8 8″ midwoofers and an Audax TM025F1 tweeter. Adire provides three different box designs that you may build, with no other changes to the kit.

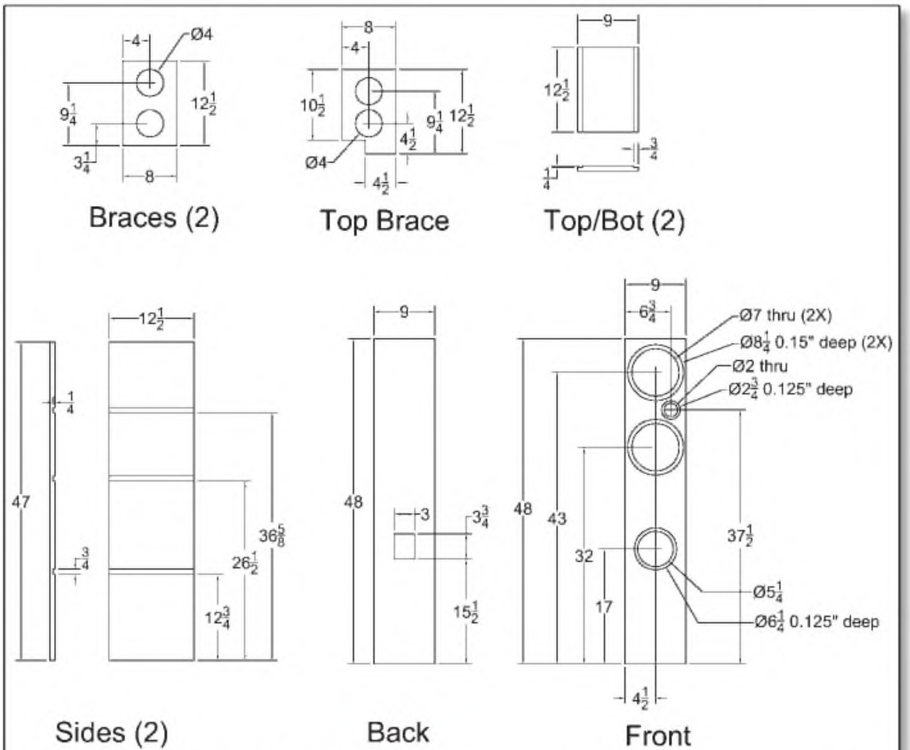

FIGURE 1: Individual parts to assemble the KIT281 boxes. B-2068-1

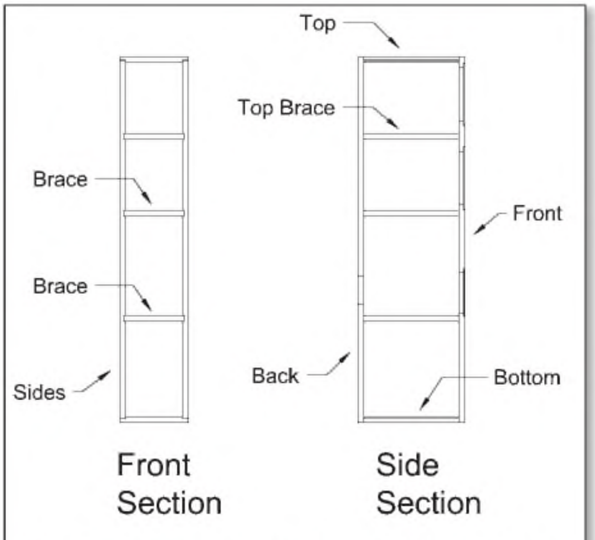

FIGURE 2: Assembly of the parts to build the KIT281 boxes.

They offer a sealed, bookshelf box design, a floor-standing transmission-line design, and the box I built: a floor-standing vented box, which promised a –3dB bass extension down to 39Hz.

The crossovers provided with the kit were completely assembled on two printed circuit boards per speaker, and even had all the needed wires attached and pre-terminated. I did not even need to pick up a soldering iron to assemble these! Components used on the crossover looked reasonably high-quality, with air-core inductors and polypropylene capacitors (no electrolytics).

BUILDING THE BOXES

Since there were no crossovers to build, the construction of this kit is mostly a woodworking project—that is, building the boxes. Though I'd never built a speaker, I had done a fair amount of woodworking, and found the box construction straightforward.

The vented floor-standing design is 9″ wide, 14″ deep, and 48″ tall—not small, but not huge, either. The box (like most speakers) is constructed of ¾″ MDF (medium density fiberboard). The figures shown here are based on the design on the Adire website, with only some assembly-related details modified (such as dadoes instead of flat joints).

Cutting the MDF

First, I transcribed the Adire drawings into an AutoCAD drawing. I made a couple of changes—mostly involving the use of dado joints (where one part fits into a groove on the other) instead of simple butt joints. The drawing of all the individual parts is shown in *Fig. 1. Figure 2* shows how the pieces fit together.

Before I started cutting, I made a drawing to figure out the best way to slice up a sheet of MDF (*Fig. 3*). I laid out the parts so they could be cut from three half-sheets of MDF. I find it's very difficult to handle full sheets on the table saw, even with a helper, while I can cut half sheets pretty well by myself. My local home-improvement store cut the sheets down the middle for me at no charge.

PHOTO 2: Top, bottom, and braces as cut.

PHOTO 3: Sides and backs as cut.

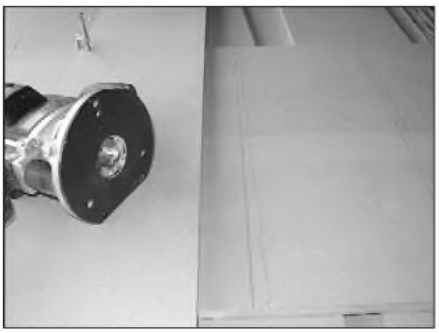

PHOTO 4: Setup to cut straight dadoes in sides, two at one time.

PHOTO 5: Cutting straight dadoes in sides two at a time.

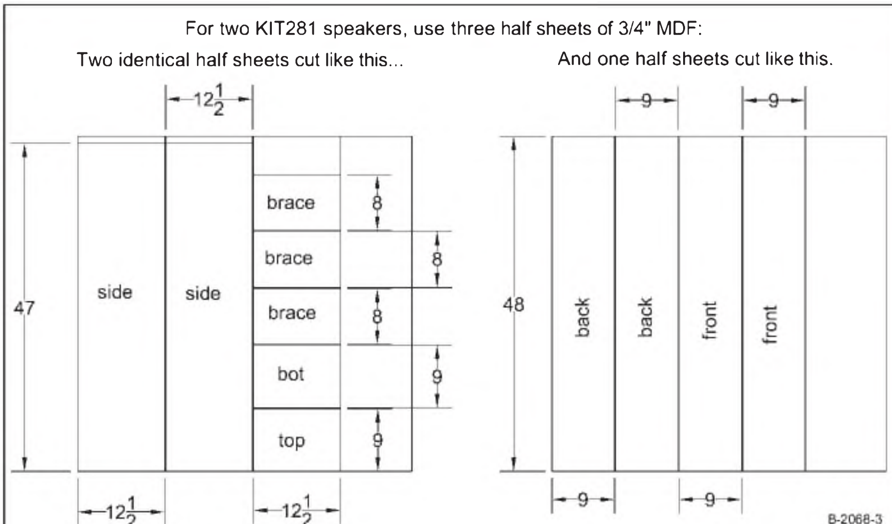

FIGURE 3: Cut sheet, showing how to cut the required parts from three half-sheets of MDF.

Note that MDF sheets are oversize—instead of 4 × 8′, they actually measure 49″ × 97″. This is a good thing for this design, because the longest parts are cut to 48″.

I used a table saw to make all of the cuts. I think you could do a decent job even without a table saw, if you have a good circular saw and use a straight edge as a guide, so your cuts don't wander too much.

Photos 2 and *3* show the individual parts all cut to size. Note that the drawings show a 5¼″ hole for the optional flared port. If you decide not to use this option, the port hole needs to be decreased to 3″.

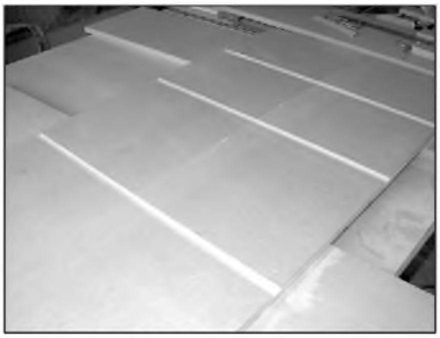

PHOTO 6: Sides with completed dadoes.

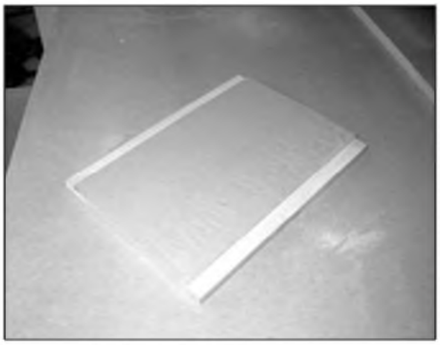

PHOTO 7: Finished top or bottom.

PHOTO 8: Router with circle jig, preparing to cut round holes in brace.

PHOTO 9: Circular dadoes in the two front panels are cut before the circular holes.

Dadoes and Holes

After cutting the individual parts, I started milling them. I cut straight da-

PHOTO 10: Finished braces. Top braces on left.

does in the sides (*Photos 4*, *5*, and *6*) for the inner braces to fit in. This makes a very strong joint, and helps align the parts so they stay in place while you glue pieces together.

These dadoes are not required, and are not shown on the Adire drawings. But, since I had a router available, it made sense to do it this way. Similarly, I cut a rabbet in the top and bottom (*Photo 7*).

As shown in the drawings, there are a number of round holes that you must cut in the braces, as well as the round holes in the front panels to install the drivers. You could cut these holes with a reciprocating saw (sabre saw), but I decided to buy a circle-cutting jig for my router. I'm sure experienced speaker builders know all about these things, but this was totally new to me. The jig (*Photo 8*) wasn't cheap (around $50), but it made quick work out of cutting beautifully round holes!

I also used this jig to cut circular dadoes on the fronts of the speakers (*Photo 9*) to flush-mount the drivers and the flared port tube. According to Adire, this isn't really necessary, and is more of a cosmetic issue than anything. But since I had the router and circle jig, I went ahead and cut them this way. [Any discontinuity around a driver rim will have a physical effect on the signal, called diffraction. Whether this is audible in any given case is still undocumented—Ed.]

I also rounded over the edges of the holes in the braces with the router and a ⅛″ roundover bit. Again, this is probably not necessary, but I figured it couldn't hurt to have a smooth edge there. The finished parts are shown in *Fhotos 7* and *10*.

Box Assembly

Assuming that you've accurately cut your parts, assembly is easy. I assembled the box using only glue and clamps—no screws or nails. I used a good wood glue, Titebond II, which worked very well with the MDF.

First, I generously applied glue to the dadoes in the sides. I then installed the braces into the dadoes and used bar clamps to hold the assembly together

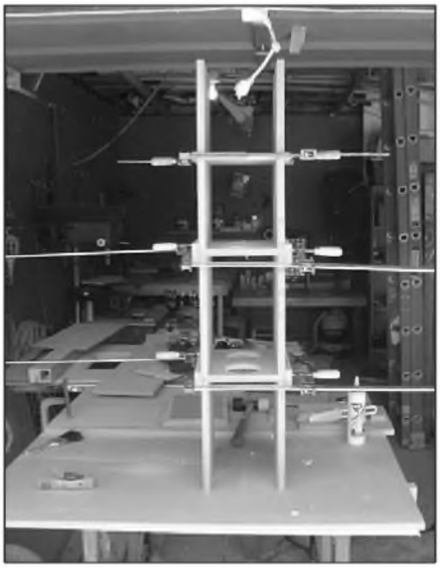

PHOTO 11: Gluing the sides together.

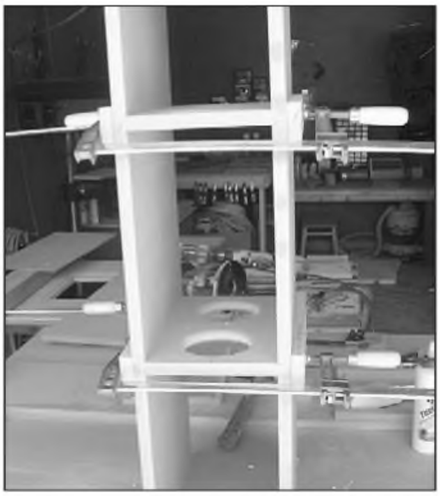

PHOTO 12: Gluing sides together (close-up of braces).

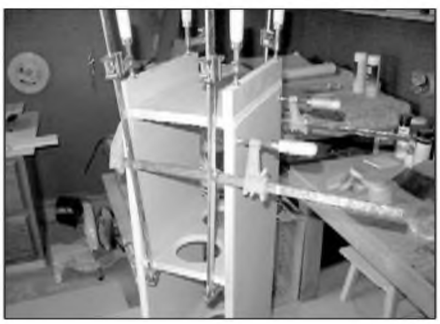

PHOTO 13: Gluing on the top. The bottom is done the same way.

PHOTO 14: The boxes, without front and back.

PHOTO 15: Gluing on the front. The back is done the same way.

PHOTO 16: Drilling holes for the driver mounting T-nuts.

PHOTO 17: A regular T-nut (right) and a threaded insert.

for a couple of hours (*Photos 11* and *12*).

After the glue was dry, I glued on the top and bottom pieces, using bar clamps to pull the top against one of the braces inside (*Photo 13*). I repeated the process for the bottom. The boxes are shown in *Photo 14* with the glue dry and the clamps removed.

I used a "long board"—actually, a 1′ scrap of MDF with sandpaper glued to it—to straighten the exposed edges that are glued to the front and back panels. I didn't want any gaps in this butt joint. After sanding the edges flat, I glued and clamped the front and back pieces (*Photo 15*).

BOX HARDWARE AND FINISHING

Next, I installed hardware into the box to allow the drivers to be mounted (and removed) using standard machine screws. The Adire kit includes all the hardware and screws that are needed...well, almost all of them (as you'll see later).

To locate the mounting holes, I carefully placed each driver and the flared vent tube into place, and marked the hole locations onto the MDF with a pencil. After removing the drivers, I used a hand drill with a jig to ensure the holes were straight (*Photo 16*).

At this point, I noticed that the "T-nuts" wouldn't fit well on the edges of the woofers. The box was too narrow to allow them to fit inside the sides of the box. Rather than try and cut the T-nuts, my solution was to use threaded inserts in the holes on the edge, which are threaded into the MDF in a slightly larger hole than the T-nuts.

These inserts are shown (along with the regular T-nuts) in *Photo 17*. Note that if you use inserts, you will also need to use shorter screws than in the holes with the T-nuts. This little glitch was the only part of the whole project that didn't go exactly as expected!

With the hardware installed, I sanded the boxes and rounded over the edges with the router, just so there were no sharp edges (*Photo 18*). The only work remaining to complete the boxes was to sand them and apply a finish of your choice—paint, veneer, or no finish. I haven't had many good experiences with veneer, so I opted to just paint the boxes.

I decided to try an automotive lacquer finish, hoping that I could get a nice, glossy black finish on my speakers. I used a sanding sealer, then several spray cans of primer, sanding at each step, then six to eight cans of lacquer finish paint. After painting, I sanded, rubbed, polished, and waxed the boxes. It turned out to be a very laborious process—in fact, more so than building the boxes.

I never did achieve that "piano" finish—more like an authentic Ford Model-T paint job, I think. Oh well, it's not too ugly (*Photo 19*). Maybe next time I will try veneer.

FINAL ASSEMBLY

With the boxes done and the crossovers pre-built, final assembly of the speakers is easy. You must cut the flared port, which is a $38 option on the KIT281, to the correct length and glue it together prior to installation in the box. The overall length of the flared port is specified at 8.5″. To get this length, cut the tubing just over 2.5″. Then glue together, using the kind of glue used to assemble ABS

PHOTO 18: The completed boxes before paint.

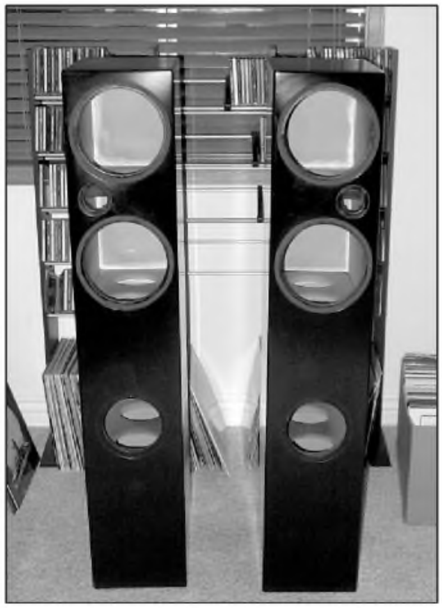

PHOTO 19: The finished boxes.

PHOTO 20: The flared port, cut to length and glued together.

PHOTO 21: The crossovers, as supplied. HF is on left, LF on right.

plumbing pipe. The assembled port is shown in *Photo 20*.

The crossovers (*Photo 21*) are installed on the back, opposite the lower midwoofer. They can be easily accessed through the woofer cutout. I placed some rubber feet (not included in the kit) between the crossover PCB and the back wall (*Photo 22*), to ensure that there wouldn't be any stress on the PCB and to eliminate the possibility of vibration. *Photo 23* shows the crossovers mounted inside the box.

The input terminal is installed using the wood screws provided, and the input wires from the crossovers are connected to it. While not labeled, the HF (tweeter) input usually goes on the top set of terminals, and the LF on the lower. Put the red wires on the "+" terminals and the black on the "–" terminals. *Photos 24* and *25* show the input terminals, from the outside and inside, respectively.

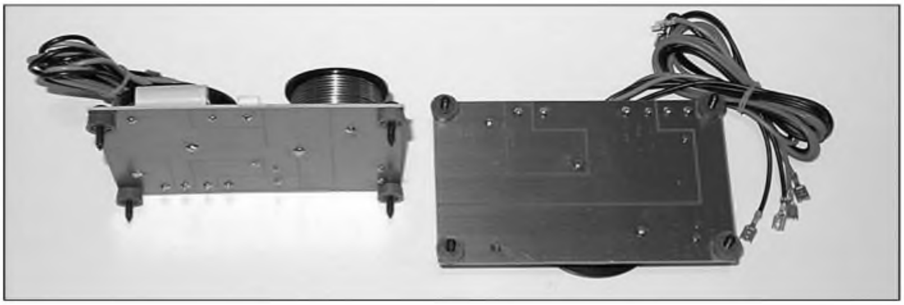

PHOTO 22: Rubber feet placed under the crossovers prior to mounting.

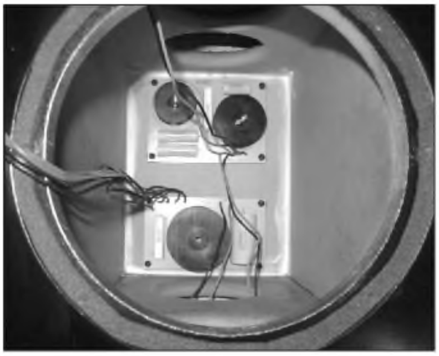

PHOTO 23: The crossovers installed inside the box.

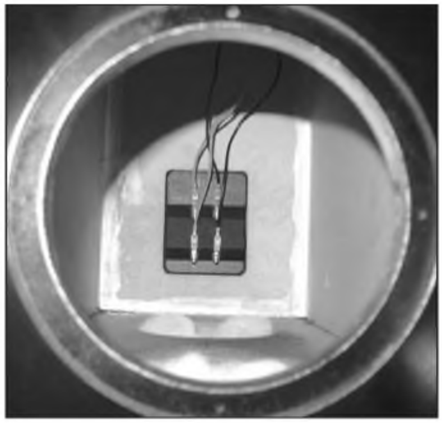

PHOTO 25: The input terminals viewed from inside the box.

Stuffing

Before installing the drivers, add polyester stuffing inside the box. Adire recommends using a total of 24 ounces of polyfil in each speaker. Use 12 ounces to stuff the lower section (below the lowest brace) and 6 ounces behind each of the midwoofers.

When adding the stuffing, you need to route the wires to the drivers so that you can connect them as you install the drivers (*Photo 26*). I used a product from Stearns Technical Fibers called Fiberloft, which is available at fabric stores. I'm sure you could also use the more expensive stuffing material sold for speaker use.

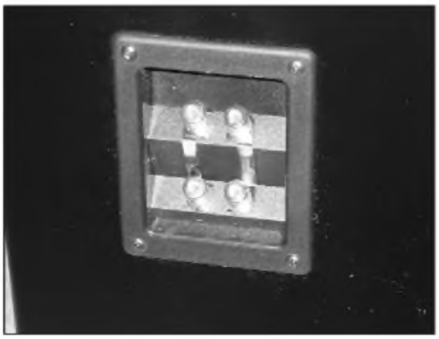

PHOTO 24: The input terminals.

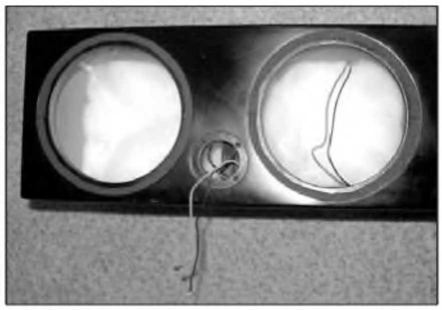

PHOTO 26: Stuffing the box. Wires from crossovers are routed to the driver cutouts.

Installing the Drivers

The KIT281 is supplied with two Adire AV8 8″ midwoofers and an Audax TM025F1 tweeter. The drivers are shown in *Fhoto 27*.

The AV8 drivers are provided with a thin gasket, which is stuck to the front of the driver. You can use this gasket to seal the driver to the enclosure, but I chose to use a thicker weatherstrip material. I applied the strip, with an adhesive on one side, to the dado in the cabinet before installing the driver.

The wires from the crossovers are attached to the terminals on the drivers as you install them into the cabinet. The tweeter terminals are different sizes, so you can't get them in the wrong spots. Connect the woofer wires so that the black wire is connected to the terminal marked "–" and the orange wires are connected to the terminals marked "+". The markings are hard to see on the driver, but they are stamped on the fiber insulator on which the terminals are mounted.

After attaching the wires, you hold the drivers in place with machine screws threaded into the T-nuts (or threaded inserts) that were installed into the box earlier.

PHOTO 27: The drivers—Adire AV8 midwoofers (2) and an Audax tweeter.

They're Done!

With the drivers installed, the speakers are done! The finished speakers are shown in *Photo 28*.

When I first connected the speakers and played some music, I noticed that the image was all wrong. I removed the input terminal from one speaker, and, sure enough, I had reversed the polari-

PHOTO 28: The Adire KIT281s, next to Klipsch KLF-10s and the rest of the system.

ty of both the HF and LF inputs on one of the speakers, causing them to be out of phase. I simply reversed the wires and tried again.

