Vintage Dutch Tube Amps Uncovered

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The Tuba 18 Sub

By Bill Fitzmaurice

Back

This small horn-loaded subwoofer delivers all the bass power you need for home use or in the car.

PHOTO 1: THE TUBA 18 SUB.

After prototyping the Tuba 24 pro-sound sub (April '04 *aX*, p. 34), I was intrigued at the thought of building a diminutive Tuba, one small enough for home or auto-sound use. As my intended outside dimension of approximately 18 in^3 is an unheard-of size for a horn-loaded sub I used the McBean Hornresp.™ simulation program to see whether it was at all feasible. The mouth would need to be very small, so I modeled in ⅛ space for corner loading.

To support loading to 40Hz the horn path would need to be at least 7′ long, and that would result in a very small rear chamber, too small even for a ten. But despite my own misgivings the simulation results indicated that the concept would work. I only needed to find an appropriate driver.

DRIVER SEARCH

I gave up on 6½″ drivers early on, as I could find none at a reasonable price with the requisite Xmax spec, which I figured would need to be at least 5mm. I looked at many eights as well, but understandably there aren't many designed for sub duty which is primarily the domain of twelves and fifteens. But one driver—the MCM #55–2421 8″—had just the specs I needed. This woofer has a claimed Fs of 25Hz, Qts .18, Vas 39 ltr, SPL 87dB, 120W power rating, and, most important, a 16mm Xmax.

My sample turned out to have a measured Fs of 30Hz, which is even better for horn loading, which pushes the effective Fs to a much lower Fs(h).

PHOTO 2: RABBETS CUT INTO ONE SIDE.

FIGURE 2: INTERSECTING PARTS.

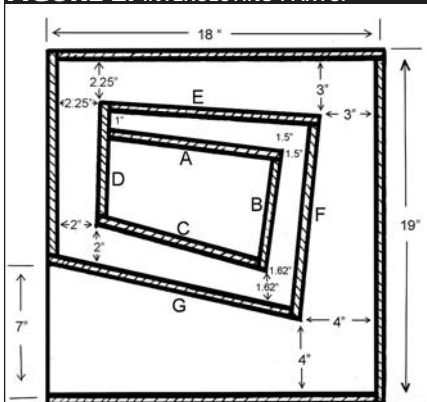


FIGURE 1: TUBA 18 SPL CHART.

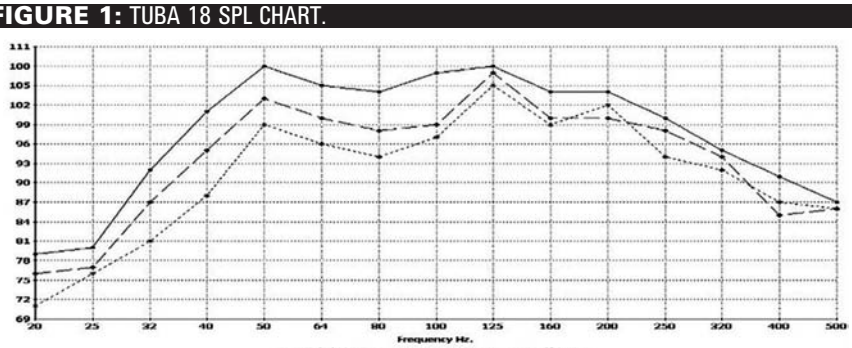
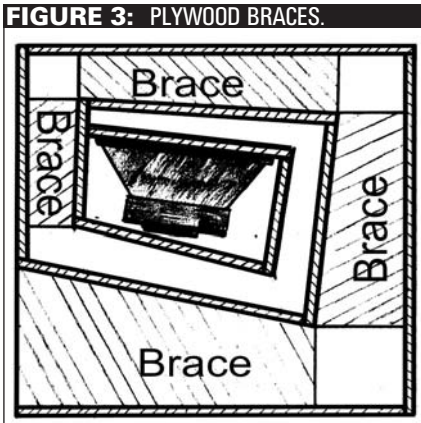


FIGURE 3: PLYWOOD BRACES.



Interestingly, Koonce and Wright considered this driver for their VB sub project (*aX* 5/04 and 6/04), but passed on it due to slightly better measured specs from another driver. I found that for a horn-loaded sub its specs were unsurpassed, and at less than $30, shipping included, the price is an absolute steal.

The lower passband required from a home theater/hi-fi sub, as opposed to the pro-sound Tuba 24 model, required another foot of horn path despite a smaller footprint, which I achieved by using more bends in the horn. The horn mouth ended up far smaller than I would have preferred, but to make it larger would have cost horn length. In the end that turned out to be no problem. As a pro-sound sub, Tuba 24 would more often than not have no available boundary reinforcement; taking that into account, I made its mouth-to-path-length ratio very high for maximum SPL in a freestanding placement.