Initially, the bass response was somewhat thin. This seems to be common with new drivers, and after a couple of hours, they sounded fine. I report my listening impressions in the review section, which also includes measurements of the finished system (p. 61).

CONCLUSION

The Adire KIT281 is an excellent value, a good-sounding speaker, and an easy speaker to build. Anyone with a bit of woodworking experience can build the cabinets. Although a table saw and router make the job easier and the results arguably better, you can build the cabinets using only modest power tools. Since the crossovers are pre-built, there is no soldering required, and no electronic experience whatsoever is required for a successful project.

Delivery of the kit was less than one week, and the few questions that I came up with while building the speakers were quickly answered by Adire via e-mail. Their website will provide all of the information that you'll need. Thanks to Dan Wiggins and Adire Audio for providing a great product! ❖

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A PL504 PP Amplifier

This tube amp won't break the bank, and promises results that are pleasing to the ear.

By Mark Taverniers

PHOTO 1: Front view panel showing a neon mains-on lamp, sided by three double-pole switches: mains on/off, filament on/off, and H.V. on/off, respectively.

The idea for this amplifier started out a few years ago, when I stumbled across four NOS PL504 tubes by Philips (*Table 1*). It seemed like a bit of a challenge at the time to build an audio amp using TV line output pentodes.

DESIGN GOALS

Because it was not my intention to build a state-of-the-art audio amp with these tubes, I set out the following basic guidelines:

1. Class-A triode push-pull
2. little or no feedback
3. approximately 10W output

The choice of Class-A triode PP seemed to be a logical one; it is a fair compromise between "triode sound" and "pentode power." Because the speakers I use are fairly efficient, 10W/channel is more than enough to get the neighbors interested.

I wanted to predominantly use available components, first, to reduce total building cost, and second, to find out what an amp could sound like if I designed and built it with component restrictions in mind, rather than the inverse. Although I am not strictly antifeedback, I prefer to use it sparingly because I believe in many cases the cure is worse than the disease.

As in previous designs and modifications, I paid special attention to phase integrity[1]. Often circuits that are less accurate regarding phase and transfer time are also less musical in the sense of rhythm and sound-stage re-creation. I simulated the whole design with PSpice[2], using the models proposed by N. Koren[3] in his series of articles together with measured data from both 12SX7GT and PL504 to produce a simulation model for these two devices. *Table 2* lists both tube model parameters together with their respective Ip/Vp curves.

AMPLIFIER

Both the first gain stage and the phase inverter use a 12SX7GT. This tube (which the *E.U.V.*[4] lists as a 12.6V heater equivalent of the 6SN7GT) has a more musical sound than its slightly analytical noval counterpart (ECC82). I have noticed in the past that the combination of certain types of stages (e.g., common cathode and split-load inverter) has a more marked influence on the final sound than the use of different types of components in the same location, which I regard more like fine-tuning the design in a final stage.

The amp's input stage (*Fig. 1*) consists of a common-cathode amplifier, the first half of the 12SX7GT set at Ip = 6.5mA by R3, gain thus being about 16×. By adapting R5 and R6/R7 to the first stage plate voltage (about 100V DC), use of a coupling capacitor between the first and second stage was not necessary. The second half of the 12SX7GT is

ABOUT THE AUTHOR

After finishing classical studies (Latin and Greek), Mark Taverniers studied physics at university, followed by a degree in electronics. Previous jobs saw him active in professional audio and video as well as microelectronics. He is currently working on industrial automation processes. His main interests range from literature, philosophy, and classical music (listening and playing) to designing and modifying vacuum tube audio.

TABLE 1
PL504 REFERENCE DATA (MBLE DATA HANDBOOK PART IV ELECTRON TUBES SEPT. 1970)

Peak plate voltage	Vpp (max.)	7000	V
Cathode current	Ik (max.)	0.250	A
Plate dissipation	Wa (max.)	16	W
Heater current	If	0.3	A
Heater voltage	Vf	27	V
Plate voltage	Vp	50–7000	V
G2 voltage	Vg2	200	V
G1 voltage	Vg1	–10	V
Plate current	Ip	0.42	A
G2 current	Ig2	0.037	A
Anode to g1 capacitance	Cag1	1.75	PF
Grid1 to heater	Cg1f	0.2	PF

PL504 CONNECTIONS

1	G1
2	G1
3	Cathode + G3 (internally connected)
4	Heater
5	Heater
6	G2
7	G2
8	Cathode + G3 (internally connected)
9	Not connected
Top cap	Plate

set at Ip = 7.5mA. Both plate current and voltage values are well within limits for this tube and will ensure long and troublefree service. First and second stage gain combined (16 × 0.93 = 14.8) gave me enough voltage to drive the PL504s directly, setting full power input sensitivity at about 1.2V RMS.

The decoupling capacitors on the cathode of TU1a will influence low-frequency phase and, to a lesser degree, low-frequency distortion. I believe it is a good idea to use high quality small values to bypass the 330µF electrolytic.

In order to balance phase inverter output and reduce distortion, I used a 4.7k variable resistor in the cathode, thus allowing R6+R7 to equal R5. I used a 10kHz square wave to balance both outputs[5]. Since the output impedance at TU1b anode is larger than at its cathode[6], you might add a 10k resistor in series with C6//C7 to equalize bandwidth[7]. In this case, bandwidth at TU2 grid is about 0.07dB down at 100kHz compared to TU3 grid.

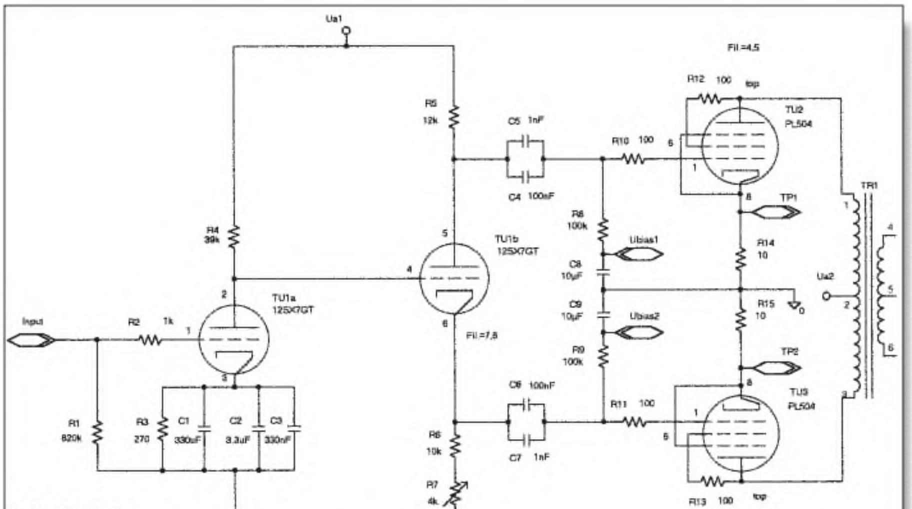


FIGURE 1:
Amplifier circuit.

Because R8 and R9 form an H.P. filter in combination with C4//C5 and C6//C7, respectively, they will influence low-frequency behavior and phase response. Increasing the value of R8/R9 will slightly influence the third harmonic distortion level. R10 and R11 are grid-stopper resistors, but in this case they don't need to be as large as their counterpart on TU1a. The quality of these components will have a marked influence on the sound of the amp, especially on low-level detail rendering.

Connecting g2 (TU2/TU3) to the plate via R12/R13 effectively switches the PL504 to triode mode and at the same time prevents H.F. oscillation. R14/R15 are connected to 2mm external test jacks to check and eventually adjust bias. Biasing each tube individually has a marked effect on distortion reduction.

With SPICE, I modeled the output transformer using the equivalent schematic of *Fig. 2*, and, on the basis of SPICE analysis results, I ordered custom-wound types. Some authors[8] suggest a somewhat higher plate-to-plate resistance than calculations indicate, thereby reducing overall THD levels but also sacrificing some output power; others suggest the opposite[9].

POWER SUPPLY

Power supply circuitry (*Fig. 3*) is quite straightforward. In view of my decision to use available (i.e., mostly salvaged) parts wherever possible, you will not be surprised to see five transformers. Obvi-

TABLE 2
VACUUM TUBE AND OUTPUT TRANSFORMER SPICE MODELS

```
.SUBCKT TRIODE A G K
+ PARAMS: LIP=1  LIF=3.7E-3  RAF=18E-3  RAS=1  CDO=0  RAP=4E-3
+ ERP=1.5
+ MU0=17.3  MUR=19E-3  EMC=9.6E-6  GCO=0  GCF=213E-6
+ CGA=3.9p  CGK=2.4p  CAK=0.7p
Elim  LI  0  VALUE  {PWR(LIMIT{V(A,K),0,1E6},{LIP})*{LIF}}
Egg   GG  0  VALUE  {V(G,K)-{CDO}}
Erpf  RP  0  VALUE  {1-PWR(LIMIT{-V(GG)*{RAF},0,0.999},{RAS})+LIMIT{V(GG),0,1E6}*{RAP}}
Egr  GR  0  VALUE  {LIMIT{V(GG),0,1E6}+LIMIT{(V(GG))*(1+V(GG)*{MUR}),0,-1E6}}
Eem  EM  0  VALUE  {LIMIT{V(A,K)+V(GR)*{MU0},0,1E6}}
Eep  EP  0  VALUE  {PWR(V(EM),ERP)*{EMC}*V(RP)}
Eel   EL  0  VALUE  {LIMIT{V(EP),0,V(LI)}}
Eld  LD  0  VALUE  {LIMIT{V(EP)-V(LI),0,1E6}}
Ga  A  K  VALUE  {V(EL)}
Egf  GF  0  VALUE  {PWR(LIMIT{V(G,K)-{GCO},0,1E6},1.5)*{GCF}}
Gg  G  K  VALUE  {(V(GF)+V(LD))}
CM1  G  K  {CGK}
CM2  A  G  {CGA}
CM3  A  K  {CAK}
RF1  A  0  1000MEG
RF2  G  0  1000MEG
RF3  K  0  1000MEG
.ENDS

.SUBCKT 12SX7GT A  G  K
XV1  A  G  K  TRIODE
+ PARAMS:  LIP= 1  LIF= 0.0037  RAF= 0.02  RAS= 2  CDO =0
+ RAP= 0.002  ERP= 1.4
+ MU0= 19.2642  MUR= 0.006167  EMC= 0.0000189
+ GCO= 0  GCF= 0.000213
+ CGA= 3.90E-12  CGK= 2.40E-12  CAK= 7.00E-13
.ENDS

.SUBCKT PL504 1 2 3 4 ;  P G1 C G2  ;6-4-99
+ PARAMS: MU=9 EX=1.3 KG1=220 KG2=4800 KP=20 KVB=15
+CCG=14P CPG1=1.75P CCP=12P RGI=1K
RE1  7 0  1MEG
E1   7 0  VALUE=
+{V(4,3)/KP*LOG(1+EXP((1/MU+V(2,3)/V(4,3))*KP))}
G1   1 3  VALUE={(PWR(V(7),EX)+PWRS(V(7),EX))/KG1*ATAN(V(1,3)/KVB)}
G2   4 3  VALUE={(EXP(EX*(LOG((V(4,3)/MU)+V(2,3)))))/KG2}
RCP  1 3  1G
C1   2 3  {CCG}    ; cathode to g1 capacitance
C2   1 2  {CPG1}  ; g1 to plate
C3   1 3  {CCP}   ; cathode plate capacitance
R1   2 5  {RGI}   ; grid current
D3   5 3  DX      ; grid current
.MODEL DX D(IS=1N RS=1 CJO=10PF TT=1N)
.ENDS

.SUBCKT PP-15W-1k5 1 2 3 4 5 6  ;20-01-00
*Primary
Lleak1   1    20  0.5mH
Lpri1     20  21   29H
Rpri1     21  2     20
Cpri1     1    2     1.20pF
Lleak2  2    22   0.5mH
Lpri2    22   23  29H
Rpri2    23    3  20
Cpri2    2      3  1.20pF
*Secondary
Lleak3  5  24  10uH
Lsec1  24  25  0.6H
Rsec1  25  6  0.8
Lleak4  4  27  10uH
Lsec2  27  28  0.3H
Rsec2  28  5  0.4
Kcoup   Lpri1  Lpri2   Lsec1  Lsec2   0.9985
.ENDS
```

Remark: the 12SXGT model uses the triode AGK model as a starting point
(to be found at http: \\www.duncanamps.simplenet.com/spice/valves)

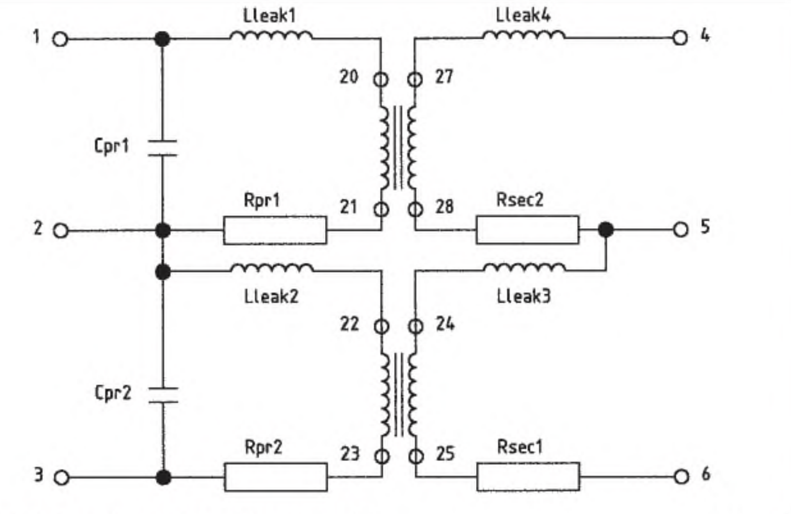


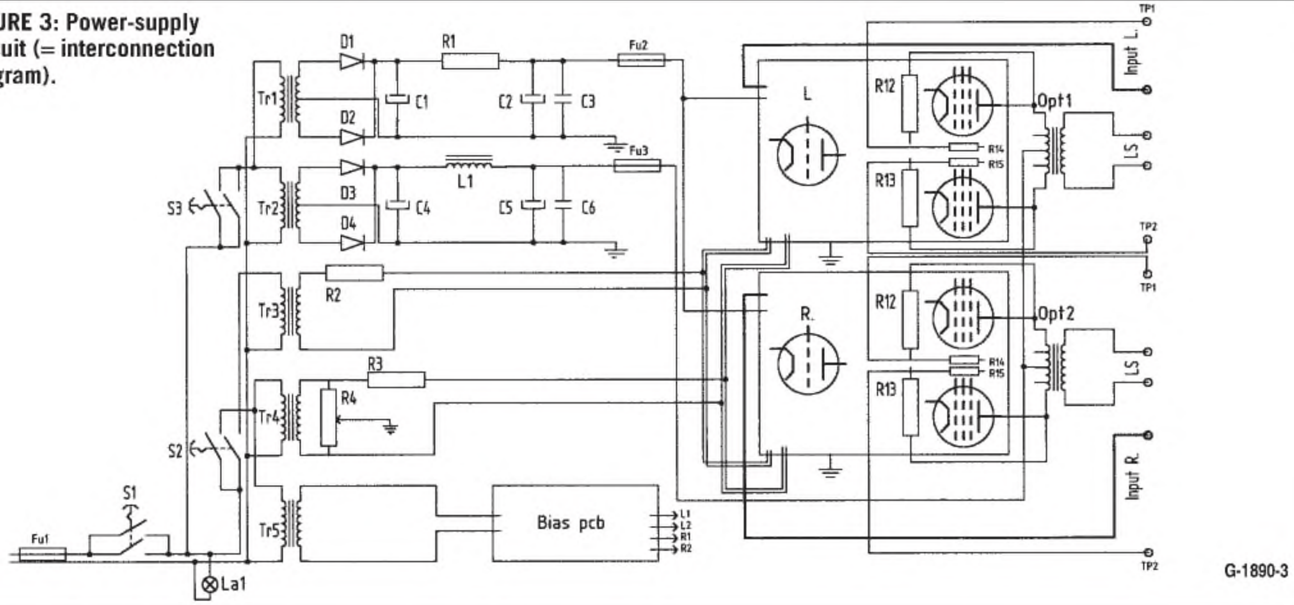
FIGURE 2: PL504 PP output transformer model. G-1890-2

ously a custom wound transformer caring for all voltages would do just as well. Tr1 gives 2 × 260V AC secondary, which, after rectification and filtering, gives 350V DC for both 12SX7GT tubes.

Tr2 secondary is wound for 2 × 115V AC at 120W, thus providing 200V DC at C5//C6. Both high-voltage sources are protected by 300mA fast fuses. Tr3 is used to supply 27V AC to the PL504 heaters. EL504 types will have the usual 6.3V heater filaments but are generally more expensive, and in my experience, somewhat harder to find.

Secondary voltage on this transformer was lowered slightly by way of R2 (use 5W or more). The same applies

FIGURE 3: Power-supply circuit (= interconnection diagram).



G-1890-3

to Tr4, which gives 12.6V AC for both 12SX7GTb heaters. Instead of feeding gain stage heaters with DC voltage to lower hum, I used R4 to balance Tr4 output voltage. I believe both ways have their distinct advantages.

Tr5 provides 20V AC to the bias board. Specs listed in the parts list (*Table 3*) are those of the available components I used. Obviously in the case of Tr5 you do not need 1.5A to supply four bias voltages! All grounds, including one side of input cable shielding, are tied to the central point at C1–C6.

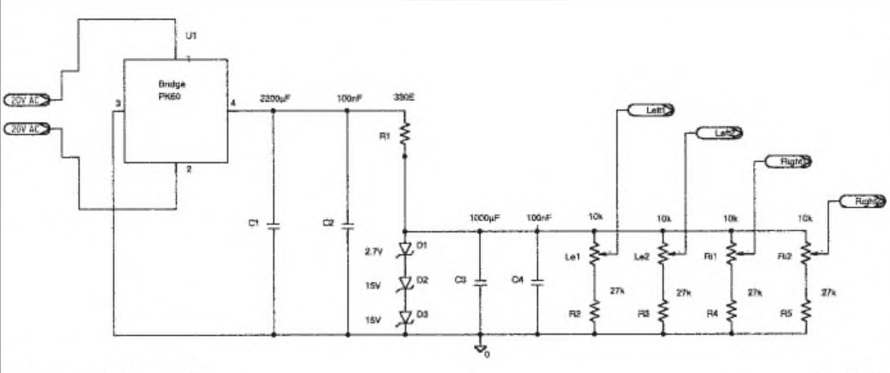


FIGURE 4: Bias voltage. G-1890-4

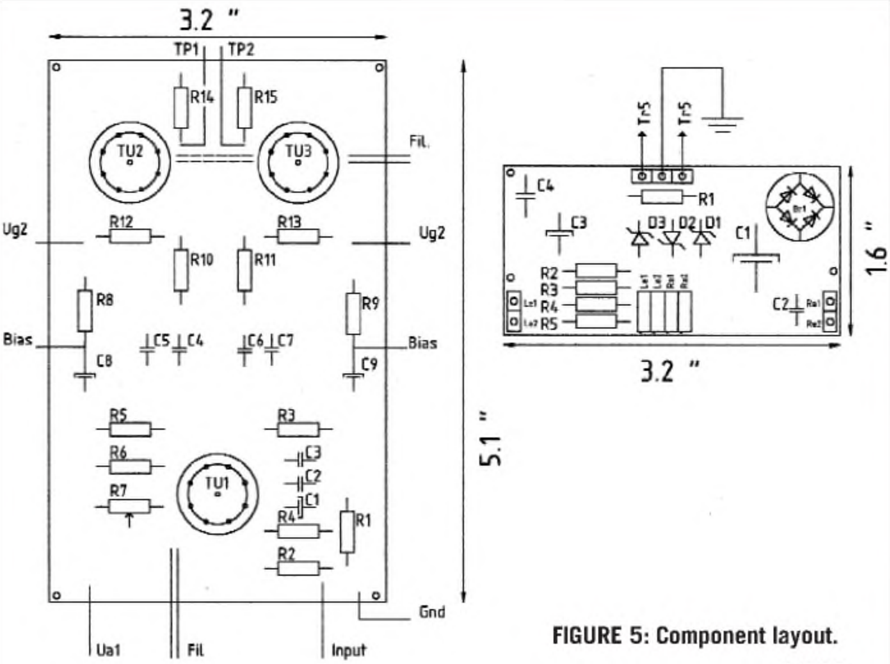


FIGURE 5: Component layout. G-1890-5

BIAS PCB

Bias voltage is rectified and filtered by U1 and C1//C2, respectively (*Fig. 4*), stabilized by means of a zener string D1–D3 (zeners connected the wrong way round due to negative bias voltage). After I added reservoir cap C3 and H.F. decoupling C4, four 10k multiturn pots provide individual bias voltage adjustment for each power tube.

CONSTRUCTION

I assembled the first working prototype on vector board (left and right channels and bias PCB) with all other components mounted on 18mm MDF board (*Fig. 5*). This had the distinct advantage of accessibility while prototyping (I did not need to turn the amplifier upside down to measure or change components). First, I thoroughly measured the unit with calculation results and SPICE analysis as a reference, and afterwards I allowed it to run in for two weeks using a pink-noise input signal.

Subsequent listening tests formed the basis for some minor modifications, and after six months of almost daily use I constructed a cabinet from 18mm MDF. Since heat generation is quite important, I paid special attention to ventilation on the bottom and top panels. Once again I carried out listening tests (*Table 4*) and finally ran the amp through a final testbench session.

PERFORMANCE

Two aspects of performance analysis should merit your attention here: measured data and subjective listening.

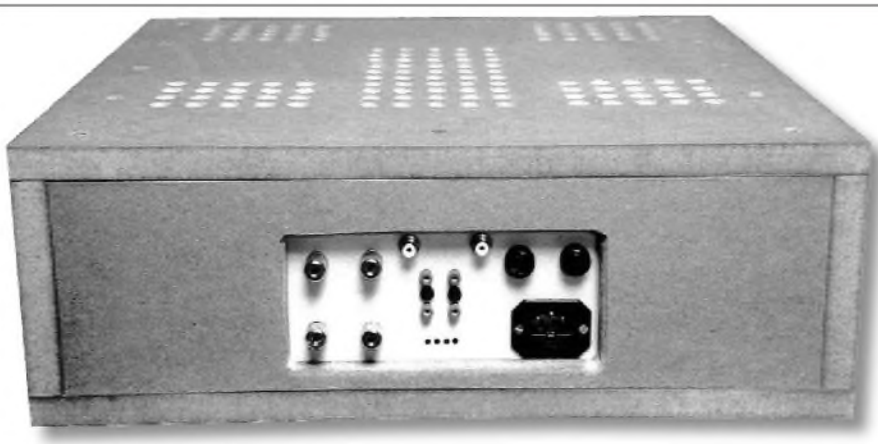


PHOTO 2: Back view showing connector panel. From left to right are 4mm gold-plated loudspeaker connectors, left and right cinch inputs (top), and 2mm test jacks connected to TP1 and TP2 on the amplifier boards. The right-hand side shows mains input connector (fused and filtered) and above it two separate fuseholders protecting both 12SX7GT and PL504 high voltages.

Test results are listed in *Table 5*, together with SPICE analysis predictions. SPICE model accuracy is proven by the close correlation between the two. I believe test results are quite satisfactory in view of the voluntary restrictions imposed and commented on earlier.

TABLE 3
PARTS LIST

AMPLIFIER BOARD		
R1	820k/0.5W	1%
R2	1k/1W	1%
R3	270E/5W	5%
R4	39k/2W	
R5	12k/5W	
R6	10k/5W	
R7	4.7k/5W	
R8/R9	100k/0.5W	
R10/R11/R12/R13	100E/0.5W	
R14/R15	10E/2W	
C1	330µF/25V	
C2	3.3µF/100V	MKT
C3	330nF/560V	MKT
C4, C6	100nF/630V	MKT
C5/C7	1nF/1000V	polyprop.
C8/C9	10µF/63V	
TU1 a/b	12SX7GT	Rogers
TU2/TU3	PL504	Philips
MAIN CHASSIS		
R1	1k5/5W	5%
R2	1Ω/7W	5%
R3	3.3Ω/10W	5%
R4	200Ω/10W	5%
C1/C2/C4/C5	270µF/400V DC	Sprague powerlytic
C3	680nF/400V DC	
C6	2.2µF/250V DC	
Tr.1	P = 220/S = 2× 260@100mA	
Tr.2	P = 115-0-115/S = 150-0-150/Psec. = 120W	
Tr.3	P = 220V/S = 2×25V@1.6A	
Tr.4	P = 220V/S = 12V@1.33A	
Tr.5	P = 220V/S = 2*10V@1.5A	
Opt.1/2	15W/Raa = 1k5/Rsec. = 5Ω	
S1/S2/S3	250V/1A bipolar switch	
La1	220V neon lamp	
D1/D2/D3/D4	1N5408	
L1	300mH/1A choke	
Fu1	2A/250V	Slow
Fu2/Fu3	300mA/500V	Fast
BIAS BOARD		
R1	330Ω/2W	5%
R2/R3/R4/R5	27k/0.5W	1%
Re1/Re2	10k/10T multi	1%
Li1/Li2		
C1	2200µF/40V	ITT
C2, C4	100nF/250V DC	MKT
C3	1000µF/40V	ROE
Dz1	2.7V/2W	Zener
Dz2/Dz3	15V/2W	Zener
Br1	PK60/2A	
MISCELLANEOUS		
18mm MDF board		
4mm loudspeaker connectors		
2mm test connectors		
Cinch input connectors		
Fuseholders		
IEC mains inlet/fuseholder		
Mounting material		

TABLE 4
REFERENCE RECORDINGS

SOLO INSTRUMENTAL			
– Bach	Sonatas/partitas for solo violin	A.Grumiaux	Philips 6768-017
– Beethoven	Late piano sonatas	M.Pollini	DG 2740-166
CHAMBER MUSIC			
– Mozart	String quintets 1-6	Grumiaux trio	Philips 6747-017
– Poulenc	Chamber music (compl.)	Menuhin-Debost	EMI 2C165-12519/22
ORCHESTRAL			
– Mahler	Symphony V	NPO/Barbirolli	HMV SLS785
– Beethoven	Symphony IV	BPO/Karajan	DG 2531-3301
– Bach	Harpsichord concertos 1-7	English Concert/Pinnock	DG 2723-077
VOCAL			
– Puccini	Tosca	Callas/De Sabata	HMV SLS 825
– Britten	Peter Grimes	Pears/Britten	Decca SXL 2150-2

TABLE 5
COMPARISON OF MEASURED DATA AND SPICE ANALYSIS PREDICTIONS

	MEASURED	SPICE	UNIT	TEST POINT	TEST CONDITION
Bias voltage	0.745	0.750	V DC	R14/R15	
PL504 H.V.	191.40	200.00	V DC	C5	
12SX7GT H.V.	358.80	350.00	V DC	C2	
Full power bandwidth	23.6-89.2E3	26-95E3	Hz	Left OPT	1kHz sine/8Ω load
	25.2-88.9E3	26-95E3	Hz	Right OPT	1kHz sine/8Ω load
Output noise	4.5E-3 (i.e., 4.5mV)		V RMS	Left OPT	1kHz sine/8Ω load*
	5.2E-3 (i.e., 5.2mV)		V RMS	Right OPT	1kHz sine/8Ω load*
Slew rate	12.50	15	V/µs	Left OPT	10kHz square/Uimax
	12.20	15	V/µs	Right OPT	10kHz square/Uimax
Transfer time	1.00		µs	Left OPT	10kHz sine Po = 1W
	0.95		µs	Right OPT	10kHz sine Po = 1W
Maximum output voltage	8.22	9.00	V RMS	Left OPT	1kHz sine/8Ω load
	8.05	9.00	V RMS	Right OPT	1kHz sine/8Ω load
Input sensitivity for Pmax	1.30	1.20	V RMS	Left OPT	1kHz sine/8Ω load
	1.39	1.20	V RMS	Right OPT	1kHz sine/8Ω load
S/N ratio	–74.15		dB	Left OPT	1kHz sine/Po = 1W
	–75.70		dB	Right OPT	1kHz sine/Po = 1W

*Input shorted

Most important, however, are the listening results: low-frequency rendering is quite good in spite of obvious limits imposed by the output transformer bandwidth. The midrange is excellent; voices are rendered without strain and well detached from the background. No listening fatigue (closely correlated to the distribution and magnitude of harmonics on any given fundamental) occurs even after several hours; this is often the case with little amps wishing to sound large. The sound stage is well defined, inasmuch as the speakers seem to vanish, which makes room for a realistically large and deep sound stage where each instrument or group of instruments is exactly located and separated.

The low-level detail is good; a very faint hum is audible when you listen without input signal directly against the speaker cones. Melodic lines are distinct and easily traceable even in

TEST EQUIPMENT USED

—Hameg HM 203-7 scope
—Daystrom AW-1U audio power meter
—Heathkit IM-38 VTVM
—Tektronics DMM 254 multimeter
—Thurlby TG 230 audio generator
—ATS A-1.2 distortion analyzer
—Test bench power supply

very dense orchestral passages. Output power restrictions become obvious at high levels because sound becomes congested, and an onset of distortion is audible in peak passages.

I must insist that these listening levels were not realistic in the sense that with this combination of speakers and listening room they were not even comfortable. At about the same power output as my reference amps (four 2A3 SE driving woofer and tweeter separately on each speaker), the amp sounds less refined, somewhat less nice, but distinctively more forward.

CONCLUSION

Summing up, I cannot be anything but positive about this project. A modest investment (about $300) has brought me new and unsuspected listening pleasures and also some new insight with regard to tube amplifiers. At the outset of this project (in spite of some articles advocating the use of TV deflection tubes in audio[10]), I was skeptical. But after completion, I am convinced they form an as yet undiscovered terrain worthy of our attention.

Generally, these tubes are priced lower than "dedicated" audio tubes of comparable specs—the PL504s cost $2.50 each. In view of the fact that I used salvaged/second-hand parts wherever possible, the results are even quite excellent, and with its basic "proven" topology, this amp will serve as a comparison when I design different circuitry in the future.

As always you must be able to compromise; you cannot have your cake and eat it too. ❖

REFERENCES

1. D. Manley, *The Vacuum Tube Logic Book*, 2nd ed. 1994, p. 48.

2. Pspice version 7.1 Microsim Corp., 20 Fairbanks, Irvine, CA 92718.

3. N. Koren, "Improved VT Models for SPICE Simulation," *Glass Audio* 5/96, p. 18, and "Errata and More on Pspice Tube Models," Letters, *Glass Audio* 6/96, p. 62.

4. *Electronic Universal Vademecum*, P. Mikolajczyk and B. Paszkowski, 2nd edition Warschaw 1994.

5. J. Marshall, "The Importance of Balance in Push-Pull Amplifiers," *Glass Audio* 4/94, p. 16.

6. Menno van der Veen, *Het VanderVeen buizenbouwboek* 1994, p. 4-4.

7. A. Kimmel, "Examining Phase Inverters," Letters, *Glass Audio* 2/90, p. 47.

8. J. Kuiper, "A Practical Monotriode Amplifier," *Audio & Techniek* no. 6, p. 25.

9. J. Hiraga, *Initiation aux amplis à tubes*, 2nd edition Eyrolles, p. 96.

10. R. Berglund, "Surplus TV Tubes for Audio," Glass Shard, *Glass Audio* 2/95, p. 52.

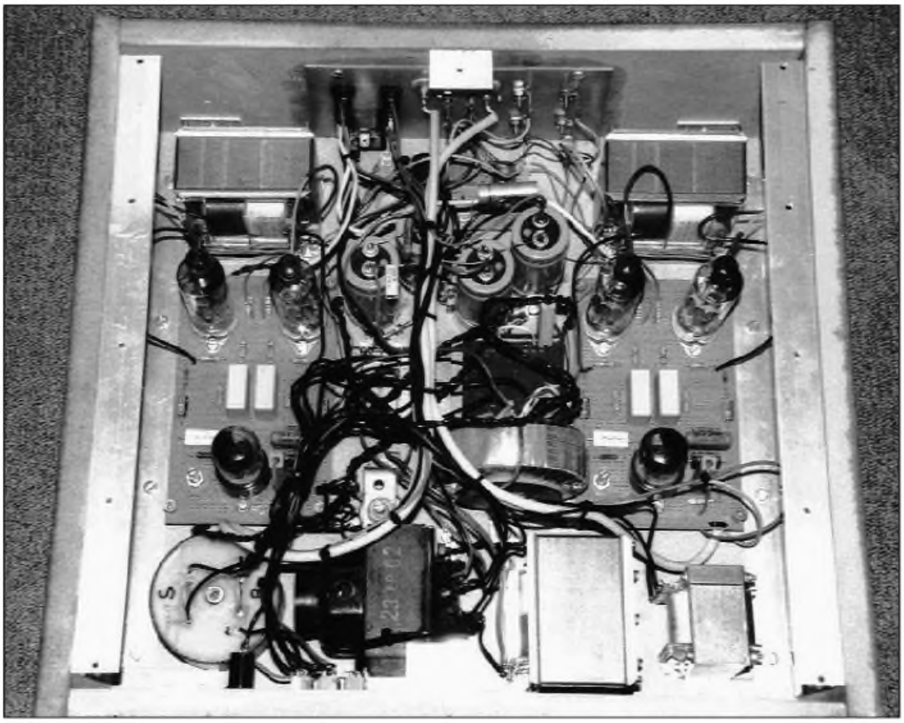


PHOTO 3: Top view showing output transformers (top left and right corners). In front of these, the amplifier boards are constructed on vector board using point-to-point wiring.

The Cadillac 16

This is one you may have been waiting for: a quick, inexpensive speaker project that promises to improve your system's sound. **By Millard Johnson**

Two of my favorite pastimes are building speakers and building tube amplifiers. I recently completed a single-ended amplifier project that took a large part of my spare time for several months, and I was not particularly looking for another project just yet. But you never know what opportunity is lurking around the corner!

BARGAIN DRIVERS

As I was looking over the April flyer from Parts Express, one ad caught my eye: a 6″ subwoofer for $2.49! Made for high-end GM sound systems, this speaker is named the Cadillac 16. It has a frequency range of 100–800Hz. I spoke to Mark in sales at Parts Express and they had 800 of the 6″ Cadillac speakers—the shipping costs more than the speaker! Anyone who wishes to order some should use the part #269-472 and ask for Mark.

The specifications stated in the flyer were not those of a subwoofer, but they might work as a lower midrange in a four-way system, which I love because you can use specially designed drivers for different frequency ranges. Up to this point, I had a very satisfactory speaker system that included Morel MDT 33 tweeters, Morel/MSD 56 2 1/8 midrange, Audax 2025 TSN 8″ lower midrange, and an Isobarik subwoofer using 10″ drivers. Crossover frequencies are 4000-800-100Hz. This all worked very well except it is difficult to get the 8″ woofer and the subwoofer to balance out—even with the help of an Audio Control electronic crossover. I always thought that if the lower midrange had very little output below 100Hz that it would be easier to get an even response.

I spent some time brainstorming this design; then it hit me like a thunderbolt! I warmed up my computer and devised a simple-to-build enclosure 8¼″ × 8¼″ × 4′ 9″ including the base. The f_3 is 130Hz, Q_{TC} is 0.643, and net sensitivity is well over 100dB. It works well with any amplifier and is very kind to tube amplifiers (*Photo 1*).

PHOTO 1: The author with finished system.

One of the most fascinating features of this 6″ speaker is the response curve (*Fig. 1*). It is like a camel's hump with a peak around 350Hz. In fact, the response curve is about the same as it would be if you used the usual 12dB capacitor and inductor crossover. When I built this speaker system the crossover was the easiest part of the project.

I used the midrange-tweeter crossover without change and connected the Cadillac directly to its own monoblock pp 6550 amp (*Fig. 2*). The midrange/tweeter is connected via its own crossover to its own monoblock amp. The subwoofer amplifier is my version of the Altec 1569. This unit uses four EL34s and is rated at 80W.

Any new amplifier or speaker needs

ABOUT THE AUTHOR

Millard Johnson is 62 and is "sort of" retired. Being in the rental business, he is not an engineer. However, he does hold an extra class ham radio license and has been building and collecting tube amplifiers and speakers for over 40 years.

TABLE 1
CUSTOM CLOSED-BOX DESIGN

Shape: Prism, square
$V_B = 0.962ft^3$
$Q_{TC} = 0.643$
QL = 7
$f_3 = 130.8Hz$
Fill = heavy

DRIVER PROPERTIES
Type: Standard one-way driver
Company: GM

NO. OF DRIVERS = 6—REG.
Mounting = Standard
Wiring = Series-Parallel
$f_S = 58Hz$
$Q_{MS} = 10.3$
$V_{AS} = 0.65ft^3$ ($3.9ft^3$ total)
$S_D = 23.72\ in^2$ ($142\ in^2$)
$Q_{ES} = 0.41$
$Re = 2.89\Omega$
Le = 0.6mH (.4mH)
$Z = 3.2\Omega$
Pe = 35W (210W)

a break-in period. Some of them really sound bad when first turned on so a little patience is required. I have a friend in Laramie, Lynn Hamblin, who has the best ears in the Rocky Mountains! I always have him over to listen to my latest project.

I was just finishing the speakers, when Lynn dropped by to listen to them. He sat down in his favorite listening spot, listened for a few minutes, and said, “One or more of the speakers is wired out of phase.” I told Lynn it would be difficult to remove the speakers to check the wiring, but he said that is not the way to do it.

Solder a wire on each end of a D battery, disconnect the amplifier and touch the battery wires to the speaker wires. All the cones must move in the same direction—in or out doesn’t make any difference. Sure enough, one was different. I removed the speaker from the cabinet and reversed the wires, and the change of sound was noticeable.

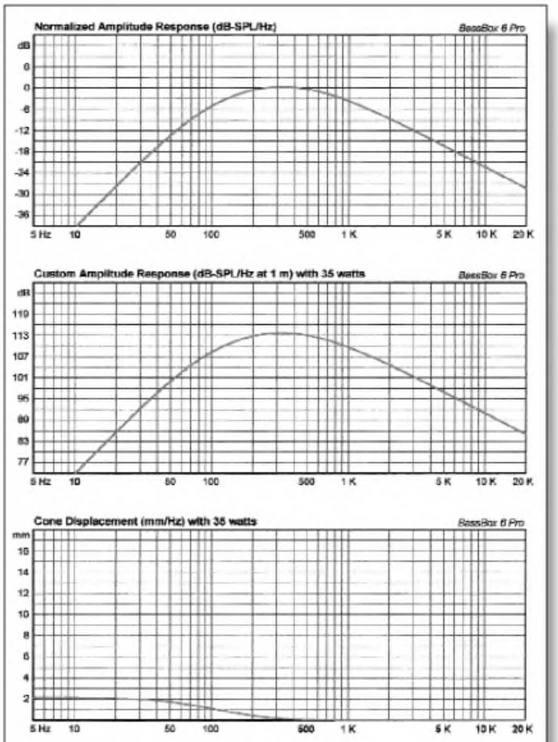


FIGURE 1: System response.

I used the midrange and tweeters from my last system. I highly recommend the Morel midrange and tweeters. However, if you have a mini monitor with a good midrange and tweeter that will fit in the cabinet (6″ diameter or less), you might be able to use them in this system. Including your existing crossover!

CONSTRUCTION

When you build your version of the Cadillac 16, use 1⅛″ particleboard (*Fig. 3*). Ask for 1⅛″ particleboard or oak stair tread material, which is rounded on one side and is about 12″ wide. I tried something on this project I had not done before. I asked the lumber yard to cut all the pieces, which cost an extra $5, but you should have seen all the miscut pieces on their floor!

Don’t use glue; use construction adhesive, which you spread with a caulking gun. It fills gaps up to one-quarter of an inch.

The results speak for themselves. This is the best speaker system I have ever heard...period! It has a sound stage you think you can walk into. The sound

lights up my listening room. The detail and location of the performers are very accurate. You do not have instruments or lips 12′ wide!

Time is important. I bought 14 speakers. With shipping, the price was $56.10. This is a bargain of the century! Mark at Parts Express informed me they have only a few hundred left. ❖

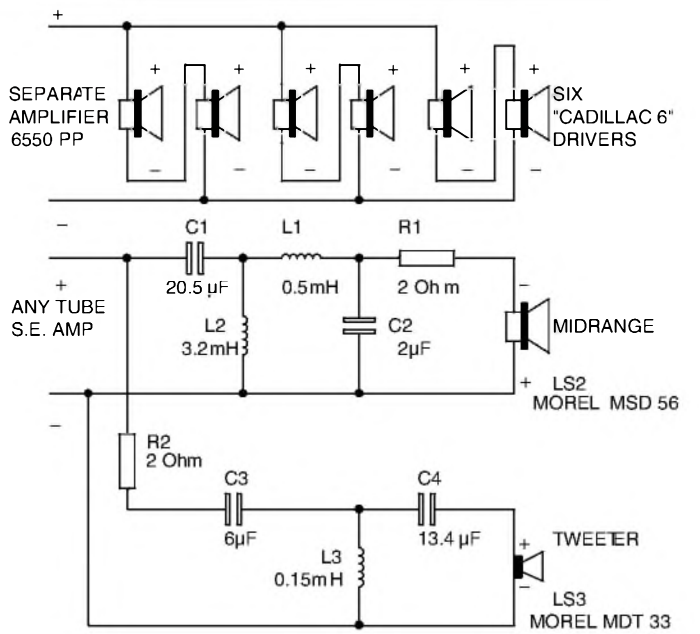


FIGURE 2: Wiring diagram.

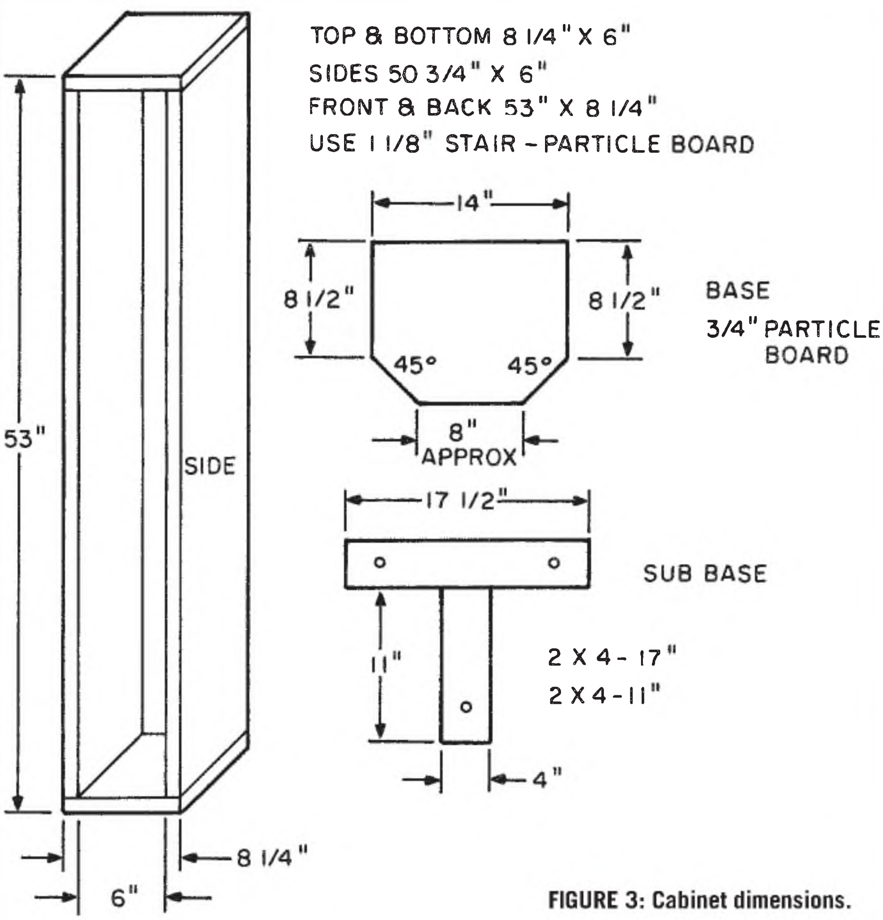


FIGURE 3: Cabinet dimensions.

The Ultra Fidelity Computer Sound System, Part 5

Now it's time to assemble and hook up the system. So grab your soldering iron and other tools and soon you'll be enjoying REAL computer sound. **By R.K. Stonjek**

The first job is to secure all the parts for this project (see parts list). The PCB must be printed as described in Part 4. Inspect the printed (and tinned) PCB for any solder bridges or tracks not properly printed. The Ultra Fidelity name should be reasonably clearly printed. Script text of this kind gives a good indication of the quality of printing.

Make sure all of the holes are drilled—try out some components to make sure the holes are the right size. The amplifiers, regulators, and 10,000µF capacitors all require larger holes.

PARTS SELECTION

To determine whether C8 and C9 are required, measure the DC output of your sound card with a 47k load (computer on). It may be up to a volt before gradually settling down to zero, which is OK. To be doubly sure, use a DMM (digital multimeter) to check the resistance of the output stage (computer off). It should be quite high. If the resistance changes while you're measuring, then the output capacitor is probably charging. Again, that's OK.

If you measure a DC output of greater than 100mV and the output stage has a low impedance (a few hundred ohms), then you'll need to include C8 and C9. Otherwise, reduce the required number of 10N polystyrene capacitors from 20 to 18 and 2.2µF MKT capacitors from ten to eight.