With Tuba 18 my intent from the beginning was to corner-load, so I made the mouth-to-path-length ratio small. This hurts response in a freestanding location, but since corner placement adds both additional path-length and mouth area, the resulting combination of response and sensitivity (*Fig. 1*) is one not normally associated with either an 8″ driver or a 3.3ft^3 box (*Photo 1*).

CONSTRUCTION

Construction of the Tuba 18 is nearly identical to the Tuba 24. The material is ½″ plywood; if weight is of no concern ½″ MDF or particleboard will work just as well. You can use heavier materials for the outer shell parts, as long as you do not alter the internal clearances. Do not use horn plates thicker than ½″, however, or the driver chamber will end up too small.

While you may opt to simply screw and glue the parts together, I used a different technique with the prototype. I routed ½″ wide by ⅛″ deep rabbets on the sides at all the joints, glued the joints with polyurethane construction adhesive, and slid the parts into place, securing them with 1″ brads while the adhesive cured. This method produces a very strong joint, though the routing process is time consuming.

I did this primarily to see how long it would take to assemble the cabinet from pre-cut pre-routed parts, as would be the case in a factory. You can do it this way yourself if you wish, or stick with simple butt joints. By the way, with pre-cut parts the assembly time was less than an hour.

To start construction, cut out the two sides. Draw the locations of all the intersecting parts (*Fig. 2*) on them, starting with the front, top, back, and bottom, and from there the horn plates, starting at the junction of plate G where it joints to the front. The internal measurements shown are the pathway clearances at each horn bend; the panel dimensions will depend on the actual measured thickness of the material used, as there is no such thing as ½″ plywood that actually measures ½″ thick.

If you are rabbeting the joints, do so now (*Photo 2*) using a guide board clamped or screwed to the side to act as a guide for a pattern following router bit (*Photo 3*). When you cut the rabbets have them run all the way to the cabinet edge so that you can slide each part in turn into place.

ASSEMBLY

Cut enough material for all of the remaining parts at one time, so that the table saw rip fence will not be moved and all parts will be of identical width. Since the cabinet is 18″ wide, the nominal width of all the parts is 17″, but if you use the rabbet joint technique, add another ¼″ to the parts width. Cut the parts to length one by one as each is installed, using dead reckoning to determine their true lengths. Start with plate A, going through plate G, and

PHOTO 3: CUTTING RABBETS WITH A ROUTER ATOP A GUIDEBOARD.

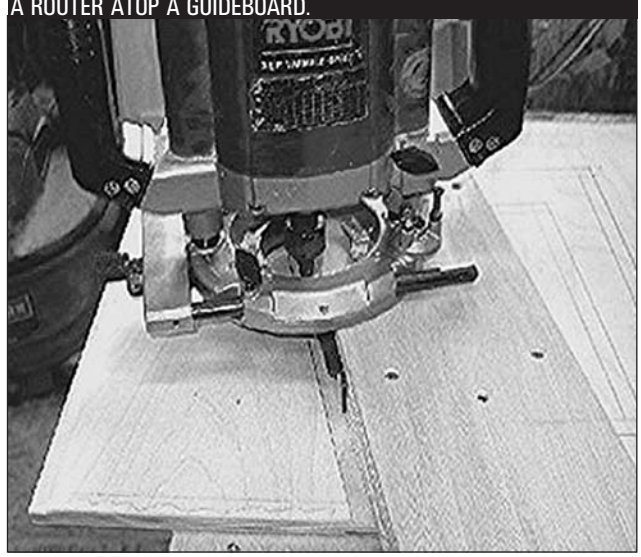


FIGURE 4: TUBA 18 IMPEDANCE CHART.

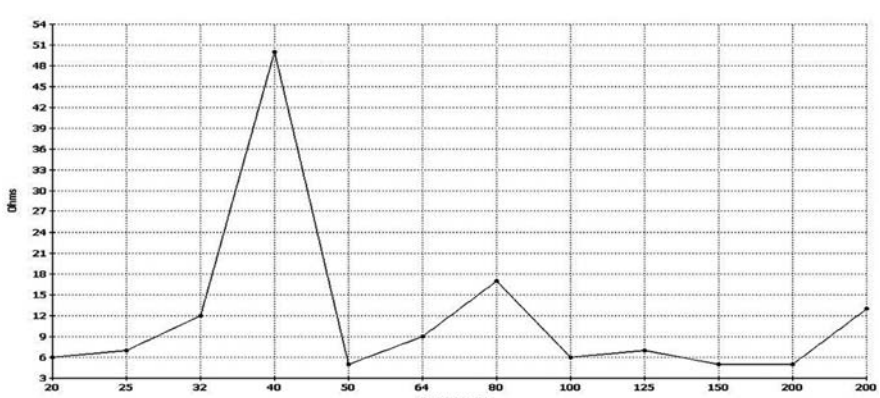


PHOTO 4: ATTACHING PLATE A TO ONE SIDE.

then in turn the front, top, back, and bottom. Plate A also serves as the baffle, through which you cut a hole for the driver.