If you decide to add a selector switch on the input (for instance, to take the signal directly from the CD ROM headphone output for better-quality CD sound), then you'll need an appropriate two-pole rotary switch. If you are not using C8 and C9, then omit the R7s from the PCB and place the 47k resistors directly over each of the input leads at the selector switch instead.

Once you have the parts, you'll need to "select" the best 470nF MKTs. Measure each one using a capacitor tester (most DMMs have such a thing. If not, borrow one). Three sets of matching caps are required.

Capacitors C16 and C17 need to be matched, but can be above 470nF. The two C20s and C22s should be close to 470nF, but it is more important that all four should be close together. If you have two matching pairs, but the capacitance of each pair is different, then make one pair the C20s and the other the C22s. C19 can be a little more than 470nF (up to 500nF), but this isn't as critical. C6 and C7 can be up to ±50%, so any leftovers will do.

Make sure the resistors are 1% tolerance metal-film types except for R4, 5, and 6, which are 5W wirewound (preferably 5% types).

You can download an easy-to-follow color-coded component guide from http:\\www.audioXpress.com on the *audioXpress* magazine page. The component guide is 1.3MB in size. Print this file from the internet explorer at as high a resolution as you can.

ASSEMBLY

Start assembly (*Photo 17*) by soldering the links (nine). Next do the resistors—all except R4, 5, 6, and 28—and then the ICs. Note the orientation. All ICs face the same way except for IC4, which faces the other way.

Now place R28 right over the top of IC4. Make sure that the resistor leads do not touch any of the ICs' legs.

You can insert the low-profile MKT, LCR, and ceramic capacitors and solder in the fuse clips. The 6A fast-blow (regular) fuse can go in as well.

If you have opted for mica washers, then you will need to use heatsink paste when assembling S1, 3, and 4. Loosely screw S4 together as shown in *Fig. 33*. Check that it fits neatly onto the PCB, then solder S4 in place. Once done, tighten the nut. Then solder in R6 and C5 (this capacitor must be oriented correctly). Repeat the procedure with S3, then R5 and C4.

Screw S1, the triac, flat onto the PCB with its heatsink and washer as illustrated in *Fig. 33*. You can now solder C1 in place. You must firmly bolt the

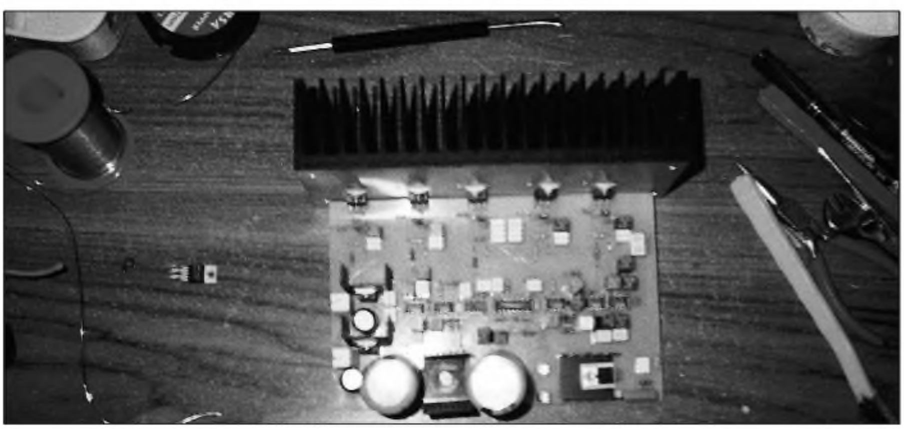
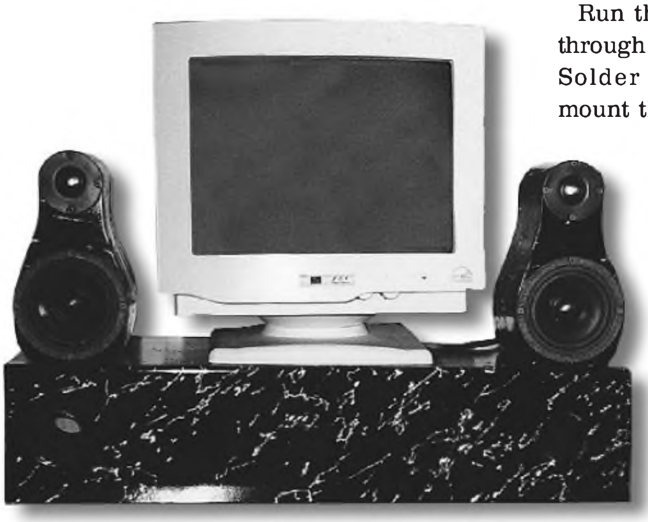


PHOTO 17: Assembling the board.

heatsink onto S2 before soldering it in place. Use heatsink paste between the heatsink and S2.

Finally, solder C2 and C3. Glue a rubber foot—like those used on metal cases—to the top of them (*Fig. 34*). You can use double-sided sticky tape for this. Double-check the finished board for solder bridges, especially around S1, 3, and 4, and the ICs.

Loosely place the amplifiers (IC3) on the board and line up the heatsink. The spacing of the fins varies between manufacturers, so some of the amp chips may need to lean one way or the other to fit. You need to drill holes between the heatsink's fins. Calculate the required height of the hole (*Fig. 34*), which will vary with the exact size of capacitors C2 and 3, as these also vary between manufacturers.



Completed Speaker System with Monitor.

DRILLING AND WIRING

Once done, drill the holes that will secure the heatsink to the case. Now you can mount and solder the amps. Note that the bolt can go in from either direction.

Drill the holes for all of the case parts: the mains IEC socket, a hole for the USB connection, and a hole for the transformer bolt. Use rubber grommets where cables pass through the case.

If you are building the advanced (round) subbass, then you will be running the attenuator lead out via RCA sockets. Otherwise, drill a hole for the attenuator. Solder the attenuation resistors directly onto the attenuator (wiring shown in Fig. 31 and *Photo 18*). For the round sub, run the dual-shielded cable through the back of the case, near where the speaker wires exit. Make sure you seal this hole.

Run the dual-shielded cable through to the front of the sub. Solder it to the attenuator, mount the attenuator onto the attenuator disk (Fig. 24), and secure it to the front of the subwoofer.

For the speaker leads, I used an external terminal strip and just ran the wires through a grommet. You might choose speaker terminals. You can run the audio input cable either straight in through a suitable hole or via a 3.5mm plug and socket.

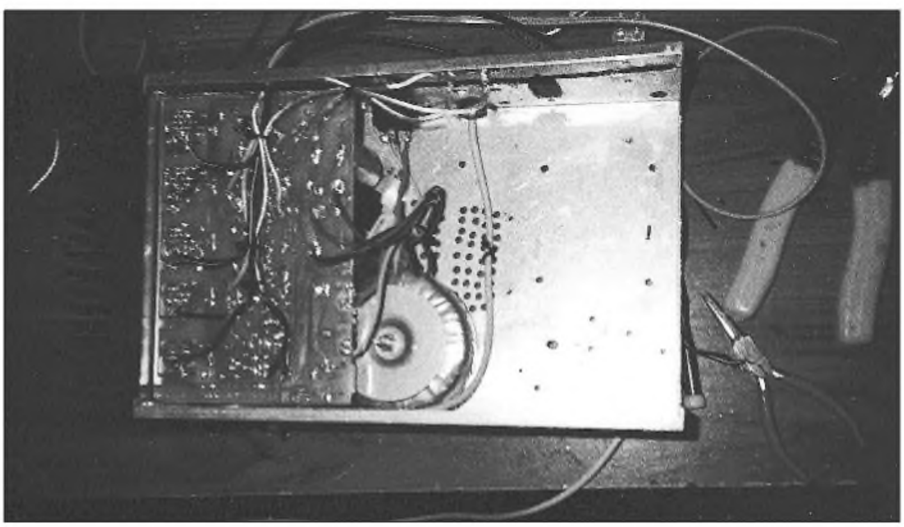


PHOTO 18: Wiring.

You need to cut the lid of the case at one end to accommodate the heatsink. This will vary with the type of case you use (see parts list). I cut off the entire end of the overlapping cover. You may add extra screws to the cover to hold it in place.

CONNECTIONS

Once done, carefully screw the heatsink in place. Be careful not to bend the amplifier's legs too many times, or they will eventually break!

Solder all the cables directly to the PCB—no pins are necessary! Mount the mains socket, earth lug, and transformer. Connect the IEC socket's earth pin to the earth lug (on the chassis) and then to the PCB via R4 (1R). Make this connection reasonably tight to help hold the PCB in place.

Connect the mains active to the PCB (*Fig. 34*). The transformer's primary connections can go either way around. Connect one to the mains neutral, and the other to the PCB at a point marked "Tr."

Connect the secondary actives to "~" and the zero volt (center tap) to "C." If your transformer has four secondary wires, be sure to connect the correct two together for zero volts. The color codes should be marked on the transformer, on the transformer packaging, or in the supplier's catalog. Don't guess if you're not sure—find out!

Now you should make the USB connection. *Figure 34* shows the plug configuration. It is best to use a complete cable and cut it open about halfway. Identify the 0V and 5V. Connect them to the PCB as marked. Note that you can still use this USB, keyboard, or PS/2 connector as usual; the amplifier draws a negligible amount of current and doesn't interfere with the data lines.

Once done, plug the USB into the computer and switch the computer on. Test to make sure that 5V appears on the PCB at the point marked "–" and "+." Now switch off the computer and carefully plug the amplifier into the mains. The small leakage voltage should be enough to test the polarity of the various lines before any serious voltage flows.

Check the voltage at the transformer secondary. There should be at least 2 or 3V present (relative to earth). Next check the adjacent line ending with a "+" (between the two "~" and the "C"). You should be able to read a few positive volts. The other power rail should read a similar negative voltage.

If these voltages don't check out—for instance, if one is very different from the other, if its polarity is wrong, or if the voltage on one line is zero—carefully recheck your work before proceeding. In particular, check that S3 and S4 have not been swapped, that the IC's orientation is correct, and that there are no solder bridges on the PCB.

Next check the two parallel tracks that run underneath the ICs. The line closest to the transformer should be slightly positive, the other slightly negative. The positive line might be a volt higher than the negative, which is OK.

TROUBLESHOOTING

If everything passes, then you can plug in the USB and turn the computer on. Make sure you can reach the mains power switch if you need to suddenly cut the power (don't suddenly turn off the computer). Recheck the voltages as above, but this time expect ±25V and ±15V, respectively.

If the 15V rail is low, say 3V, this is a sure indication that one of the ICs is mounted the wrong way. Luckily, I remembered the current-limiting resistors (R6 and 7), or ICs would be burning! The Burr-Brown ICs will not be affected by your little mistake. If the 25V rail is low, then there may be a problem with one of the amplifiers.

Check the amplifier's output (near the main heatsink, marked "T" for tweeter, "B" for bass, and "S" for subbass). They may be quietly oscillating, which is OK, but there shouldn't be any DC voltage showing. If there is—particularly if only one is doing it—then you should be suspicious.

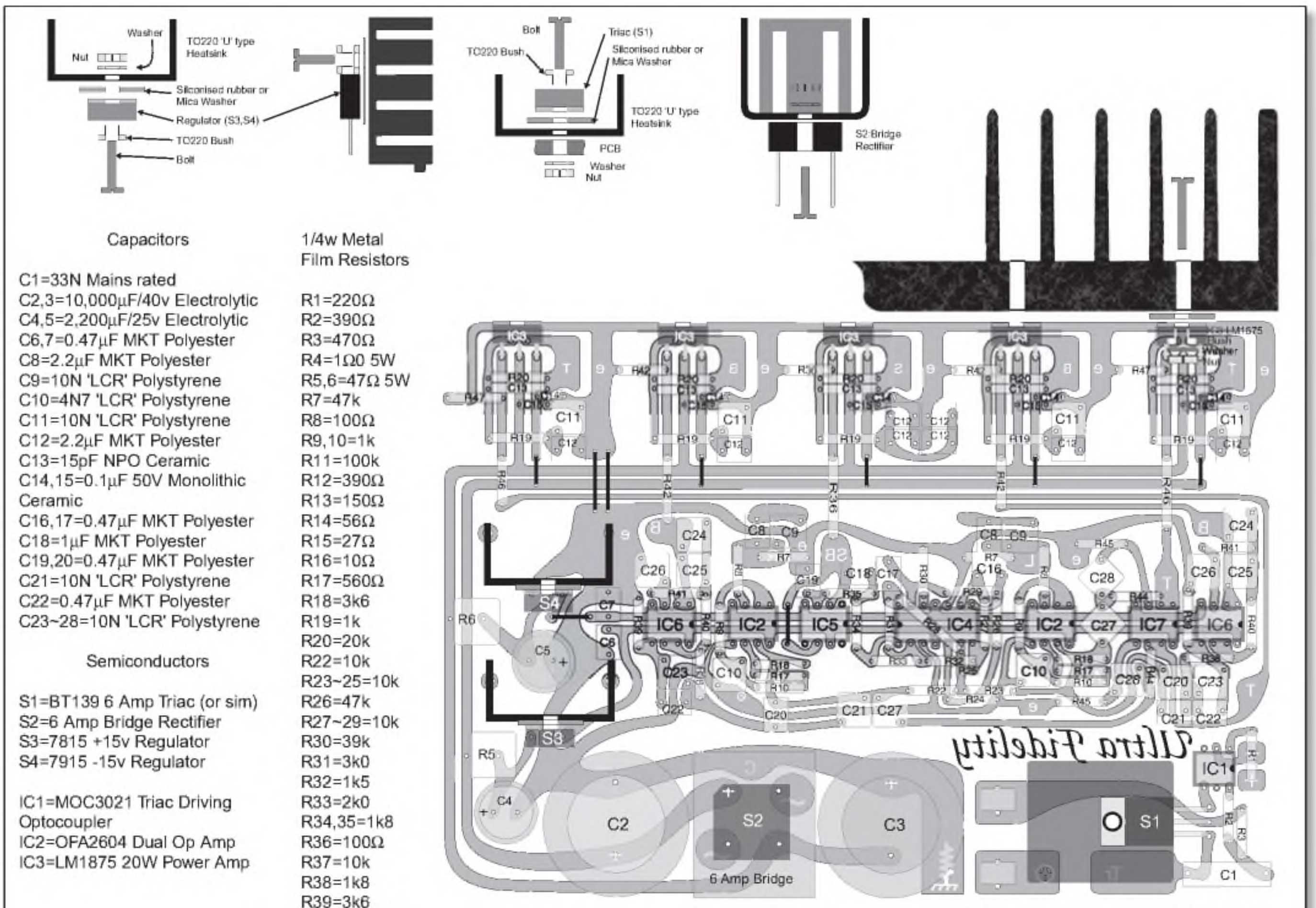
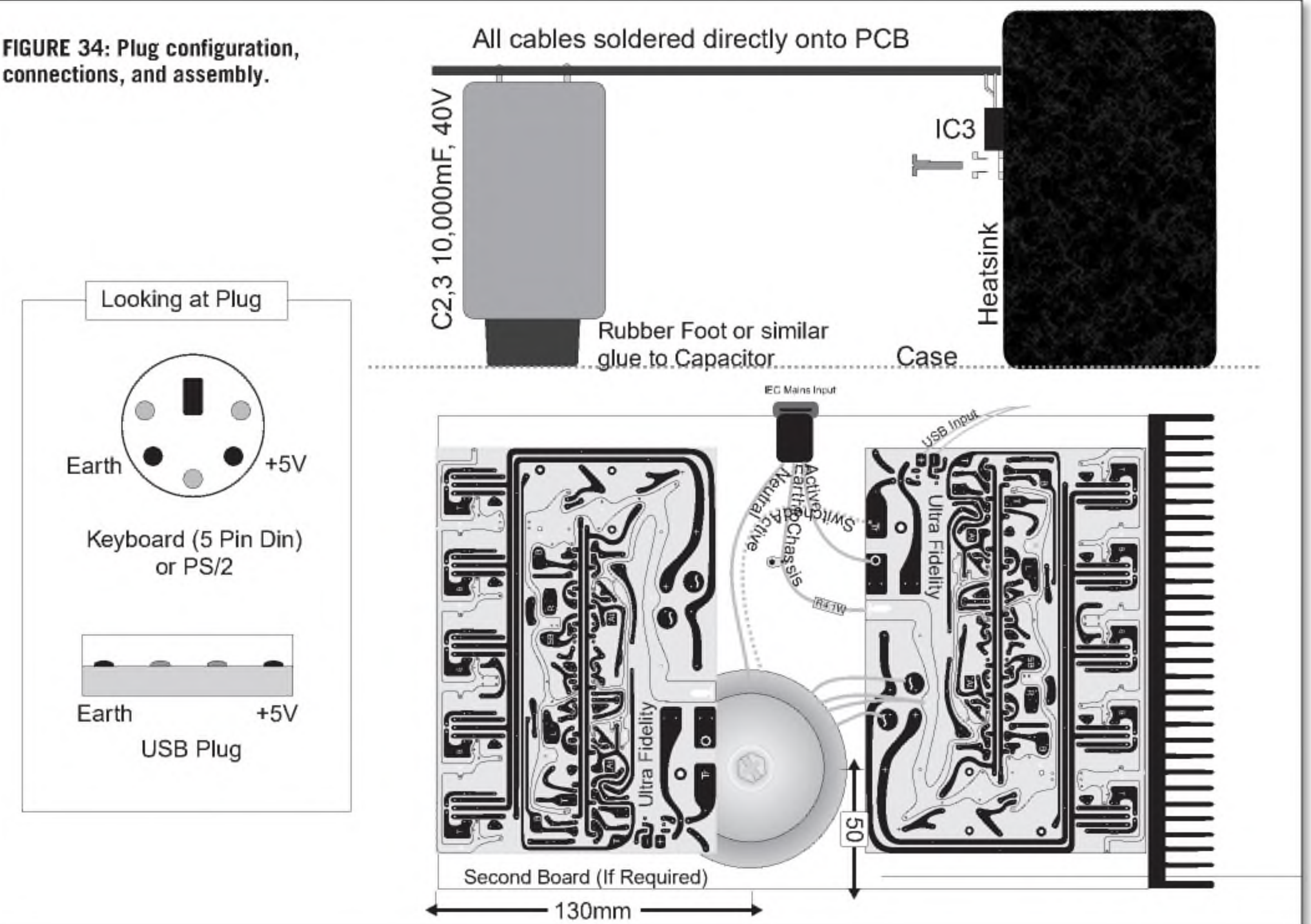


FIGURE 33: Component placement.

FIGURE 34: Plug configuration, connections, and assembly.



If the unit did not switch on, then switch everything off and unplug the mains. Change the transformer's switched line to the mains plug (active). This bypasses the solid-state switch.

Now try it and see whether the voltages are right. If so, then unplug everything and change the transformer's active line back to the switched output ("Tr"). Make sure the USB line is giving 5V at the correct polarity. Test the active line with the computer on and off. If the voltage rises only slightly, then the optocoupler is not switching properly.

If you have used a different optocoupler (it must be a triac driver), then the "on" state may require a larger forward current. R1 is responsible for this.

If you can't calculate the correct value for R1, then, with everything switched off and unplugged, solder a couple of temporary pins over R1 (on the copper side). Clip a resistance substitution box onto the pins and set the resistance high. Then switch on and monitor the voltage at the transformer while increasing the current to the optocoupler. You can solder the extra resistor onto the copper side of the PCB when you find the correct value.

Make sure you have used a mains-rated capacitor for C1. If C1 goes, the triac will probably go as well. Check the fuse and the orientation of IC1.

TESTING THE SYSTEM

Next, connect the attenuator marked "ATT" on the PCB. The earth run is marked with an "e." Then connect the input and plug it into the computer. To test the system, download the sine-wave test files from the *audioXpress* website at http://www.audioXpress.com on the *audioXpress* magazine page.

This download contains 11 ten-second test tones in MP3 format (totaling 1.6MB). The files are named: 40Hz-6dB; 80Hz-6dB; 250Hz-6dB; 500Hz_0dB; 500Hz-3dB; 500Hz-6dB; 500Hz-9dB; 1kHz-6dB; 5kHz-6dB; 10kHz-6dB, and "Pink Noise."

Power up everything and run the 500Hz-6dB signal through the system. You should get a good clean signal through the bass amp on either side.

The amplitude should be the same for both channels. Check the action of the attenuator. Each setting produces approximately double or half the previous voltage.

Check the treble and subbass amps with the 5kHz and 80Hz tones. If all checks out, then you can wire in the speakers.

The "T," "B," and "S" near the amplifiers and the accompanying "e" mark the speaker outputs. Note that the tweeter is not wired out of phase as with the passive setup described in Part 2.

With the speakers connected and the system switched on (attenuator at the lowest setting), you should hear only silence until you hear a music file, CD, or system sound. If you're getting a lot of extraneous noise, you can determine whether the noise is coming from your amplifier or the computer by taking the output directly from your CD-ROM (the headphone output). If this fixes the noise, then you have a noisy sound card, and it's time to upgrade.

If you're still getting noise, check the system with the inputs shorted (all three connectors on the 3.5mm plug together). This should cancel the noise. If the noise is still there, then you have noise in the amplifier somewhere. Check the earth lead on the connector cable and the lead to the attenuator (you should be able to test with the attenuator unplugged). Failing that, search for dry solder joints and loose wires.

Next try the 500Hz, –9dB, –6dB, –3dB, and 0dB test tones. There is quite a lot of variation between sound cards. Typically, a 0dB signal will distort quite badly because the sound card has more gain than it can actually deliver. This distortion is heard as an additional "buzz" over the correct signal.

Once done, go back to the –6dB signal and adjust the mixer settings until the buzz goes away and you hear a smooth signal. Sometimes the "Wave" setting, the master volume, or both, needs to be backed off a bit.

The amplifier can actually take three times the maximum output (around 10V) of the sound card (typically 0.6 to 3.5V), so distortion is nearly always caused by the sound card. Run the 40, 80, 250, 500Hz, 1kHz, 5, and 10kHz test tones. To check individual right and left, change the balance on the computer's mixer settings (usually in the system tray).

The final job is to screw on the lid. Place a spacer of some kind between the PCB and the lid—scrap rubber or polystyrene—as long as it is flexible and non-conductive.