Install "T"-nuts for driver mounting if you wish, though screws are adequate. The angles where the plates intersect are slight enough so that cutting the plates at an angle is not required; the expanding polyurethane adhesive will fill any voids. The exception is the junction of plate G and the cabinet front, where the angle is severe enough to require cutting to suit. Begin attaching the plates (*Photo 4*). With butt joints, use a straight guide-board and clamps to hold the pieces together in proper alignment while drilling pilots and driving screws. With rabbet joints, keeping things aligned while attaching the first parts is easy if you use another part as a temporary spacer (*Photo 5*).

PHOTO 5: ATTACHING THE SECOND SIDE TO THE ASSEMBLY.

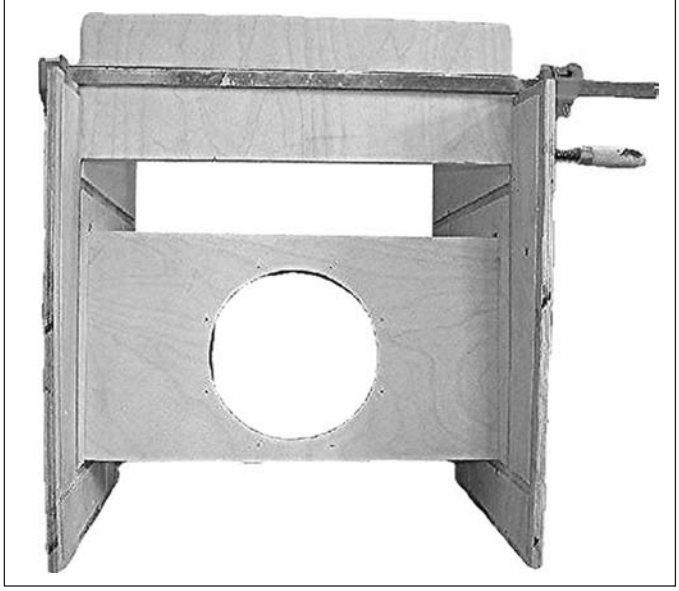
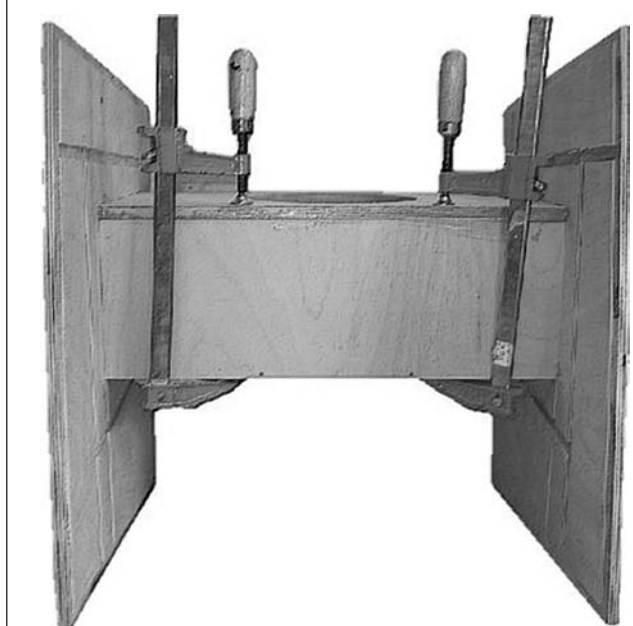


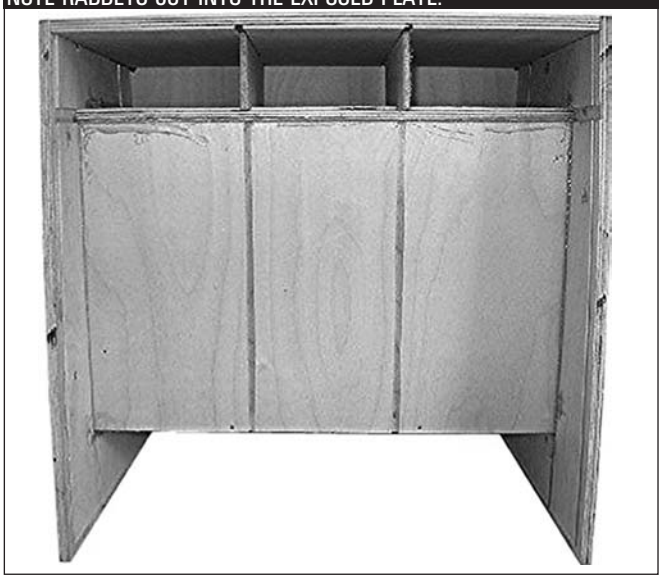
PHOTO 6: CLAMPING PARTS FOR ASSEMBLY.