That's it! Try some MP3s, WAVs, CDs, games, midi files, or spoil yourself with a DVD movie. I hope you get as much pleasure from your computer sound system as I do! ❖

PARTS LIST

RESISTORS

5W, WIRE WOUND

VALUE	QUANTITY	
1Ω	1	R4
4Ω7	2	R21
47Ω	2	R5, R6
Total	5	

¼W, METAL FILM

VALUE	QUANTITY	
10Ω	2	R16
27Ω	2	R15
56Ω	2	R14
100Ω	5	R8, R36, R42
150Ω	2	R13
220Ω	1	R1
390Ω	3	R2, R12
470Ω	1	R3
560Ω	2	R17
820Ω	2	R46
1k0	9	R9, R10, R19
1k5	3	R32, R47
1k8	4	R34, R35, R38
2k0	1	R33
2k7	2	R44
3k0	1	R31
3k6	4	R18, R39
4k3	4	R40, R41
5k6	2	R45
10k	10	R22–R25, R27–R29, R37, R43
20k	5	R20
39k	1	R30
47k	3	R7, R26
100k	2	R11
Total	73	

CAPACITORS

VALUE	QUANTITY	
NPO Ceramics		
15pF/50V	5	C13
Monolithic Ceramics		
0.1µF/50V	10	C14, C15
"LCR" Polystyrene		
4N7	2	C10
10N	20	C9, C11, C21, C23–C28
MKT Polyester		
0.47µF	(470N) 9	C6, C7, C16, C17, C19,C20, C22
1µF	1	C18
2.2µF	10	C8, C12
Mains Polyester		
33N	1	C1
Electrolytic		
2,200µF/25V	2	C4, C5
10,000µF/ 40V	2	C2, C3
Total	62	

SEMICONDUCTORS

Op Amps:		
MC33079P	1	IC4
OPA2604	5	IC2, IC6, IC7
OPA604	1	IC5
20W Amplifier:		
LM1875T	5	IC3
Optocoupler:		
MOC3021	1	IC1
Voltage Regulators:		
+15	LM7815	S3
–15	LM7915	S4
BT139 6A triac		S1
6A bridge rectifier		S2
Mains Transformer:		
80V A 18-0-18		TR1

Fuse 6A M205 FastBlow
Fuse clips

PLUGS, SOCKETS, AND CABLES

RCA plugs and sockets (2)
3.5mm stereo plugs (2)
3.5mm stereo panel socket
Speaker connectors (5)
IEC panel socket
IEC mains cable
USB cable and plug
Stereo mini plug
c.4m twin shielded cable
2 panel grommets

HARDWARE/MISC.

10 BA nuts and bolts
9 TO-220 silicon impregnated rubber washers (or Mica)
9 TO-220 insulation bushes
3 TO-220 "U" type heatsinks
1 rectifier heatsink
200 × 75mm heatsink
Case: 90 × 200 × 300 or sim.
2 pole, 5 position make-before break (shorting) rotary switch
knob to suit
Transfer for dial

SPEAKERS

2 × AP100Z0 bass/midrange
2 × TM020J7 tweeter
1 × AP100G0 bass
All Audax speakers

New Chips on the Block

Burr-Brown PCM2702 DAC

By Charles Hansen

The PCM2702 is a single chip digital-to-analog converter offering two D/A output channels and an integrated USB 1.0 compliant interface controller. The newly developed SpAct™ (Sampling Period Adaptive Controlled Tracking—Patents Pending) system recovers a stable, low-jitter clock for internal PLL and DAC operation from the USB interface audio data.

The PCM2702 is based upon Burr-Brown's Enhanced Multi-level Delta-Sigma Modulator, an 8× oversampling digital interpolation filter, and an analog output low-pass filter.

The PCM2702 can accept 48kHz, 44.1kHz, and 32kHz sampling rates, using either 16-bit stereo or monaural audio data. Digital attenuation and soft-mute features are included, and are controlled via USB audio class request. Applications include stand-alone USB audio speakers, CRT/LCD integrated USB audio speakers, USB audio amplifiers, and other USB audio applications.

FEATURES

Integrated USB interface:
- Full-speed transceiver supports 12Mbps data transfer
- Fully compliant with the USB 1.0 specification
- Adaptive mode for isochronous transfer
- Self-powered device

Accepts 16-bit stereo and mono USB Audio data streams

Analog performance (VCC = 5V):
- Dynamic range: 100dB (typ at 16-bit)
- SNR: 105dB (typ)
- THD+N: 0.002% (typ at 16-bit)
- Full-scale output: 3.1V p-p

8× oversampling digital filter:
- Passband: 0.454fS
- Stopband: 0.546fS
- Passband ripple: ±0.002dB
- Stopband attenuation: –82dB

Sampling rate (FS): 32kHz, 44.1kHz, 48kHz

On-chip clock generator with single 12MHz clock source

Multi-functions:
- Digital attenuator: 0dB to –64dB, 1dB/step
- Soft mute
- Zero flag
- Suspend flag
- Playback flag

Dual power supplies:
- +5V for analog portion
- +3.3V for digital portion

Package: SSOP-28

BUDGETARY PRICING

Pricing will vary according to grade and package; 1,000 piece; Recommended resale in US$; FOB USA: $5.45

SpAct™ is a trademark of Burr-Brown Corporation.

Burr-Brown, Tucson, AZ, 1-800-548-6132, www.burr-brown.com. ❖

Audio Aid

Optimize Second-Order Filter Resistors

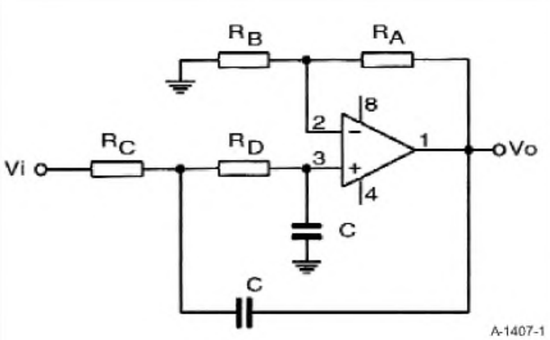


FIGURE 1: Optimize second-order filter resistors.

You can select the feedback resistors in a second-order filter to simultaneously set the required gain and minimize offset voltage errors from op amp input bias current. This is only significant when using bipolar input op-amps, not when using FET input amps.

Figure 1 shows the circuit and designators.

DC offset voltage errors occur when the impedances at the op-amp inverting and non-inverting inputs are unequal. The error is caused by the op-amp input bias current through the input impedances. It is especially important to minimize the DC output error when you wish to avoid input and output coupling capacitors.

For a second-order Butterworth filter, the parameters are Q = 0.707 and gain G = 1.586. These are related by

$$G = 3 - \frac{1}{Q} \qquad [1]$$

The gain is set by R_A and R_B

$$G = 1 + \frac{R_A}{R_B} \qquad [2]$$

Assume both capacitors have the same value, C, then let $R_C = R_D = R$, then the –3dB cutoff frequency is given by

$$f_C = \frac{1}{2\pi RC} \qquad [3]$$

Typically, f_c is selected, C is chosen to be a standard value, and the equation is solved for R. This means that R_C and R_D are fixed by this circuit design, and can't be changed without changing the cutoff frequency.

If the impedances and input bias currents at both op-amp inputs are the same, then the two error voltages cancel through the differential amplification of the op amp. Consider the following:

Input impedance at inverting input

$$R_- = \frac{R_A R_B}{R_A + R_B} \qquad [4]$$

Input impedance at non-inverting input

$$R_+ = R_C + R_D = 2R \qquad [5]$$

You know R in equation [5], because you chose it to set the cutoff frequency. Since you need to know $R_+ = R_-$, solve equations [4] and [2] as two equations in two unknowns, to find R_A and R_B. Some algebra results in

$$R_B = \frac{2GR}{G-1} \qquad [6]$$

$$R_A = 2GR \qquad [7]$$

As an example, suppose you want a cutoff frequency of 80Hz for a subwoofer crossover application. Choose $C = 0.1\mu F$, then from [3], R = 20k. Using [6] and [7], you find R_A = 63.4k and R_B = 108.3k. Choosing standard values of 68.1k and 110k results in G = 1.619, for an error of about 2%. Alternatively, you could parallel resistors to get a better match for the ideal values of R_A and R_B, which would reduce the gain error. However, with 1% resistors, the output error will be a little more than 2%, anyway, so it's not really worth the effort.

Again, this process is only worth the effort with bipolar op amps. You're essentially reducing the error from the input bias current to that of the input offset current. For FET input amps, the bias and offset currents are small and nearly equal, so you won't gain anything.

In addition, some bipolar op amps have input bias cancellation circuitry, so that the bias current is on the order of the offset current. These amps won't benefit from impedance balancing, either. For more on op-amp error sources, see *Operational Amplifiers*, Second Edition, Jiri Dostal, Butterworth-Heinemann. ❖

Doug Burkett
Eaton, Ohio

Adire KIT281

Reviewed by Joseph D'Appolito and Peter Millett

PHOTO 1: The Adire KIT281 loudspeaker under test.

According to Adire, their KIT281 loudspeaker is a high-efficiency MTM speaker designed for home audio reproduction as well as home theater applications. It features two Adire AV8 8″ mid-bass drivers and an Audax TM025F1 tweeter. The tweeter is offset from the front baffle centerline in order to reduce the mid-bass driver spacing and to control diffraction. Efficiency is stated to be 92dB SPL at 2.83V. Peter Millett (p. 34) built the vented version of the kit that I test here (*Photo 1*).

I ran a series of impedance, frequency response, and distortion tests on the KIT281. *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays a highly suppressed version of the double-peaked curve typical of vented systems. The impedance minimum of 2.72Ω at 24.3Hz indicates the vented-box tuning frequency. There is a second local impedance minimum of 2.79Ω at 150Hz.

As part of the low-frequency crossover, a 7.5Ω resistor is placed directly across the mid-bass driver terminals. This produces the rather low impedance levels below 500Hz. Coupled with impedance phase angles ranging from +44° to −56°, this may be a difficult load for many amplifiers.

The impedance peak of 34Ω at 1827Hz in *Fig. 1* is caused by the interaction of the woofer and tweeter crossover networks forming a parallel resonance at that frequency. We have seen this phenomenon several times before, and perhaps it is worth a moment to explain why this happens.

Figure 2 shows the impedance curves for the system and the individual drivers. Notice that the mid-bass driver/crossover combination shows a rapidly rising curve above 1kHz. Above this frequency the mid-bass impedance is essentially inductive.

Conversely, the tweeter/crossover combination impedance is falling rapidly in the same frequency range and is therefore essentially capacitive. The curves cross at 1827Hz, where the inductive and capacitive values are equal. Since the two networks are connected in parallel, they produce a parallel resonance at that frequency characterized by an impedance peak with zero phase.

FREQUENCY RESPONSE

Figure 3 shows my first set of measurements of KIT281's on-axis frequency response. The first measurement (shown dotted) displays a deep response notch centered on 2022Hz. I suspected a tweeter polarity problem.

Fortunately, the KIT281 is set up for bi-wiring, so I removed the terminal straps and reversed tweeter polarity with a jumper set. The resulting measurement is shown by the solid curve on *Fig. 3*. Relative to the second curve the notch depth is 10dB. All further frequency response data was taken with the new tweeter polarity. (Note: Since this speaker was sent to me after listening tests, this response anomaly may affect reviewer results.)

Figure 4 shows the full-range response of the KIT281. This response is obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and near-field port responses. I placed the microphone along the tweeter axis at a distance of 1.25m to produce the far-field response. I then spliced together the near- and far-field responses at 200Hz to produce the full-range response[1]. The response data is ⅙ octave smoothed.

The response shown in *Fig. 4* has been normalized to 1m to obtain system sensitivity, which averages 89.7dB/2.83V/1m in the two octaves centered about 1kHz (500Hz–2kHz). This is 2.3dB lower than Adire's specification for this system. Relative to this level, the −3dB points are 38Hz and 19.5kHz. Response over this range is +2.5dB and −3.4dB. The pronounced response ripple between 800Hz and 2kHz is explained in the next plot.

CROSSOVER ACTION

As stated previously, the KIT281 has two pairs of binding posts for bi-wiring. This allowed me to measure the response of the individual drivers (*Fig. 5*). Crossover occurs at 2348Hz. In the octave below crossover, tweeter response falls 40dB. Above crossover, mid-bass pair response falls off at 20dB/octave in the first octave.

These results are consistent with the electrical crossovers, which are second-order and fourth-order for the mid-bass drivers and tweeter, respectively. Notice the evenly spaced ripples in the woofer pair response. Ripple peaks occur at 1200, 2400, and 3600Hz and are inherent in the woofer response as they are shown in the AV8 response plot on the Adire website.

WOOFER/TWEETER TIMING

Figure 6 shows the KIT281 step response. It is obtained by a numerical integration of the system impulse response. The ideal step response should be a single rapid rise followed by a smooth decay through the 0.00 level. *Figure 6* shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival. It is followed by the woofer pair arrival, peaking about 0.7ms later. (The small initial negative-going spike is inherent in the response of the high-order tweeter crossover.) The drivers are both connected with positive polarity, but the system is not time-coherent.

The exact woofer delay is best determined by examining the excess group-delay plot (not shown). Excess group delay is a very sensitive measure of inter-driver delay[1]. This plot shows the woofer pair to be 350μs behind the tweeter.

CUMULATIVE SPECTRAL DECAY

The KIT281 cumulative spectral decay (CSD) response is presented in *Fig. 7*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.11ms increment of time. The total vertical scale covers a dynamic 35dB range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis would indicate the presence of a strong system resonance.

The first time slice in *Fig. 7* (0.00ms) represents the system frequency response. Initial tweeter high-frequency decay is relatively good; the bulk of its response above 10kHz has decayed away in about 1.2ms. The low-frequency decay is fairly typical of vented loudspeakers.

HORIZONTAL POLAR RESPONSE

Horizontal polar response is examined in *Figs. 8–10*. *Figure 8* is a waterfall plot of horizontal polar response in 10° increments from 60° right (+60°) to 60° left (−60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00°. For this reason, the plotted curves show the change in response as you move off-axis.

For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles. For home theater applications a more restricted high-frequency coverage is desirable.

The KIT281 sent to me for measurement has its tweeter mounted to the left of the baffle centerline. Notice when moving to the left, a deep response notch develops, centered on the crossover frequency. This is caused both by the steep tweeter response rolloff and the directivity of the 8″ mid-bass drivers.

When moving to the right, the notch is much less pronounced. However, response ripples above the crossover frequency become more pronounced as you move off-axis to the right. Remember that the off-axis curves are relative to the on-axis response. Thus the increase in ripple amplitude means one of two things: either the off-axis curves are actually smoother than the on-axis response, or the off-axis ripples are shifted in frequency relative to the on-axis ripples.

Figure 9 answers this question and reveals another interesting aspect of the horizontal polar response. First, notice that the response ripples at 30° have shifted in frequency relative to the on-axis ripples. Second, as you move off-axis, response to the right is smoother than to the left and closer in shape to the on-axis response.

For best coverage of the primary listening area, you should place this speaker on the left side. Adire's enclosure drawings state that the baffles should be built in mirror-image pairs. These measurements show that you should place the tweeters to the outside of the primary listening area.

The average response over a 60° horizontal window (±30°) in the forward direction is shown in *Fig. 10*. This response is very similar to the on-axis response, especially below 10kHz. There will be little change in spectral balance as you move off-axis if you place the speakers as I just suggested. Due to the size of the KIT281, I could not safely mount it on my test turntable for vertical polar-response measurements.

HARMONIC DISTORTION

I ran harmonic distortion tests at an average level of 90dB SPL. Ideally, harmonic distortion tests should

MANUFACTURER'S DATA:

Frequency Response: 39–20000, ±3dB
Impedance: 4Ω
Sensitivity: 92dB SPL @ 2.83V RMS, in-room
Power handling: 150W RMS, long-term
Weight: 85 pounds (typical) per speaker
Size: 48″H, 9″W, 14″D (typical, vented cabinet)
Cost: $339 per pair, not including cabinets

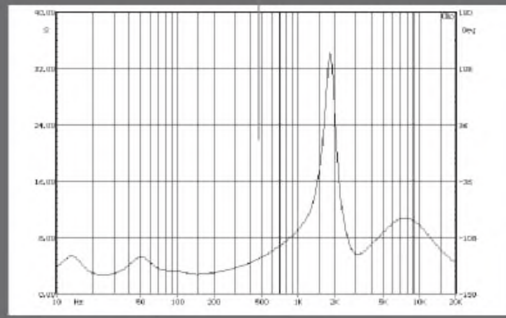


FIGURE 1: Adire KIT281 impedance.

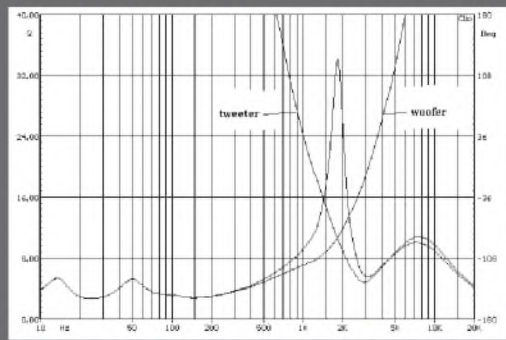


FIGURE 2: Tweeter and woofer impedance curves.

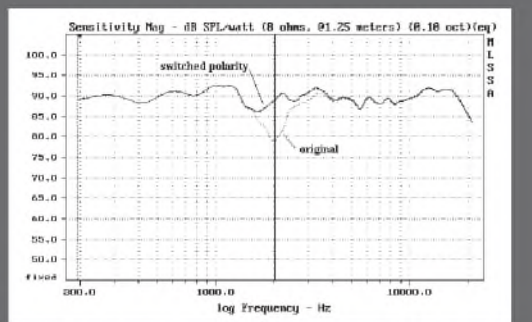


FIGURE 3: First measurements showing tweeter polarity problem.

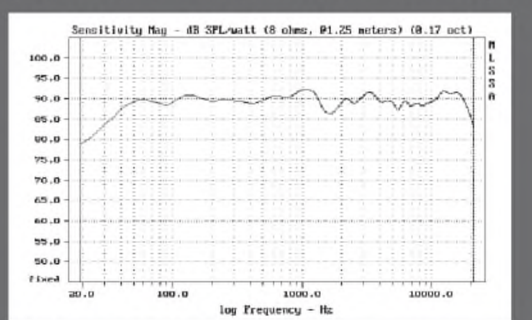


FIGURE 4: Adire KIT281 full-range frequency response.

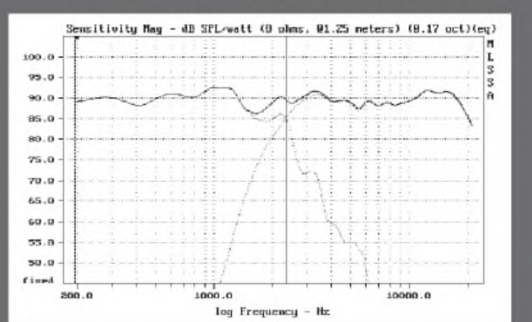


FIGURE 5: KIT281 system and driver responses.

Reviewed by Pete Millett

I listened to a variety of music with the KIT281 speakers, using both analog and digital sources and two different tube amplifiers. Overall, I was impressed with the capabilities of this speaker for such a low price.

BREAK-IN

I noticed that the bass was a bit thin when I first started listening to the speakers. This weakness disappeared fairly quickly. I played music through the speakers for several hours before doing any serious listening.

LISTENING IMPRESSIONS

As in other speaker reviews in *audioXpress*, I used the *Hi-Fi News and Record Review* Test Disc III CD to record some of my observations. This was a little difficult, partly because I am not familiar with the recording, and partly because I don't listen to this style of music (basically all classical) very much. Nevertheless, here are my impressions:

Track 2—Good imaging, but sounded a little flat. Perhaps this is the recording?

Track 4—The strings sounded very natural. Trumpets were bright (as they should be), almost sounding like a horn speaker.

Track 5/6—The sound of the narrator (as though he were in a garbage can) made me suspicious of the recording quality, but the orchestra sounded pretty good. Very good, clean bass extension, especially on bass horns and strings. Oboe and clarinet sounded detailed and natural.

Track 7—Nice imaging, but female vocal sounds a little harsh. Might be the recording, since I didn't notice this on other recordings.

Track 10—This percussion track sounded good, but seemed to lack some bass extension. This must be the recording, since bass extension was fine on the previous tracks.

Track 14—The bass was back on this track. This sounded very dynamic.

I also listened extensively to the KIT281s with the music that I'm very familiar with. My references include Klipsch KLF-10 loudspeakers and Sennheiser HD600 headphones.

Modern jazz was truly amazing on the KIT281s. Piano and guitar sounded as though they were in the room with you. The same could be said of Steve Earle's Train 'A Comin CD, which I guess I'd call bluegrass-like acoustic blues music. Strings, especially the bass, sounded very real. Both female and male vocals, such as Tracy Chapman and Bruce Cockburn, also sounded excellent.

FREQUENCY RESPONSE

Adire says that the KIT281 is voiced for far-field listening. I often listen in the near field, so I started out with the KIT281 located about 4–5′ from me. When I first tried the KIT281 at this close distance, there was an odd hollowness to the midrange that was most noticeable on male vocals. When I moved progressively farther from the speakers, this effect diminished. At a distance of 10′, I no longer noticed this hollowness.

As Joe D'Appolito discovered in his testing, there was an anomaly in frequency response, which was caused by the polarity of the tweeters being incorrect. After reversing the tweeter leads, this problem was gone. Voices sounded as they should, both in the near field and far away.

The bass response of the KIT281 was excellent—better than I expected. In my listening room, the bass was clean and not at all boomy. In-room response (measured unscientifically with a hand-held meter) was within ±3dB down to about 40Hz, just as advertised. There was no in-room rise in the midbass that I could detect.

IMAGING AND SOUNDSTAGE

Being used to horn speakers that beam like a flash-

to page 65

be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of an harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Second and third harmonic distortions at 50Hz and 90dB SPL were 4.9% and 4.4%, respectively. A good, though not exceptional, result. All HD distortion was below 1% above 100Hz. Also a good result.

INTERMODULATION DISTORTION

I measured intermodulation distortion next. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then, a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I examined woofer intermods first by inputting 400Hz and 500Hz signals at equal levels. These frequencies should appear predominantly in the woofer output. Total SPL with the two signals was adjusted to 87dB at 1m.

Because steady tones are used in the IM test, I thought it safer to use a lower power level to prevent possible tweeter damage. Principal woofer IM products occurred at 600, 1300, 1400, 2200, and 2300Hz. However, the overall level was only 0.43%, a better than average result.

I measured tweeter intermods with a 9kHz and 10kHz input pair also adjusted to produce 87dB SPL at 1m. IM products were observed at 8, 11, 12, 18, and 19kHz. Total distortion was 0.2%, a fairly typical result for tweeters.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 400Hz and 9kHz. Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. IMD products appeared at 8600 and 9400Hz at a level of 0.036%. This is one of the lowest figures I have measured for this type of distortion. The tweeter's fourth-order electrical filter is primarily responsible for this very low figure.

A Note on Testing: The Adire KIT281 was tested in the laboratories of Audio and Acoustics, Ltd., using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data were measured with an ACO 7012 ½" laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamp. Polar response tests were performed with a computer-controlled OUTLINE turntable on loan from the Old Colony Division of Audio Amateur Corporation.

Manufacturer's Response:
Thank you for the opportunity to have our KIT281 reviewed, and thanks to Peter and Joe for their diligent work. We would like to add a little to what they wrote.

The 92dB SPL we specify for the 281 represents the SPL we think the end user will achieve in their listening room given 2.83V input, i.e. an effective in-room sensitivity. Joe's anechoic measurement of 89.7dB is consistent with this.

The 281 was originally targeted towards the common low-output-impedance amplifier, thus the observed peaks in impedance would have little effect on the sound. An impedance-conjugate filter schematic is available for the KIT281 and should have been promoted to Peter. We are sorry not to have made this available during the review in light of his use of low-power tube amplification. Our own testing shows that with this filter in place, the impedance and phase variations are strongly reduced (to less than ±0.7Ω and ±15°), greatly improving the sound on high-Z amps. Visitors to our room at the 2001 Vacuum State of the Art Conference (V.S.A.C.) were able to

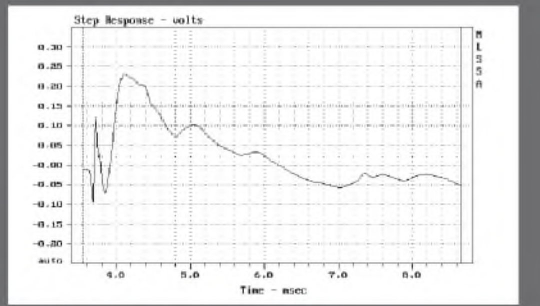


FIGURE 6: KIT281 step response.

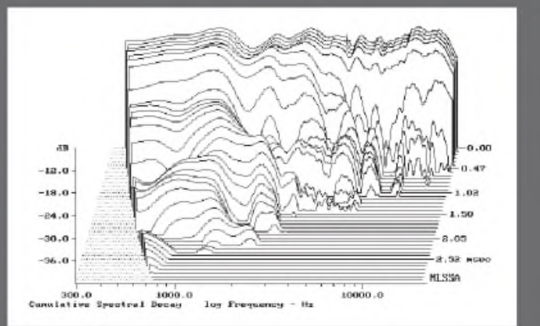


FIGURE 7: KIT281 cumulative spectral decay.

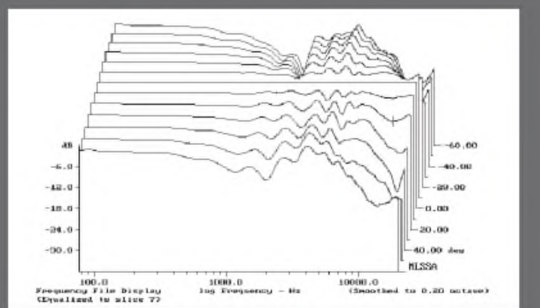


FIGURE 8: KIT281 horizontal polar response waterfall.

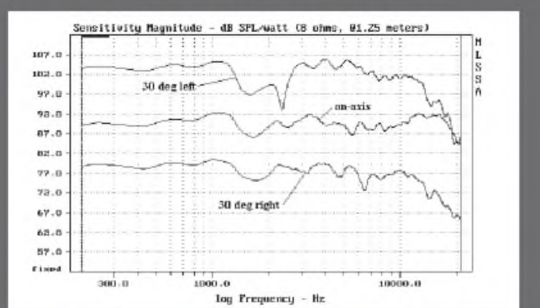


FIGURE 9: KIT281 horizontal polar response at 0 and ±30°.

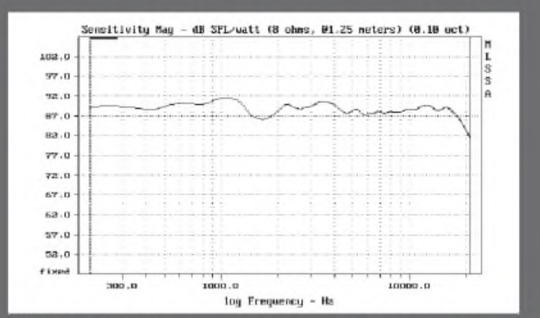


FIGURE 10: KIT281 average response over ±30° window.

confirm this for themselves. On a related note, the 7.5Ω resistor noted in the woofer filter serves a related impedance-adjusting function in addition to shaping the response curve, and contributes to the well-damped impedance peaks around box resonance.

Similar to the Z-filter, we developed a series of easy add-on "tweaks" in response to a few customers' wish to soften the treble in certain listening environments. These have been available almost since the release of the 281, and represent our determination to continue to improve the value of our products, even once shipped.

Joe's measurements show the box tuning to be 24.3Hz, lower than the 28.8Hz specified in the kit instructions. We feel that the specified tuning would have given even better distortion results than were measured in this review, due to better control of the woofer cones at the test frequency. Peter's review does not contain the information necessary to understand the discrepancy in tuning frequency, but we are in contact with him to see if improvements are possible.

The tweeter phasing was indeed an issue, for which we apologize. After sending Peter his kits with the pre-assembled crossovers (standard in the 281), with his and Joe's feedback, we discovered that a number of tweeter crossover boards were shipped mis-marked. Affected customers have already been contacted. Though we wish we had discovered this sooner, the audible effect was mitigated somewhat by the steep filter attenuation reducing the driver overlap and thus the bandwidth of the affected spectral region.

We are glad that both Peter's comments and Joe's measurements confirm that our design goals were readily achieved and give audible results, consistent with what we hear repeatedly from our customers. The drivers, crossover, and overall system design targeted low distortion, good dynamics, balanced bass, lifelike timbre, great clarity at low and high SPLs, ease of room placement, large listening sweet spot, and superior image width and stability, all of which were noted in this review. We believe that the 281s are easy to build, place, listen to, and live with. We wish Peter many years of enjoyment and are glad to answer any further questions he might have. ❖

Dan and Dave
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REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

from page 63

light, I was very pleased with the KIT281 imaging. I noticed very good off-axis response, both in the horizontal and vertical planes. There was no distinct "sweet spot" in the room—in fact, the KIT281 could fill the room with sound very nicely.

I found the best positioning seemed to be about 10′ from the speakers, with the speakers about 8′ apart, toed in slightly—but again, the speakers were not very sensitive to positioning. They did not seem to become boomy when I moved them near the back wall. The image held up nearly anywhere in the room.

The soundstage was slightly wider than the speakers and seemed taller than my other speakers, perhaps because these are tall speakers. The tweeter is centered at ear-height, and the enclosures stand 4′ tall.

AMPLIFIERS AND POWER REQUIREMENTS

I drove the KIT281 speakers with two different tube amplifiers of my own design. The first was a 6W push-pull triode amplifier (using 6B4Gs). This amplifier sounded fine with the KIT281, but is a little lacking for power for these speakers, at least if you want to listen very loud. At clipping the sound level was adequate—I measured 93dB at 10′ from the speaker.

The KIT281s were excellent when driven from my 25W KT88 amplifiers. They sounded perhaps a little better in triode mode than in pentode, but both were fine.

LISTENING ROOM AND EQUIPMENT

My listening room is about 14′ wide by 20′ long, with the system set along the shorter wall. The room is typical, containing carpeting, furniture, and so forth.

In addition to the amplifiers already mentioned, my system includes an AVA super-PAS preamp, an NAD turntable with AT OC9 cartridge and Sowter step-up transformers, and an EAD CD-1000 CD player.

CONCLUSION

The Adire KIT281 is an excellent speaker for the money. It's a great choice if you want to fill the room with music, especially if it's a large room. I suspect that they would make excellent home-theater main-channel speakers as well.

Of course, the KIT281s aren't perfect, but the same can be said of commercial speakers that cost five to ten times as much, to which I think they compare favorably. I would not hesitate to recommend them to anybody who can do a little woodworking and is looking for a good all-around speaker.

Software Review

Wave Repair

Reviewed by Richard Mains

Wave Repair is a shareware program that runs on Windows 95/98/NT (and also reportedly on 2000 and ME, although it is unclear from the product website whether the program has been fully tested on these platforms). Wave Repair was written by an individual in the UK for use in his hobby of transferring LPs to CDs. You may try out the program for free, but the save feature is disabled until you register the software. Registration costs $30 US and may be carried out online. Registration entitles you to e-mail support by the software developer, and to unlimited free upgrades to future versions of Wave Repair.

INTRODUCTION

When I decided that I wanted to transfer some of the LPs in my collection to CDs, I found that I was faced with several computer hardware and software issues. I searched the web for any sites I could find that might address some of these issues. I found several sites that were helpful, but one site in particular not only discussed all these questions, but also offered a very useful and reasonably priced program for recording, cleaning up, and preparing audio tracks from LPs for burning onto CDs. I thought this program, called Wave Repair, would be of interest to many *audioXpress* readers.

The website discusses some rather esoteric topics that I had never heard of before, such as wet playing and the Keith Monks record cleaning machine. There are also several links to other software products, and some hardware issues such as computer soundcards and preamps are discussed. I don't care to address the issue of computer soundcards in depth in this review, but I will mention that I followed one of the author's suggestions for my own system: using the Midiman Flying Calf external A/D converter and DiO2448 digital sound card for capturing LP recordings.

First, let me explain what Wave Repair is not. It is not a general .WAV file editor. For example, you can't cut and paste sections of waveforms. You also can't re-sample .WAV files at different sampling rates, nor change the basic .WAV format (e.g., to .MP3 format).

For this type of program I would suggest, for example, Cool Edit 2000, which was reviewed by Perry Sink in April '01 *audioXpress*. Wave Repair also cannot perform some other functions that Cool Edit provides, such as broadband noise reduction.

Wave Repair is primarily a tool for repairing clicks, pops, and other defects in vinyl recordings. Although there are automatic de-clicking features, the author stresses that the best way to remove defects from LP recordings is to manually repair them. Whether this is true is probably best left as an issue for debate among *audioXpress* readers.

My own experience is that automatic de-clickers (available both in Wave Repair and Cool Edit 2000) can be useful, especially for highly damaged LPs; however, there is always the chance that they will distort the music, possibly leaving the recording in worse shape than before they were applied. I personally prefer to manually de-click if possible, but if the defects are very dense I first apply an automatic de-clicker with the parameters set not too aggressively, so that the real music is not disturbed; afterwards, I go back and manually remove any remaining defects.

Wave Repair also includes some useful features for processing .WAV files, such as compression and volume ad-

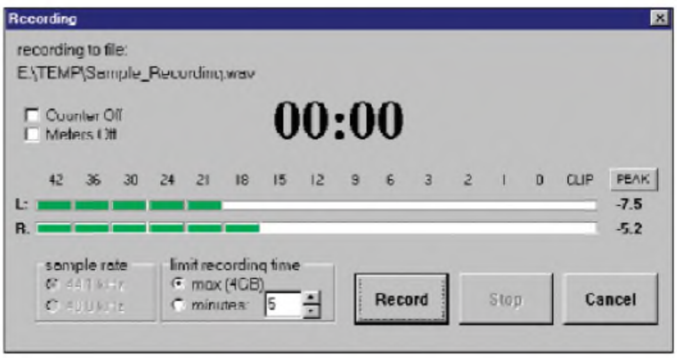


FIGURE 1: Wave Repair's recording control screen.

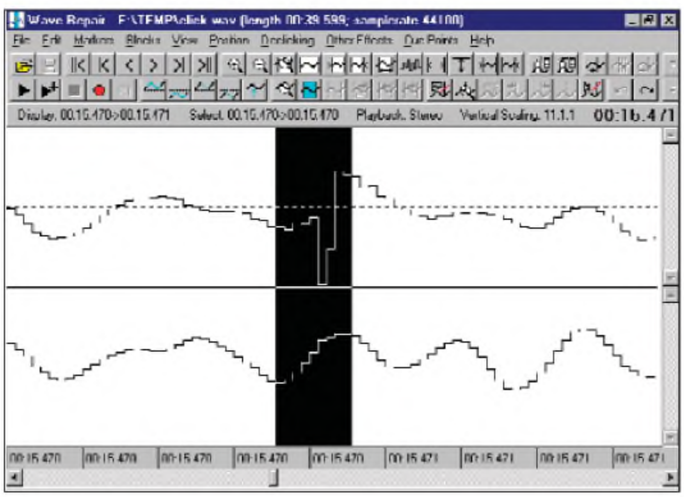


FIGURE 2: Anatomy of a click. The region in black has been selected for repair.

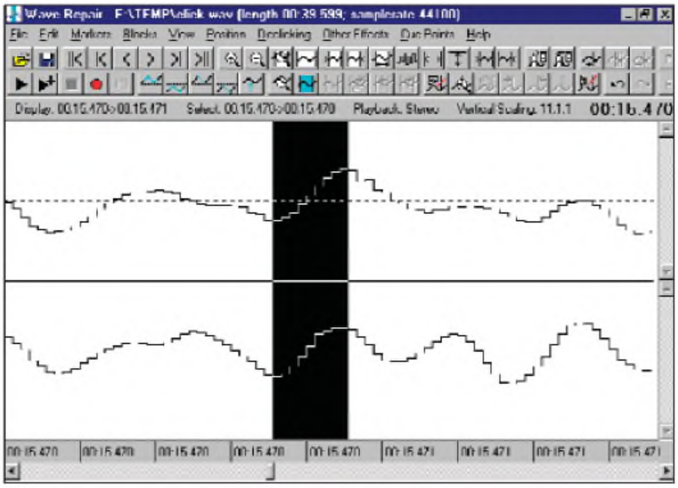
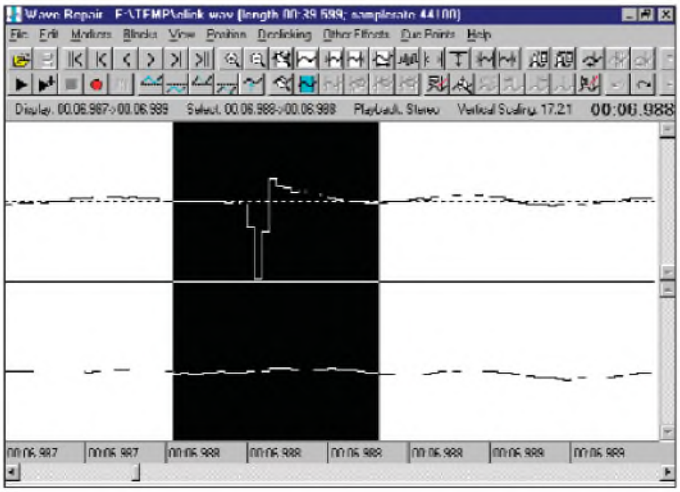


FIGURE 3: After Bezier correction has been applied to the click of *Fig. 2*.

FIGURE 4: Region with another click, selected for linear repair.



justment. You can use Wave Repair to record .WAV files directly from your soundcard, as you will see.

MAKING THE RECORDING

Before you can repair the pops and clicks that plague vinyl recordings, you need to record the information digitally on your computer. Wave Repair is useful for making recordings directly from your soundcard, although the recording options are limited to 16-bit stereo at sampling rates of either 44.1kHz or 48kHz. *Figure 1* shows the window that comes up when you start a recording. The first step is to use the "monitor" mode, which does not transfer any data onto your disk, but simply monitors the sound level. This is the mode that is depicted in *Fig. 1* (you can tell because the recording time counter stays at 00:00).

In the figure, the green LEDs are monitoring the instantaneous levels of the left and right channels. On the right under the "peak" button are the overall peak values that have been observed so far; you can reset the peak values by clicking the "peak" button.

The object during the monitoring stage is to set the input volume level about as high as possible without causing clipping. If clipping should occur, red LEDs under the CLIP indicator light up, and remain until you reset the peak indicator. So, if you wish you can play the entire side of an LP and come back when it is done to see whether any clipping has occurred.

When you are satisfied with the volume setting, you can start the actual recording process by clicking the "Record" button. The clock shows the elapsed time in the recording, and the LEDs continue to monitor the input level. You can pause and restart the recording (for example, while you are turning over the LP to the second side). Wave Repair can record .WAV files up to 4GB in one recording session.

If you record an entire LP in a single .WAV file, Wave Repair can split the file up into separate tracks for CD burning. (In this regard, the program also addresses the issue of CD block size, which is important for avoiding "clicks" at track boundaries when you play the CD). You can also use it to generate cue sheets that some programs use to split the wave file into separate tracks as they burn the CD. You cannot use Wave Repair to burn CDs directly, however.

In addition, Wave Repair has the capability of placing fade-ins and fade-outs at the beginning or end of a track, and portions of the .WAV file can be deleted. Areas of silence can also be placed in the .WAV file.

REPAIRING CLICKS AND POPS

If you have done any work repairing defects in .WAV files from LPs, you are probably all too adept at recognizing the appearance of pops and clicks. For those who are not familiar with them, *Fig. 2* shows how a typical click appears in a .WAV file. The top trace in *Fig. 2* is the left channel, and the bottom trace is the right channel. Note that in this case the click appears almost entirely in the left channel; it is quite common for clicks to appear asymmetrically in the two channels. The area in black has been selected for repair.

Wave Repair uses basically four different methods for repairing defects. The first is Bezier interpolation, in which a Bezier curve replaces the defect, such that the endpoint values and slopes of the curve coincide with the endpoints of the selected region. *Figure 3* shows the result of applying Bezier correction to the click of *Fig. 2*.

With Wave Repair you can listen to the waveform both before and after the correction, to determine whether the repair has eliminated the click. In this case the repair is probably adequate, although you can still see a vestige of the click just to the right of the region where the repair was made. (You can remove this by a second application of Bezier interpolation, or with another method that I will describe later in the review).

Since Bezier correction takes into account the slopes at the endpoints, it is sometimes not the best choice, particularly for repairs that extend over a larger region. *Figure 4* shows another click amid low-level music that was selected

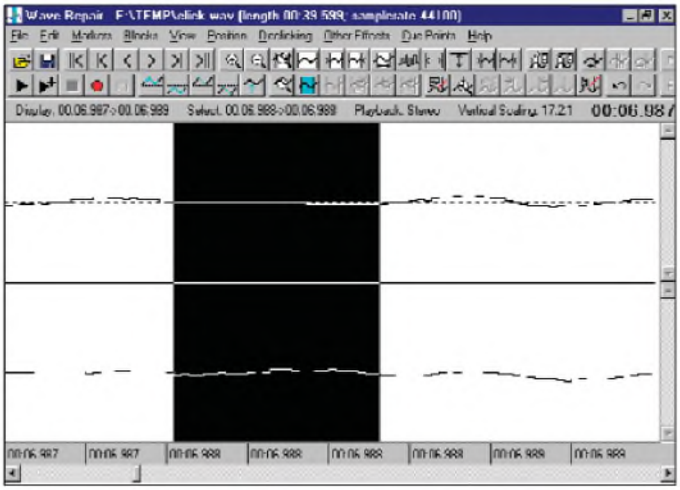


FIGURE 5: After applying linear correction to the click of *Fig. 4*.

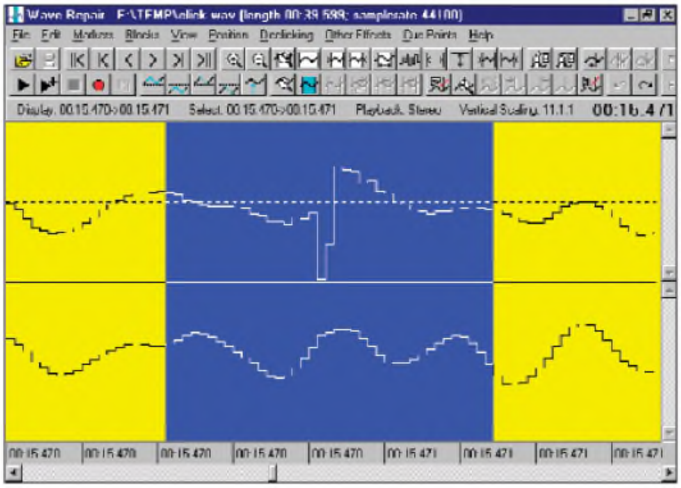


FIGURE 6: The region of *Fig. 2* set up for redraw mode.

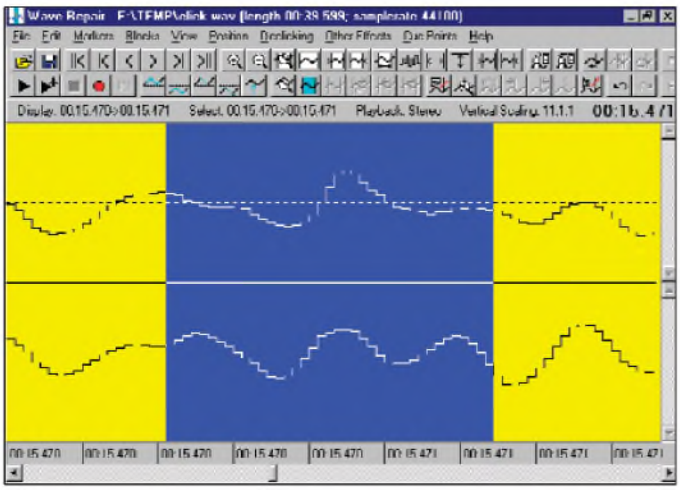


FIGURE 7: After the region with the click in *Fig. 6* has been redrawn freehand with the mouse.

for repair. Wave Repair also offers the option of linear correction, in which a straight line is drawn between the endpoints of the selected region. *Figure 5* shows the result of applying linear correction to the region of *Fig. 4*.

As I mentioned earlier, Wave Repair does have the option of automatically finding and repairing clicks and pops, although the documentation discourages the use of this option. Another possibility is to use the automatic de-clicker to find pops, but then to repair them manually. You can set up various parameters to control how many and what kinds of clicks and pops the automatic de-clicker will detect. I normally use only the automatic de-clicker in regions that are supposed to be silent—at the beginnings and ends of tracks. Even then, I must go back afterwards and manually eliminate any vestiges of clicks that the automatic de-clicker did not fully remove.

So far I have discussed the repair of clicks and pops, but how do you find them? For some clicks this is quite easy, but for others it can be extremely difficult. I normally use what I call a "bracketing" technique.

A region is first selected such that the click is heard near the end of the region. The right-hand boundary of the region is gradually moved backward to the left, until the click disappears; at that point you know that the boundary has just passed through the click. Unfortunately, though, it sometimes becomes difficult to hear the click as it gets close to the right-hand boundary, so you are not always certain when you have passed over the click.

I also use this method for regions with multiple clicks that are close together. I keep moving the boundary and noting the positions of any clicks that I see, until they have all disappeared. Then I go back and correct all the clicks on my list.

Most of the time you can see a click, but sometimes even though you know it is nearby, you can't see in the waveform what is causing the problem. In a case like this, I keep trying and un-doing corrections of different nearby regions of the waveform, until I find the defect.

REDRAWING AND OVERLAY BLOCKS

Sometimes defects extend over a larger region of the waveform and neither the Bezier nor the linear repair technique is adequate. For these situations, Wave Repair has a "redraw" mode, by which you can manually redraw any portion of the waveform using the mouse.

For example, remember that the Bezier repair of *Fig. 3* was not entirely successful, because it left vestiges of the defect to the right of the repaired region. *Figure 6* shows the same region after it has been set up in redraw mode. Note that the background screen colors change in this mode.