As you add more parts, use clamps to hold them in alignment until you nail them (*Photo 6*).

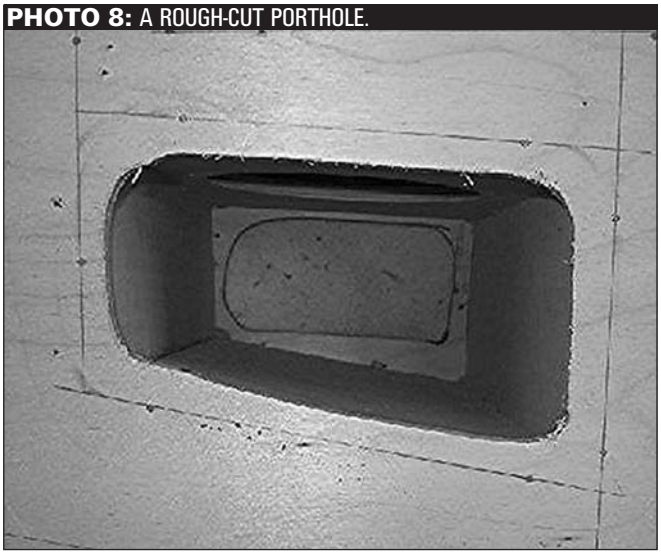
There are ¼″ plywood braces between the outer horn plates and the front, top, back, and bottom (*Fig. 3*), approximately dividing the panels into thirds, their sizes determined by dead reckoning. They slide into ⅛″ deep rabbets cut into their mating parts either on a table saw or a router table. Make sure before you install the braced parts that you cut the necessary rabbets into them (*Photo 7*), and thoroughly glue the rabbets as you install the braces.

PHOTO 7: BRACES IN PLACE; NOTE RABBETS CUT INTO THE EXPOSED PLATE.



The porthole openings are framed by plates A through D. Determine their locations by the nails or screws through the sides going into those plates. Rough-cut the porthole opening (*Photo 8*) with a saber-saw, then use a flush-trimming router bit to final-trim the opening to the plates (*Photo 9*). There are portholes on both sides of the cabinet to provide access to the driver mounting bolts/screws. Speaking of the driver, now's a good time to make sure it fits (*Photo 10*).

PHOTO 8: A ROUGH-CUT PORTHOLE.



From the opposite side of the box, reach through the driver chamber to trace the opposite porthole cover on a piece of plywood, cutting the cover ½″ oversize all around. Use a router to cut a ¼″ deep flange on the porthole covers (*Photo 11*) and also on the cabinet sides (*Photo 12*) so that the portholes fit flush to the cabinet sides. Apply the finish of your choice and then install the driver, wiring it through one of the porthole covers, thoroughly sealing the hole. Weatherstrip the flanges and install the porthole covers.

Note that horn loading the 4Ω driver gives an average impedance over 6Ω (*Fig. 4*), so this is not a difficult box to drive. Also note that the rear chamber is not lined or stuffed. I tested the cabinet with and without stuffing (*Fig. 5*) and found that damping materials actually hurt performance.

The best placement for the box is

PHOTO 10: TIGHT SQUEEZE!

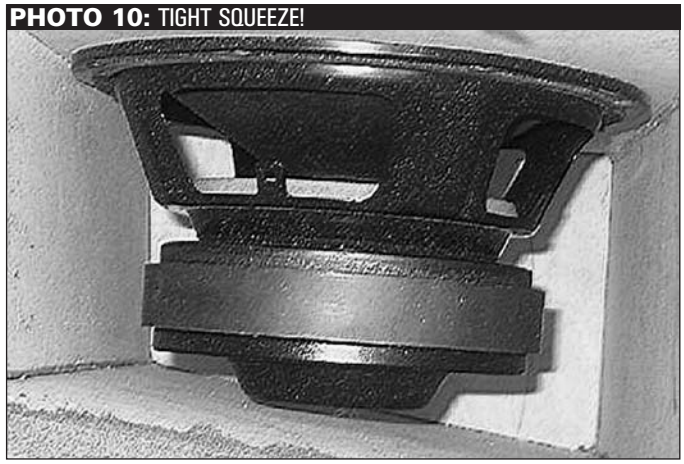


PHOTO 9: A FINISHED PORTHOLE.

PHOTO 11: A FLANGED PORTHOLE COVER.

facing into a room corner, tight to the walls. Alternately, it will work almost as well facing the wall from about 6″ distance in a center of wall placement. If you currently have a non-horn-loaded sub, you'll notice a big difference with Tuba 18. I used mine to replace a pair of $2ft^3$ sealed boxes loaded with twelves, and even after reducing the amp output by some 10dB the Tuba still delivers a low-frequency extension of which the twelves were never capable. You could easily drive this sub with SET power. If you have a variable low-pass filter on your sub amp, a setting of 40 to 50Hz gives the best result in smoothing response.

While intended primarily as a home/auto sub, this box still has the requisite efficiency and power handling required for pro-sound in small venues, or for permanent installations such as churches where very high SPL is not required. It matches well with my DR200 (August '04 *aX*, p. 24) for those of you who want a high-efficiency full-range PA from the smallest possible cabinetry.

As for auto-sound use, I substituted my Tuba 18 for my regular car sub for a couple of days and it was amazingly better in low-frequency extension, sensitivity, and total output. If you want bass power that will leave the local teenagers slack-jawed in disbelief, stick one of these in your trunk. It will deliver the goods. *aX*

PHOTO 12: FLANGING A PORTHOLE WITH A ROUTER AND GUIDEBOARD.

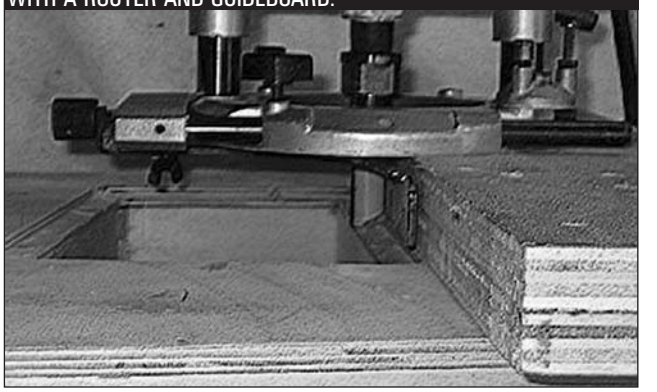
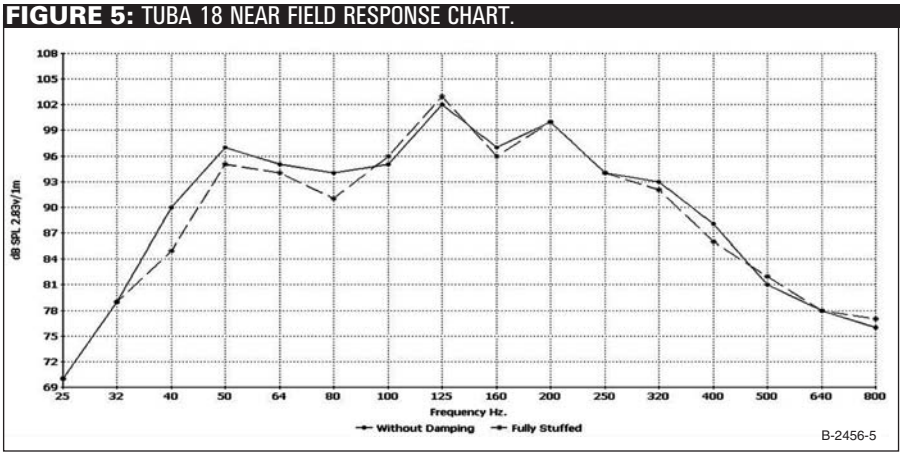


FIGURE 5: TUBA 18 NEAR FIELD RESPONSE CHART.



Tubes
Vintage Glass

Back

Recently I acquired some old Dutch books on electronics which contained the following tube amp schematics:

1. Single-ended 4W amplifier
EF94, EL90

This 4W amp (*Fig. 1*) has a rather primitive type of tone control. "Laag" means bass, "hoog" treble. The amp was used to amplify the signal of a turntable with a crystal cartridge.[1]

Push-pull amplifiers

2. "Fidelio" amplifier (*Fig. 2, Table 1*) ECC83, ECC85, 2 × EL84, 5Y3GT
Power output specified as maximum 10W[2].
3. 70W versterker (*Fig. 3, Table 2*) 2 × EF86, ECC40, 4 ×

FIGURE 1: 4W AMPLIFIER.