Figure 7 shows the waveform after the area with the click has been manually redrawn with the mouse. Of course, the success of this method depends on how skillfully you can infer what the shape of the waveform should be.

Yet another technique for handling

extended defects is the use of "overlay blocks." The basic idea here is that if a particular region is too badly damaged, rather than attempting to repair it, you simply replace it with another nearby, compatible region of the waveform. I have found this to be a very powerful technique for repairing extended defects such as loud pops.

Figure 8 shows a screen display in which an overlay block has been set up. This display will require some explanation.

First of all, the region with black background has been selected to be replaced by an overlay block. You can see that there is a defect in this region in the left channel. Once you select a region to be replaced, Wave Repair lets you slide it back and forth along the time axis, so you can compare it with nearby sections of the waveform to find a compatible block to replace it.

The overlay curve inside the vertical dotted lines is a replica of the region selected to be replaced, which I have moved over to the left. I have placed it over a section of the waveform which appears to be a reasonably good match at the endpoints, and which has similar features in the interior. When you find a region that is a candidate for replacing the damaged area, you can try it out by clicking the "copy overlay block (left or right)" option.

Figure 9 shows the result of copying the selected overlay block for the left channel. After you have made the replacement, you can audition the waveform to see whether or not the overlay resulted in an improvement. If not, you can undo the change and try other regions as overlays.

Yet another option for replacing portions of the .WAV file is to select a region and copy the contents of the left channel to the right, or vice versa. This can be a very effective correction provided that the two channels are similar in that area.

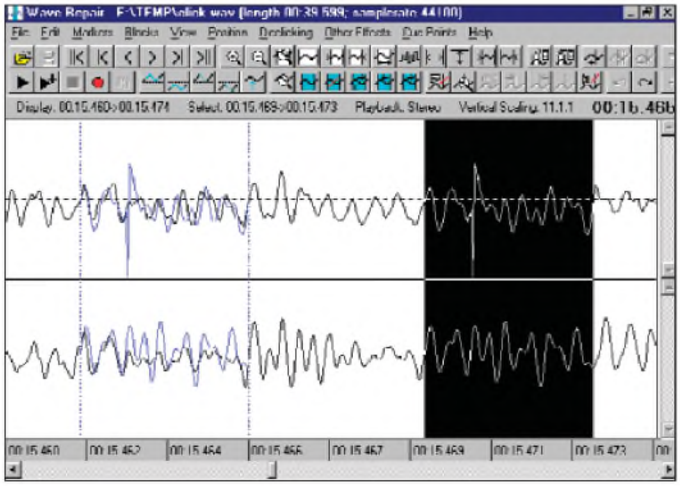


FIGURE 8: Setting up an overlay block to repair a region (in black) with a defect.

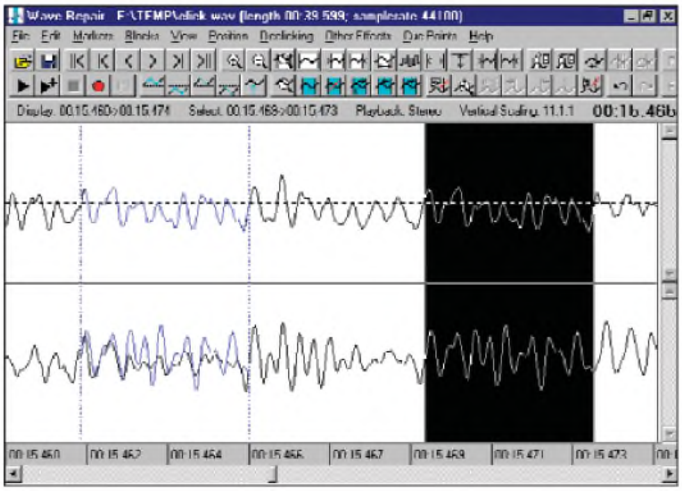


FIGURE 9: After replacing the damaged region of *Fig. 8* with the selected overlay block.

SUMMARY

Wave Repair is a highly useful and reasonably priced tool for anyone interested in cleaning up his/her LP collection for transfer to CDs. The author of the program is genuinely interested in the subject, and I have found him to be very responsive and helpful via e-mail. Since you can try the program for free, I encourage anyone who is interested in digitally recording LPs to investigate it.

The only disadvantages of Wave Repair that I have to report are related to its simplicity. As I mentioned previously, it does not offer some of the options such as cutting and pasting and broadband noise reduction that are available in other software packages. Even the waveform display is rather simple, as you have seen in the figures. The samples are shown in a staircase representation, with no attempt to smooth the waveform.

Recording options are currently limited to 16 bits, at 44.1kHz or 48kHz sampling rates. Also, the methods that it uses to repair defects are fairly simple: linear and Bezier interpolation, redrawing, or waveform block replacement. It does not use FFT transform techniques, which are used, for example, in the Cool Edit 2000 click and pop eliminator.

The documentation with Cool Edit 2000 claims that its method of dealing with clicks and pops is much more accurate than drawing a line or simple curve through the defect. I can't say for certain whether this claim is valid, but I can say that the repair options available in Wave Repair are normally sufficient for me to remove defects to the point where they are no longer audible to me.

However, I might mention that Wave Repair's simplicity also has some advantages. It is relatively easy to use and understand. And when you install the program, it doesn't alter the Windows registry or other such nonsense, so the program is quite easy to install and upgrade.

RESOURCES

The author of Wave Repair offers several websites related to the program and to the process of transferring LPs to CDs. General information about many aspects of LP to CD transfer may be found at http://homepages.nildram.co.uk/~abcomp/lp-cdr.htm. There is also a FAQ page that provides much useful information (for example, the format of a .WAV file) at http://homepages.nildram.co.uk/~abcomp/pcaudiofaq.htm. Information specifically about Wave Repair and how to register the software is at http://homepages.nildram.co.uk/~abcomp/wavrep.htm. Finally, you can contact the author of Wave Repair at clive@capita.nildram.co.uk.

Manufacturer's Response:

Thank you for giving me the opportunity to reply to this review. I feel that it is a well-balanced and fair description of Wave Repair. I would just like to comment on the "bracketing" method described to home in on glitches that are difficult to find. I too use such a technique, although in my case I move the start of the selection forward rather than the end of the selection backward. Apart from this difference, the two techniques seem pretty much identical. I had never thought of doing it the way Richard describes, but will certainly try out his method. I suspect both approaches will be equally effective.

Clive Backham
Author, Wave Repair

Xpress Mail

LAYOUT ERRORS

For those building the speaker systems depicted in the article "Build the Audax/D'Appolito Home Theater System" by John Calcote (Oct. '01, p. 8), please be aware there are several errors in the crossover layout diagrams.

Mr. Calcote appears to have taken the schematics and layouts for the crossovers from the Audax website. I queried Joe D'Appolito as to the correctness of the layouts and he replied that he was aware of the errors and that the schematics were correct and not the layouts. Mr. D'Appolito also mentioned that if the polarity was reversed in the output of the woofer terminals for the center channel (as shown in the layout), it would afford a slightly flatter frequency response.

The errors in the layouts are as follows:

1. Left/right woofer layout: There should be no connection between the junction of C1 & L2 and the junction of R2 & R3.
2. Center channel midrange layout: The designations of C2 & L2 should be reversed.
3. Center channel woofer layout: All plus and minus signs on the woofer connections should be reversed (see prior notation from Mr. D'Appolito) to agree with the schematic.
4. Surround tweeter layout: L1 should be L2 to agree with the schematic.

On another note: I'm sure you have been informed of the error on the cover on the Oct. issue; i.e., naming this project "Focal" instead of Audax.

William Eckle
Phoenix, Ariz.

The editor, just returned from a visit to the Focal/JMlab plant, had Focal on the brain.—Ed.

NONPROPORTIONAL CORRECTION

The first three parts of my article have gone well, except for a rather embarrassing error on page 75 of the November issue ("The Ultra Fidelity Computer Sound System," Part 3). Figure 20, an LMS printout, shows the frequency scale incorrectly. This is a Corel Draw problem that sometimes occurs when drawings are displayed on

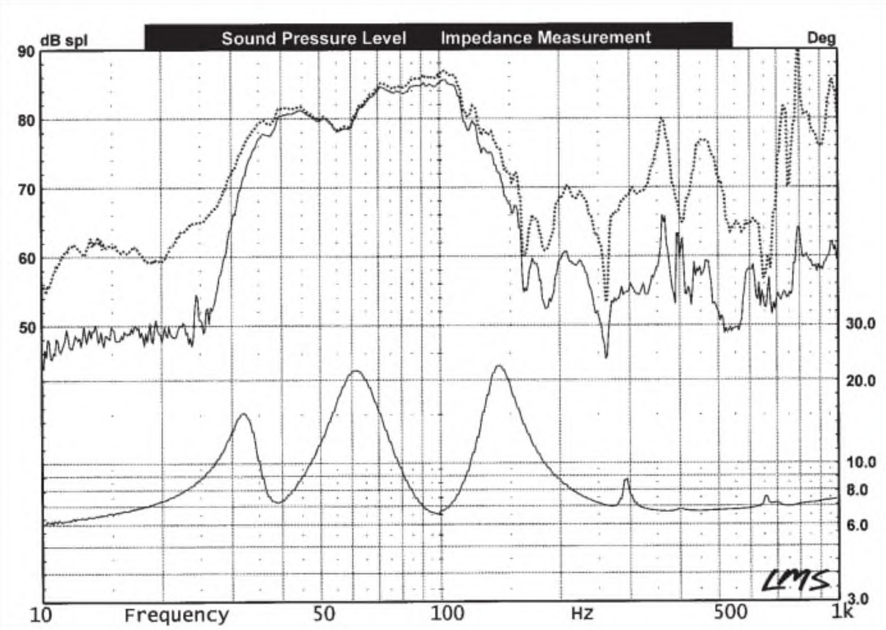


FIGURE 1: Stonjek's corrected graph of sub with and without active crossover, and impedance.

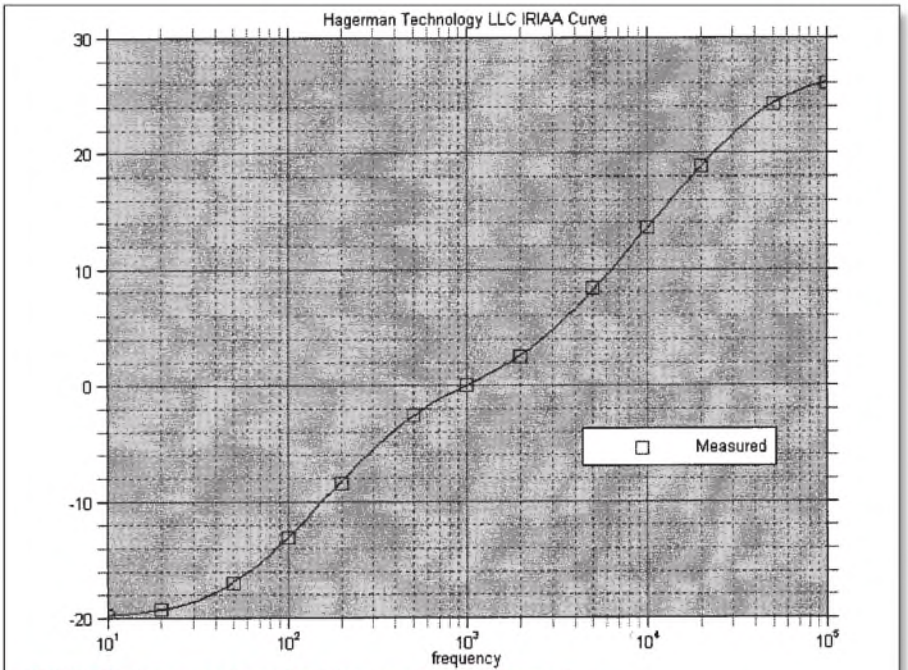


FIGURE 2: Hagerman's revised frequency measurements.

different computers. It displays correctly on my computer, and on the proof copy it is correct, so the problem occurred after the proof that I checked. This is an embarrassing error because it shows the subwoofer's performance to have no deep bass at all, whereas the deep bass is the one feature I wish to highlight. Looking at the graph as presented on page 75, you would think that the unit rolls off just below 100Hz instead of below 40Hz. The correct printout is shown in *Fig. 1.*

R. K. Stonjek
Tasmania, Australia

PREAMP RESPONSE

When I read Satoru Kobayashi's article "An Easy Solid State Preamplifier" (July '01 *aX*, p. 8), I nearly fell out of my chair. The author used my passive inverse RIAA network (thank you!) to measure the frequency response of his phono preamp. His measured data for my filter showed huge errors at the frequency extremes. Could this be correct?

So I got out my trusty HP651B oscillator and HP333A distortion analyzer and took my own measurements. I used the 600Ω output of the oscillator and took the −40dB output from the filter using the HP333A as a voltmeter. I double-checked all waveforms on my Tek7704A and a DVM just to make sure everything was OK. Sure enough, my filter was dead on from 10Hz out past 100kHz (*Fig. 2*).

I'm not sure what caused Satoru-san's errors, but I did notice his HP3467A DVM was labeled as an attenuator. Fortunately, this means the frequency response of the DACT CT100 is flatter than shown in the article (no rise in bass).

Jim Hagerman
www.hagtech.com

Satoru Kobayashi responds:

As for the measurement on the reverse RIAA module by the Hagerman company, I did not put any focus on the frequency range below 100Hz and above 10kHz, even though the deviation between the genuine RIAA curve and the measured data was probably caused by the poor setup of measurement. The difference is mainly because of a low level of measuring data and the equipment that I have used, compared to the raw data that Mr. Hagerman measured. I think I used −60dB output, causing me to measure the characteristic stably and precisely.

On the other hand, I guess Mr. Hagerman is very confident in his product's performance, so is probably very much surprised with my measured results. Actually, I was less confident with my measurement of this curve below 100Hz and above 10kHz compared to Mr. Hagerman's measurement. But please note that the description and graph of the reverse RIAA curve attached to the kit itself seems not as good as I could assume the performance to be.

Thus I have measured the characteristic of this kit myself, since I used old equipment such as the HP334A and HP3467A attenuator, which was mislabled DVM.

Finally, I agree that the response curve of the DACT CT100 Equalizer amplifier is very flat, as I could listen to the music compared to the tube-made equalizer with a feedback CR circuit forming RIAA characteristic. So the overall sound impression is very natural, as though the sound comes from the professional base monitor equipment.

SMALL PREAMP

The preamp shown in Fig. 1 of "Another Subminiature Preamp" (Glass Shard, May '01 *aX*, p. 90) will not work. The 220nF cap C1 forms an audio frequency short across the output of V2. It is 1dB down at 10Hz and 3dB down at

FIGURE 3: Stewart's alternate sub preamp circuit.

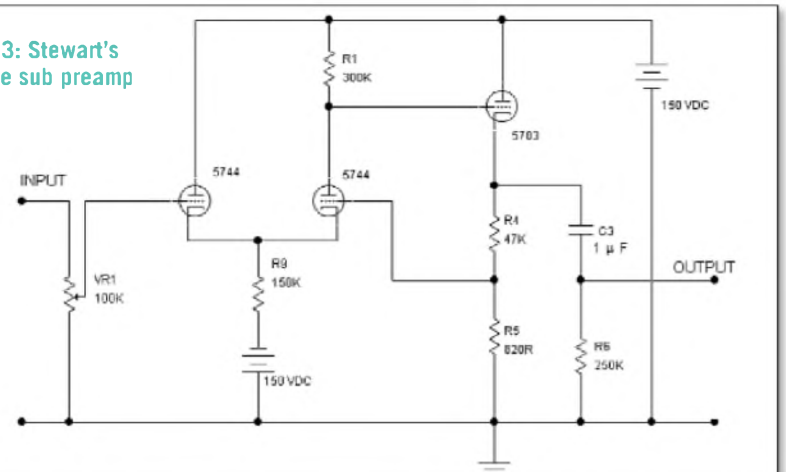


TABLE 1
COMPONENTS USED TO BUILD MR. STEWART'S PREAMPLIFIER CIRCUIT

COMPONENT	VALUE	DESCRIPTION	SOURCE
VR1	100kΩ (102kΩ)	Clarostat 10-turn wirewound pot	Newark Part No. 01F2626
R1	300kΩ (301kΩ)	½W carbon film resistor	Mouser Part No. 293-300K
R4	47kΩ (46kΩ)	½W carbon film resistor	Mouser Part No. 293-47K
R5	820Ω (811Ω)	½W carbon film resistor	Mouser Part No. 293-820
R6	250kΩ (267kΩ)	½W carbon film resistor	Mouser Part No. 293-270K
R9	150kΩ (149kΩ)	½W carbon film resistor	Mouser Part No. 293-150K
C3	1µF (1.05 µF)	"Orange Drop" Vishay polyester capacitor	Mouser Part No. 75-225P100V1.0
Wire		Silver-Plated Copper with Teflon insulation 24 Ga. solid	Newark Part No. 2854/1-100′
Terminals		Turret Terminal tin plated with threaded base	Keystone 1587-2 (Digikey Part No. 1587-2K-ND)
RCA phono jack		Gold-plated, grounded back style	Mouser Part No. 161-4003
Input Differential tubes		Subminiature Raytheon 5744WB	surplus—may be available from New Sensor
Cathode follower tube		Subminiature Raytheon 5703WB	surplus—may be available from New Sensor
Phenolic board/baseplate		3/8" thick Garolite G-10/FR4	McMaster-Carr Part No. 8667K217

17Hz. Perhaps Joel meant the cap to be 220pF, although he indicates 220nF again in the parts list.

He also mentions in the text setting the combined currents of V1 and V2 to one milliamp. That won't happen with a negative supply of only 50V. You would get about ⅓mA. He probably means 150V, in which case the 1mA occurs. This may be merely a typographical error.

The network connecting the plate of V2 to the grid of V3 can be replaced by a single wire to advantage. As shown, the maximum output is limited to about 25V RMS. You will get more than 35V RMS with a direct connection. This will translate as well to less distortion at lower levels.

The constant-current source provided by V4 is of little value in this application and can be replaced by a single 150k resistor. This kind of circuitry is useful in instrumentation amplifiers where you are trying for good common-mode rejection. I've tried it a few times with disappointing results. It looks good on paper, but the added complexity does not translate to improved performance in an ordinary audio amp.

Figure 3 shows what I think would work well.

John Stewart
Ontario, Canada

Joel Hatch responds:

Thank you for noting the incorrect value for capacitor C1. Indeed, I made a mistake when typing up my article. The correct value is 20pF (picofarad).

This gives an approximate –3dB dropoff breakpoint of around 26kHz, not the 2.4Hz as calculated using the 0.22µF.

I took the liberty of building your circuit as shown in Fig. 3. Table 1 *shows the parts list. The only change I made was substituting ±120V power-supply sources versus the ±150V power supplies mentioned. Your circuit performs very well. I had a brief listen, and it sounds fine. I measured the output THD as less than 0.08% for an output of 1V RMS at 1kHz (Stanford Research Systems Spectrum Analyzer, SR770). This is a nice improvement over my circuit. Gain measured 25, ±1dB from 0–3V RMS input signal, dropping off to 21dB at 15V RMS.*

I also noticed that the output waveform, when driven by a square wave at the input, had a large overshoot. I am in the process of seeing whether or not a grid-stopper resistor placed on the input to the cathode follower can reduce this, or whether a bypass capacitor is still needed around the 300kΩ load resistor, R1, of the input differential stage.

Again, thank you for your comments. I especially liked your suggestion to replace the coupling network between the differential stage and the cathode follower with a piece of wire. This is much simpler and works very well.

I used the following test and measurement equipment in evaluating Mr. Stewart's preamplifier circuit.

Signal Generator: Stanford Research Systems SR770 (internal source) and Kenwood AG-253
Spectrum Analyzer: Stanford Research Systems SR770

Oscilloscope: Tektronix TDS-210
Power Supplies: Kenwood PA 18-1.2 (filament supply), Kenwood PR250-0.42A (negative) and Harrison/HP 6207B (positive)
Multimeter: Fluke 8020B
LCR Meter: GenRad Digibridge 1658 (operated at 1kHz)

"GENERIC" AMP

In response to Lance Cochrane's "Classic Circuitry" item (August '01 *aX*, p. 87) regarding the unusual output transformer connections in the "generic amp from 1962": this is actually the way it should be done! What you see is a center-tapped 8Ω output for balanced speaker cable drive. Finally, somebody got it right.

Most amplifiers drive one output and tie the other to ground. This is an unbalanced configuration. What is desirable is to drive the speaker cable differentially (often called bridged); one side +, the other –, balanced about ground.

Ever wonder why they sell tweaks such as wood blocks to raise your speaker cables off the floor? Well, this is the reason. It's because the cable (think of it as a transmission line) is driven unbalanced relative to your supposedly grounded floor. When driven differentially, the stray coupling is balanced and errors cancel out.

Also, feedback is only from the "C" output. The other components form a Zobel network across the 16Ω taps to ensure a stable, resistive 16Ω load at high frequencies (when the cables become inductive). It also damps any ultrasonic ringing in the OPT. There would be no point taking feedback from both taps, because they are out of phase and would cancel each other (one would be positive feedback).

Jim Hagerman
jim@hagtech.com

Lance Cochrane responds:

Actually, I'm guilty of just dashing things off to Audio Amateur far too quickly. The Pilot schematic was one of a group of four schematics that I sent off one day when I was all excited about getting articles published. I remember looking at my blurb and thinking, "Nah, it's just a snubber," but I just sent it. The Pilot 264 is most definitely not a generic amp. I would love to find one.

The Sams Photofact shows AC balance adjustments and DC balance adjustments. The transformers in the photos look really large. Perhaps my phrasing was amiss; I meant that the Pilot was a generic-looking amp. A wolf in sheep's clothing type of product. The photo of the amp that was not published shows a bland-looking rectangular box. Engineers and stylists rarely converse, it seems.

In any case, yes, the feedback is from one tap of the snubber; shame on me for saying otherwise. The publisher usually gives me a chance to proofread prior to publishing, but in this case failed to do so. (That makes it all their fault.)*

I do think you are overdoing the balanced business a little. Balanced configuration is great for lowering noise for long runs, but does not, to my ears, do a whole lot more.

About the tweaks and wood blocks, it's really not like that at all. The guys who sell that type of product are just selling dreams. That is what a lot of hi-fi is and always has been. I don't have a problem with the tweakmeisters of the world. They have provided us with some wonderful silliness at times and that can make it fun. If you like wood bricks, that's just fine with me.

** Author checks on materials is normal for original articles, but not for classic circuitry reprints.—Ed.*

HI-FI BUILDER

I am in the process of building my own hi-fi system—from turntable to speakers. So far, I have built the speakers, which use Audax drivers:

10″ HDA bass
4″ HDA midrange
HD3P elliptical gold dome tweeter

The turntable has been in development for over two years and is my own design. I am hoping to go into small-scale production of it.

The preamp I am building is one of your designs featured in March 2001 *aX* ("High-Quality ($180) Control Unit," p. 36). I have had two alloy boxes made for the preamp and power supply, which I needed to modify for UK use. I have also included a regulated 12V DC heater supply, running off a 15V toroidal transformer.