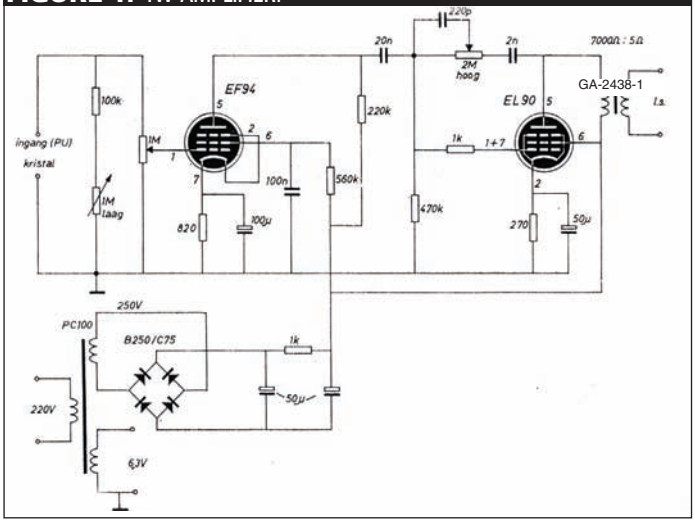
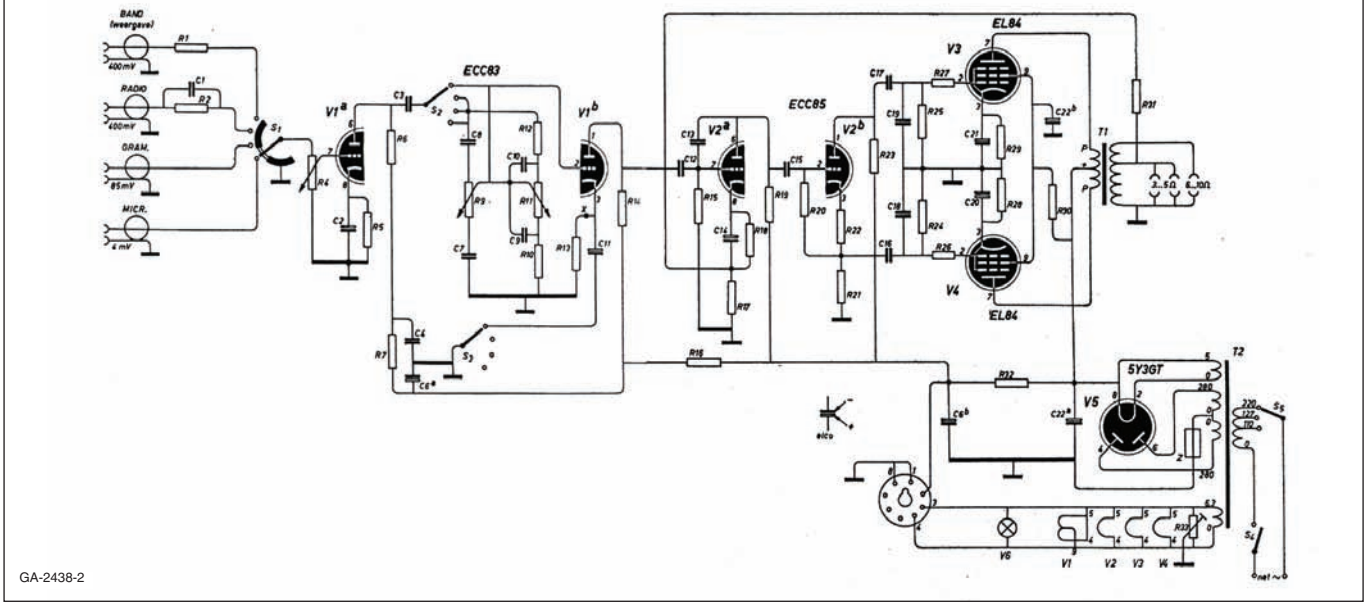


TABLE 1

Schakeling en onderdelen van de fidelio

C1	polystyreen cond. 100pF (Amroh/Mial)
C2-11-14-20-21	elektrol. cond. 100µF/15V (Amroh)
C3-4-12-15	polyester cond. 0.022µF (Amroh)
C6a-b-22a-b	elektrol. cond. 32 + 32µF (Amroh)
C7	polystyreen cond. 5000pF (Amroh/Mial)
C8	polystyreen cond. 250pF (Amroh/Mial)
C9	polystyreen cond. 0.01µF (Amroh/Mial)
C10	polystyreen cond. 1000pF (Amroh/Mial)
C13	polystyreen cond. 50pF (Amroh/Mial)
C16-17	polyester cond. 0.047µF (Amroh)
C18-19	polystyreen cond. 150pF (Amroh/Mial)
Alle	½W weerstanden "Vitrohm"
R1-2	820kΩ ½W
R4	potmeter 220kΩ (Amroh nr. 54.835)
R5	5.6kΩ ½W
R6	22kΩ 1W
R7-14	47kΩ 1W
R9	potmeter 220kΩ m. schak. (Amroh nr. 54.925)
R10	15kΩ ½W
R11	potmeter 1MΩ (Amroh nr. 54.827)
R12	270kΩ ½W
R13	3.9kΩ ½W
R15-20	1MΩ ½W
R16	33kΩ 1W
R17	680Ω ½W
R18-22	2.2kΩ ½W
R19	220kΩ 1W
R21-23	100kΩ 1W
R24-25	470kΩ ½W
R26-27	1kΩ ½W
R28-29	270Ω 1W
R30	2.7kΩ ½W
R31	3.9kΩ ½W
R32	12kΩ 1W
R33	ontbrompotm. 100Ω (Amroh)
S1-2-3	kiesschak. (Amroh 48.086 of 48.118)
S4	netschak. Op R9
S5	spanningsoverzetter (Amroh)
T1	bal. Uitg. Transf. U73U (Amroh)
T2	voed. Transf. P141NN (Amroh)
V1	buis ECC83 (RFT/Amroh)
V2	buis ECC85 (RFT/Amroh)
V3-4	buis EL84 (RFT/Amroh)
V5	buis 5Y3GT
V6	schaalverl. Lampje 6V/0.1A
Z	smeltveiligh. 200mA

FIGURE 2: 4W FIDELIO AMPLIFIER.



EL34, 2 × AX50 Power output specified as 70W at 1.2% THD, line input sensitivity 290mV, microphone input sensitivity 1.8mV.[3]

Aren van Waarde
The Netherlands

REFERENCES

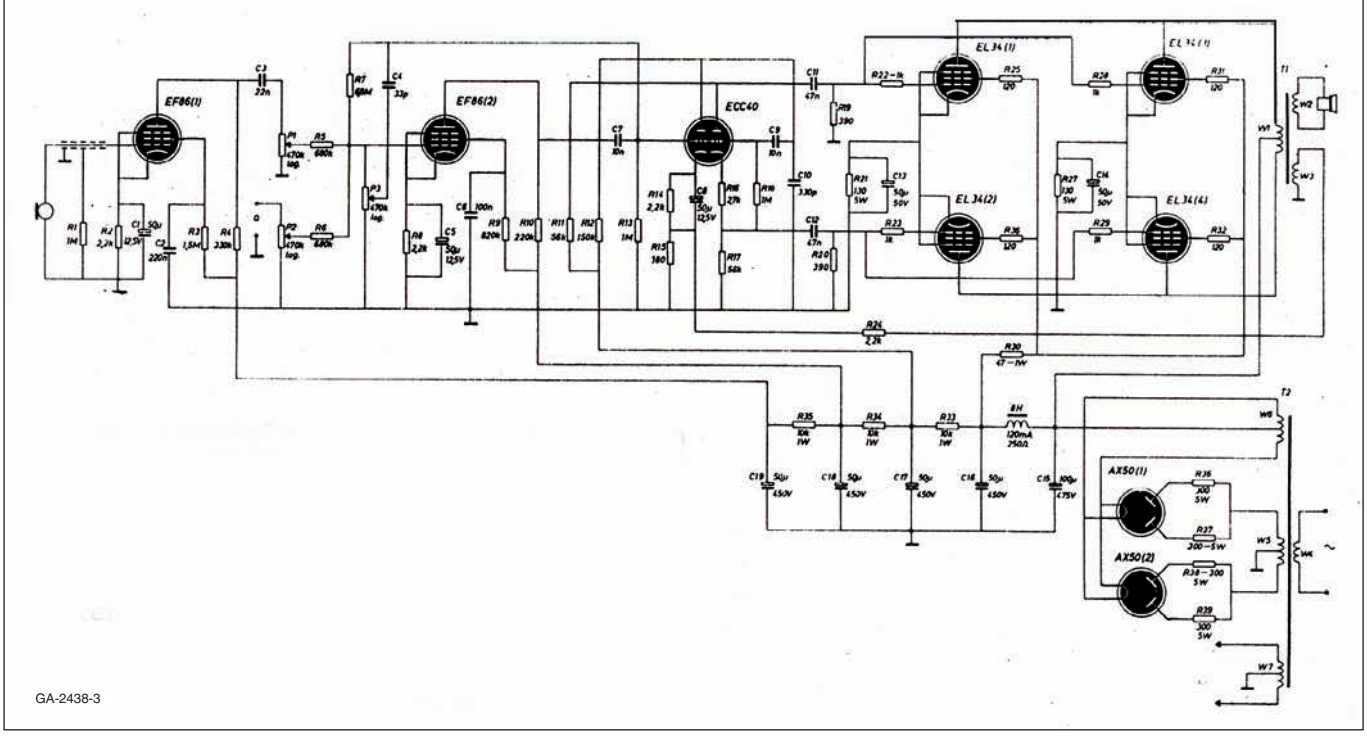
1. From the company *Amroh* (Muiden, The Netherlands).
2. From the book "*Jongens Radio*" (Boys' Radio), which was published in 17 editions between 1948 and 1971 (author M. van Geelkerken).
3.From the company *Philips* (Eindhoven, The Netherlands).