I have recently had a chance to listen to your preamp in my system, and compared it to an Audio Innovations product. Two things were very noticeable: The bass and detail were superior in your own design. I was impressed!

Steve Rooke
Somerset, England

Joseph Norwood Still responds:

I was very happy to hear you enjoyed the performance of the preamp published in aX. The modifications you made to the power supply are indeed an improvement, and locating the supply in a separate chassis and using toroidal transformers indicate you did your homework. The preamp, as you realize from your construction efforts, is not an easy project, and I'm pleased you had a successful outcome. I assume you noticed that the value of C7 is 470µF and not 100µF as shown on the schematic.

I appreciate your comment that the sound was superior to the Audio Innovations preamp. I have no knowledge of commercial products, so it is gratifying to have an analysis from an informed person such as yourself to make such a flattering comparison.

Your project of building a speaker system, a self-designed turntable, preamp (control unit), and power amplifier is very impressive. I'm sure you will have a great system! You are definitely a true audiophile.

LED EFFECTS

In the article "Odds and Evens" (July 2001 *aX*), it appears that D1 in Fig. 11 is oriented incorrectly. The cathode should be connected to ground for it to serve as a bias voltage source.

Additionally, the technique of using an LED as a reference on a simple class-A gain stage is one that I have long used to actually introduce nonlinearities into a circuit by adding musically useful harmonics. A bias LED is not a steady reference voltage. As the signal voltage swings higher or lower, the Vf of the LED will change in proportion to the current producing exactly the warping shown for the triangle wave in Fig. 8. I would never recommend this technique in a gain stage where more linear response is the desired operating condition.

Jack Orman
http://www.muzique.com

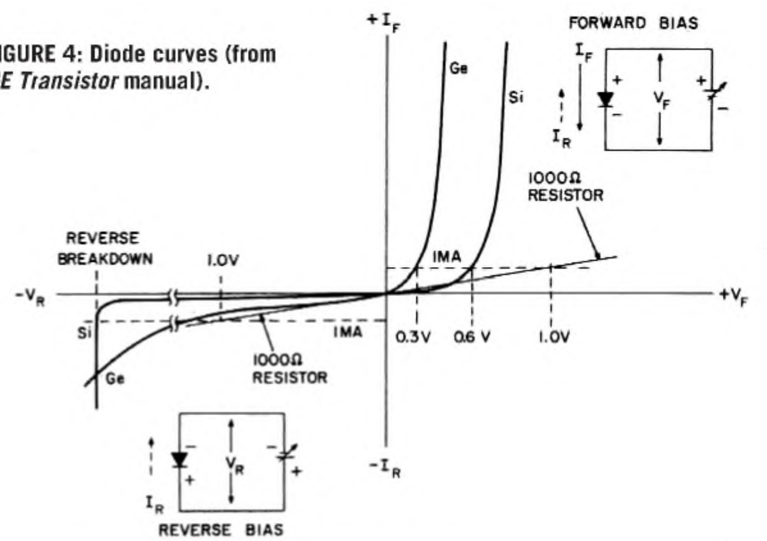


FIGURE 4: Diode curves (from *GE Transistor* manual).

Graham Dicker responds:

Jack, you are quite right about the direction of the LED on the schematic in Fig. 11. As it turns out, the error was picked up during proofreading, but as Murphy's law would have it the errant circuit was still published.

The original idea in using the LED for bias is to get rid of the pesky and often ill-sounding bypass electrolytic in the cathode of the circuit. As the voltage gain of the 807 is around 11, the required P-P grid drive from the voltage amplifier is 45V, which is 15% of the total available voltage swing of the stage. The quiescent current is 1.5mA, so the anode (and cathode) current will change from 1.3875mA to 1.6125mA. This would result in a change in voltage across a 3k3 cathode bias resistor of about 750mV. Using these same currents with a 5MM round 3.8Mcd red LED results in a change of 11mV. This represents a much more stable bias voltage for the stage.

Figure 4 *shows that the slope of the diode curve is steeper than the plotted 1k0 resistor, which indicates it is close to a constant-voltage device.*

Jack's idea to use an LED to introduce even harmonic distortion is a good one, if you wish to build a musical instrument amplifier. After having built a few guitar amplifiers in my time for customers who reported the sound as too clean and requested that I dirty it up a bit, I wish I had thought of the idea earlier. So why does the LED have a good effect on the linearity of my amplifier, yet cause lots of good second harmonic distortion in a musical instrument amplifier? The answer lies in the transfer curves of an LED, or, in the case of Fig. 4, *the diode curves from the* GE Transistor *manual.*

The portion of the curve that my voltage amplifier uses is the linear region; for an LED the maximum current is around 10mA. Selecting a 1.5mA bias current places it in the middle of the linear part of the LED transfer curve. Also, because I am using only 15% of the available voltage swing, the change in current and voltage across the LED is small.

The norm for a quality guitar amplifier is to have two gain controls—one in the front end as an overdrive, the other as a master volume. If you increase the input gain and decrease the master volume, you end up with a nicely overdriven input stage in which the stage output voltage is maximum (and often clipped). If you set the master gain to maximum and the input gain to the level required, the stages produce significantly less P-P (and linear) output voltage.

If you use an LED in a musical instrument amplifier, you can operate over the entire—including the nonlinear portion—of the LED transfer curve, which produces lots of second harmonic distortion.

If you wish to optimize this effect, you can run the input stage down to anode starvation by using a high value plate resistor and select a Q point down into the middle of the nonlinear section of the LED transfer curve. This is in itself not a bad idea, as it will also maximize the stage voltage gain. A good starting point is a plate current of 75µA and a plate resistor of 2M2. Make sure that the following grid load resistor is changed as well. With many tubes you will need to watch how high a value grid resistor can be used, as it will affect the noise, stability, and frequency response of that stage.

Because the maximum stage P-P output swing is obtained when RK=(RA+RL)/mu for this circuit, and to meet the bias conditions of the Q point, a cathode resistor of 3k3 would be required.

TESTING SUB AMPS

As an amateur speaker builder, I've read Tom Perazella's article ("Low Down Power, Part 2," June '01 *aX*, p. 52) with much interest. In general, I think it is a very nice article with a good evaluation of the presented sub amps.

However, when Mr. Perazella describes the listening test using two very (mechanically) different subwoofers, there is something I don't understand: He uses one enclosure for both subwoofers, producing a mismatch for both woofers (V_{AS} and the resulting f_B). The results of his listening test support this mismatch—"no floor shake" (and in some cases also some compression, due to the small enclosure), as he says. This is a correct conclusion for a system with an f_B of 40–43Hz.

In my opinion and experience, I believe it would have been better (and I suppose most sub builders would have done so) if he had constructed two different enclosures, matching the mechanical properties of the subwoofers. The results would have been more representative.

I'd like to know Mr. Perazella's reasons for performing a listening test this way.

Edwin Slee
eslee@attglobal.net

Thomas Perazella responds:

Thanks for your letter on the sub amp article. I understand your concern about not producing an "optimum" enclosure for each of the drivers tested, but determining what would be optimum is a matter of choice that could fill another article in itself, if not start a feud among the ported vs. sealed vs. horn vs. transmission-line camps. For example, a tuned port enclosure would provide more bass at some frequencies, but at the expense of a faster rolloff below the port tuning frequency. There is also the matter of power handling of the driver at very low frequencies.

The main purpose of this test was not to produce the "best" result for the two drivers used, but rather to provide a comparison between two different loads for the amps under test. Also note that although the f_s and V_{AS} specs were different for the two drivers, in the sealed box used, their performance came closer together. However, again, the goal was not to optimize each, but rather to see how the amps reacted with both.

As far as floor shake, in reality, I consider the floor shake region to be below 20Hz. In my loft, for example, there is a resonance around 15Hz that if excited by a sub can produce some interesting sensations. With the sealed box enclosure, the hope was that there would be enough residual energy in the lower ranges to produce some of that effect. The amps that had a bump in response at around 30Hz, but rolled off below that, would produce more bass, but less floor shake before going into clipping. Alas, there simply was not enough volume displacement from either driver with any amp to produce substantial floor shake.

Remember that if you want to maintain the same SPL for every octave lower that you are trying to reproduce, you must have four times the volume displacement. That's why my sub uses eight 12″ woofers in a 450ft^3 infinite baffle enclosure. Having the best sub amp in the world will not make up for a lack of volume displacement from the driver or drivers used.

The net result of the test was that, except for the very low frequencies that were beyond the displacement capabilities of the drivers used, the amps did very well. To see what is available from commercial pieces at those very low prices was a real eye—and ear—opener. The prices were low enough that you should have money left over to buy one or more of the excellent value drivers that are available. And remember, if you are going after very low frequencies, unless you have a lot of long excursion drivers, you will probably run out of excursion before amp power. I hope this answers some of your questions.

CONSOLE SOUND

I was not going to renew my subscription to *audioXpress*, until I read the article on the Westinghouse radio model #H-169 by Larry Lisle (August '01 *aX*, p. 54). These heavy, high-quality, multi-band hi-fi sets from the late 1940s have been my specialty for many years. As a recently degreed engineer, I still become fascinated by the incredible workmanship and sound quality that some of these old sets have.

Like Mr. Lisle, I, too, grew up in the 1950s listening to my parents' large combo from 1949. It just so happens that I recently restored an H-169 for a good friend of mine; a friend who has once again become a daddy. So, with this "great radio" in their livingroom, his little girl will now grow up hearing music as it really should sound!

Let us not forget some of the other "great radios," such as the Capeharts, the Hoffmans, the Magnavox series (i.e., the #155B/Regency Symphony/Belvedere or the #300BF), the E.H. Scotts, or the Stomberg-Carlson, (i.e., the 1135 (1947) or the SR-406, (1957)), just to name a few.

So, keep up the good work. These old "great radios" are truly a genuine treasure.

Michael K. Tremper KC2HFR
Cheektowaga, N.Y.

HELP WANTED

I am writing to ask whether any reader has a source for repair or replacement of the achilles heel of the venerable AR turntable: the headshell. I presume there are no more headshells available at a reasonable price, and I know of no replacement. I tried to repair the threads, which have worn down over time. The plastic repair kit that has worked in other circumstances was not strong enough for this project. I suspect there is a way to repair it, but I have yet to discover the answer. I would appreciate any help my fellow readers could offer.

Lloyd Foster
lfoster@classicwineimports.com

I am looking for a resource for Mogami 2534 wire—a place where I can buy bulk wire as in a 100′ roll.

Bill Penninger
Bill.Penninger.us.bosch.com

Try www.usedcable.com—Ed.

Readers with information on these topics are encouraged to respond directly to the letter writers at the addresses provided. –Eds

Classifieds

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Ribbon tweeters: one pair Newform R30 ribbons (www.newformresearch.com) at half of factory cost—like new, in original packaging, used only for concept development, $340/pair. David, nabada@qwest.net.

Altec: 290 Alnico drivers, $375/pair; 808 Alnico drivers, $250/pair; 811 horns, $175/pair. Stephens 12" woofer, $75; Gauss 1" driver, $100; JBL 2416H drivers, $225/pair. Harman-Kardon CD91 cassette, $75; AR10 (Pi) speakers, 60 lbs each, $475/pair, extra mid $60. David 854-688-5024.

Assemblage L-1 Platinum version preamp, $600, and DAC 3.0 Signature version (has both Burr Brown 1704 24/96 and HDCD PMD 100 digital filters), $900. dave.pit@home.com or 905-819-8462.

WANTED

Hobbyist seeking pre-recorded, 2-track reel-to-reel tapes—7.5 or 15 ips. Can be 1950s/1960s commercial product or old studio dubs. Andy Pennella, 203-329-7498 or andrew.penella@pfizer.com.

Can anybody help with schematic or CAT (Convergent Audio Technology) Preamp SL 1 (hand-drawn is ok). Reinhard Hoffmann, Wiembecker Str. 1, 32657 Lemgo, Germany, phone/FAX 05261 87586, in.audio@t-online.de.

Wide bandwidth 15Ω loudspeaker Axiom 80 from Goodmans—even not operating. Please make an offer at reasonable cost (incl. transport). C. Bourdel, 2 Allee Soulezard 95100 Argenteuil France, 33 1 34 11 16 26.

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TEST TRACKS— What are your favorite music selections you use to test audio gear or to show off some special capability of your system? Share your favorites with others. Simply describe your seven top pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to the address above. We will pay a modest stipend to readers whose submissions are chosen for publication.

Driver Report

The Scan-Speak 18W/8531G00

By Vance Dickason

I recently examined a new product from Scan-Speak, now one of the companies that is part of the new Vifa/Scan-Speak/Peerless (Denmark) organization, DST (Danish Sound Technology). The 18W/8531G00 is a 6.5″ version (more or less) of the Scan-Speak 15W 5.25″ mid/woofer, used in numerous high-end loudspeakers. The woofer is built on a newly designed cast frame with very minimal reflection areas, very similar to the 15W frame, with totally open area below the spider (the voice coil is visible in this configuration). With 16mm from the spider mounting flange to the front plate, the frame is designed to accommodate very long rear excursion.

The cone assembly combines a paper cone that appears to be the same material and wedge construction as the 15W cone, which is to say that the cone is cut in a wedge spiral fashion with ten cuts across the cone surface that are glued back together. The object of this attention to detail is purportedly to reduce cone break-up mode resonances. This same technique is also used with the dustcap, which has two slits that are glued back together.

Completing the assembly is a rubber surround and 4″-diameter flat spider. Driving this assembly is a 1.5″-diameter voice coil wound on a non-conducting former. Other features include gold terminals and a back plate bumpout for rear voice-coil excursion.

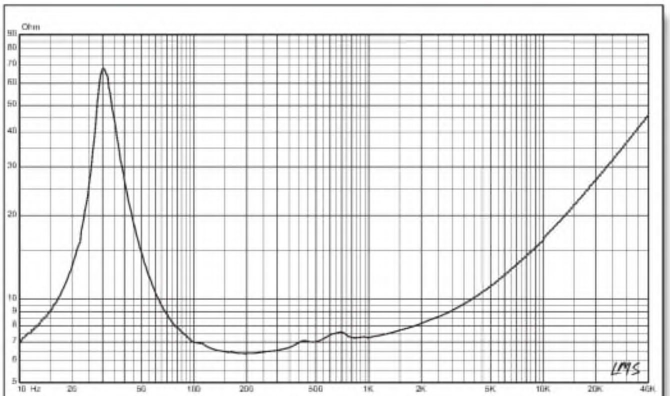

FIGURE 1: Scan-Speak 18W8531G00 free-air impedance plot.

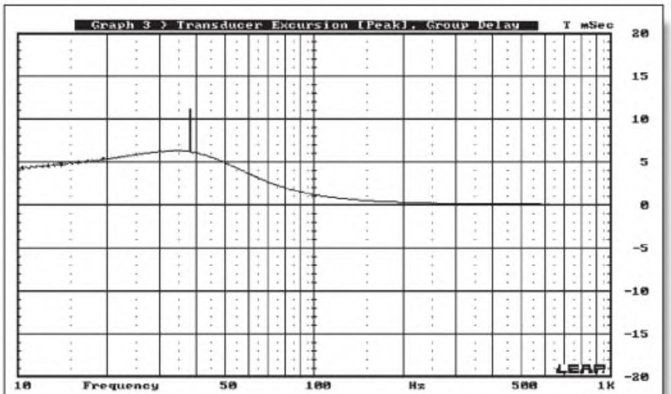

FIGURE 3: Group delay curves for the 2.83V curves in *Fig. 2*.

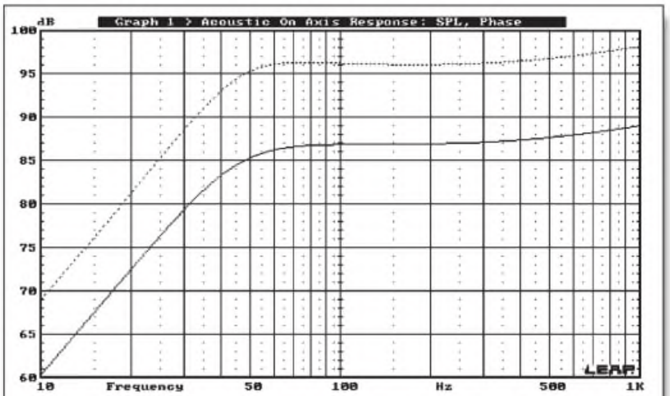

FIGURE 2: Scan-Speak 18W8531G00 computer box simulation (solid = sealed box @ 2.83V; dot = sealed box @ 9V).

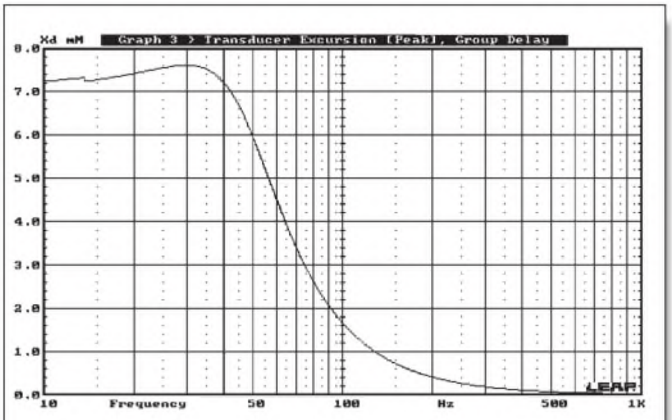

FIGURE 4: Cone excursion curve for the 9V curve in *Fig. 2*.

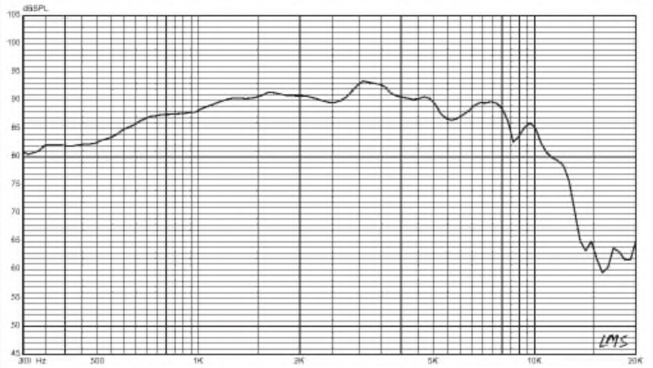

FIGURE 5: Scan-Speak 18W8531G00 on-axis frequency response.

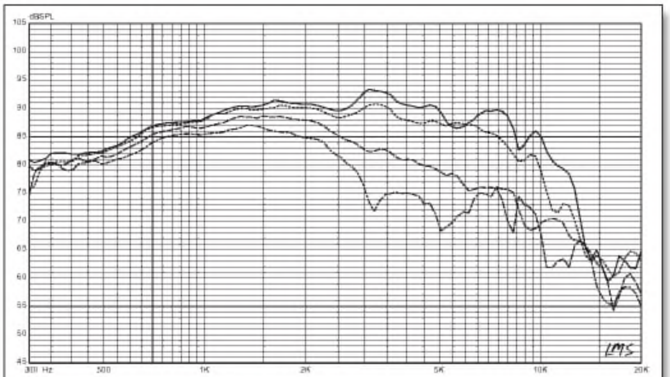

FIGURE 6: Scan-Speak 18W8531G00 on- and off-axis frequency response (0° = solid; 15° = dot; 30° = dash; 45° = dash/dot).

I began testing by running both free-air (*Fig. 1*) and delta-compliance (test box) impedance plots. I loaded this data into the LinearX LEAP software, calculated the operating parameters, and compared them to the factory data in *Table 1*.

Since there was a significant variance between the measured Q_{ES} and the factory data, I compared a sealed box simulation with the factory data and the measured data. While damping was obviously different, performance was an f_3 of 45Hz for the factory data and 42Hz for the measured data, so the difference is not as extreme as the numbers might first indicate. Given this, I proceeded to use the measured data to perform a sealed box simulation in a 1.2ft^3 enclosure with 50% fill material (fiberglass). *Figure 2* shows the results with the 2.83V curve yielding an f_3 of 41.6Hz with an enclosure Q_{TC} of 0.72.

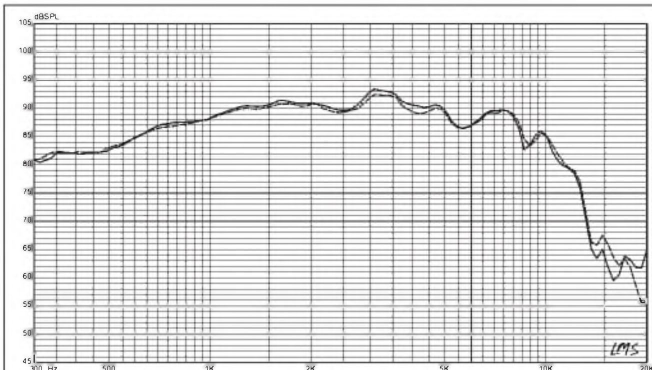

FIGURE 7: Scan-Speak 18W8531G00 two-sample SPL comparison.

Increasing the simulation input voltage to achieve a maximum cone excursion equal to X_{MAX} + 15% (7.48mm for the 18W), it took 9V to achieve a maximum linear output of 96.5dB, which is not bad for a 6.5″ woofer (see *Figs. 3* and *4* for the associated group delay and cone excursion curves). Two of these woofers in parallel would likely produce very clean output on program material up to 108dB with a sealed box f_3 that is considerably lower than most 6.5″ woofers.

TABLE 1
SCAN-SPEAK 18W/8531G00

	SAMPLE A	SAMPLE B	FACTORY
F_S	31.5Hz	30.7Hz	27.5Hz
R_{EVC}	5.85	5.89	5.80
Q_{MS}	5.95	5.83	5.0
Q_{ES}	0.52	0.53	0.39
Q_{TS}	0.48	0.49	0.36
V_{AS}	48.3 ltr	51.7 ltr	61 ltr
Sens.@2.83V	87.3dB	87.2dB	87.0dB
X_{MAX}	6.5mm	6.5mm	6.5mm

I then loaded the 18W (surface mounted, not recessed) into a small enclosure with a 15″ × 7.5″ baffle area and filled with damping material (again, Mahogany Sound's Acousta-Stuf). I generated on- and off-axis response curves at 2.83V/1m and illustrated the on-axis response in *Fig. 5*. The curve is about as smooth as I have seen (in recent memory) and would likely yield a system response in its passband of ±1.5dB or so. *Figure 6* shows the on- and off-axis response out to 45°.

The –3dB at 30° off-axis with respect to the on-axis curve occurs at 2.1kHz, so a high-quality dome capable of a moderately low (for a 1″ dome) frequency would be well suited to this product. *Figure 7* shows the two-sample SPL comparison with the typical high level of matching always present with Scan-Speak drivers. For more information on this high performance 6.5″, visit the company website at www.vifa-scan-speak.com. Several suppliers stock the Scan-Speak line. ❖