TABLE 2

70W versterker (vervolg)

C1-5-8	50µF 12.5V elco koker
C2	0.22µF papier
C3	0.022µF papier
C4	33pF peram.
C6	0.1µF papier
C7-9	0.01µF papier
C10	330pF keram.
C11-12	0.047µF papier
C13-14	50µF 50V elco koker
C15	100µF 475V elco (zie tekst)
C16-17-18-19	50µF 450V elco
	(alle weerstanden 10% 0.5W, tenzij anders aangegeven)
R1-13-18	1MΩ
R2-8-14-24	2.2kΩ
R3	1.5MΩ
R4	330kΩ
R5-6	680kΩ
R7	6.8MΩ
R9	820kΩ
R10	220kΩ
R11-17	56kΩ
R12	150kΩ
R15	180kΩ
R16	2.7kΩ
R19-20	390kΩ
R21-27	130kΩ 5W
R22-23-28-29	1kΩ
R25-26	120Ω
R30	47Ω 1W
R31-32	120Ω
R33-34-35	10kΩ 1W
R36-37-38-39	300Ω 5W
P1-2-3	470kΩ potm. Log.
T-1-2	zie tekst
L1	8H 120mA 250Ω

FIGURE 3: 70W AMPLIFIER.



solid state

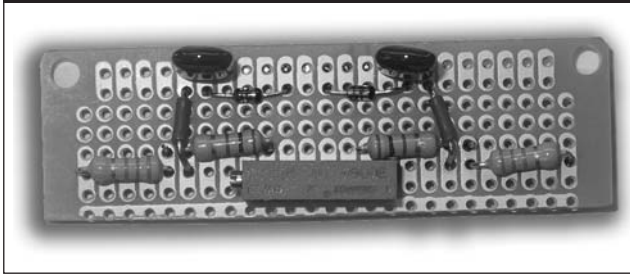
Back

A Balance Control for the Hafler DH-200

By Cornelius Morton

Make a classic even better . . . with this easy, inexpensive mod for this notable amp.

PHOTO 1: BALANCE BOARD, COMPONENT SIDE.



When David Hafler brought out the DH-200, it was a very good power amplifier at an exceptional price. It was simple, with good parts; the kit was easy to assemble; and the sound was good: 100W/channel loud and about as bullet-proof as an Abrams tank. There were a few warts though. Hafler cured a couple with the DH-220, adding a balance control and an inverting input to ease bridging, increasing the input impedance to 47kΩ.

To ensure acceptance, he added a few windings to the power transformer secondary winding so that the output voltage went from ±60V to ±65V, increasing the rated power to 115W/channel. There are numerous available upgrades for both the DH-200 and the DH-220, with many costing more than the original unit. The DH-220 balance control was a major improvement over the 200.

A balance control generally requires new circuit cards at a cost of $100–$200 per channel. This article describes a balance control that allows each channel offset to be adjusted to ±1mV for a cost of about $12 and a couple hours of time.

OVERVIEW

Figure 1 shows the basic balance circuit. The two 33kΩ resistors and two 1N4148 diodes connect between the negative and positive 58V sources for the low-level circuits. The center junction of the diodes is grounded. A current of about 1.7mA flows through the 33kΩ resistors. A series combination of a 1kΩ resistor, a 1kΩ potentiometer, and a 1kΩ resistor is connected across the two diodes. As the diodes maintain a voltage drop of about 635mV per diode, the voltage across the potentiometer is –212mV at one end and +212mV at the other.

The center tap is connected to what was the ground end of the input resistor R3, 22kΩ, and provides a variable bias voltage to the input transistors Q1 and Q5. As the potentiometer specified is a 15-turn unit, the bias adjustment is very smooth and easy to set.

This is basically the circuit that was used in the DH-220 adapted to the DH-200 to minimize effects on input impedance and overall performance. While the drawings and photos supplied with this article should allow you to build and install this mod without reference to DH-200 drawings or schematics, a copy of the manual is very handy. Complete manuals including schematics and part lists are available as free downloads from Hafler-Tech Support at the following URL: http://www.hafler.com/techsupport/index.asp?1D=3.

FIGURE 1: BALANCE CIRCUIT SCHEMATIC.

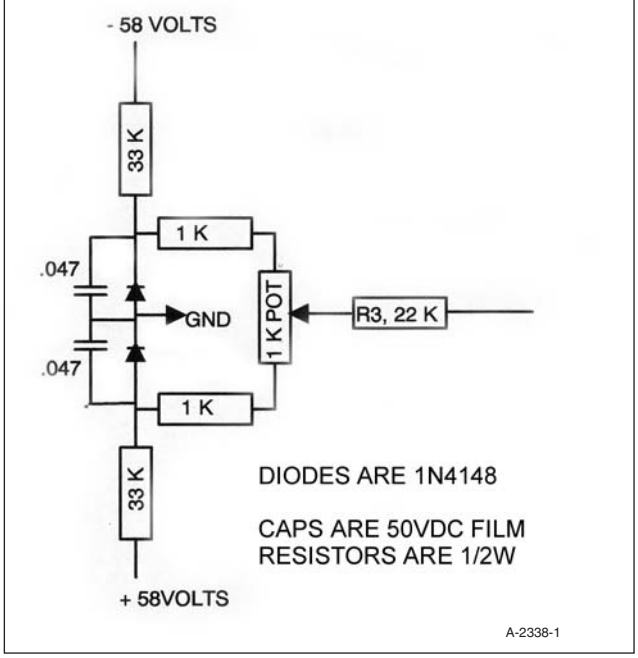
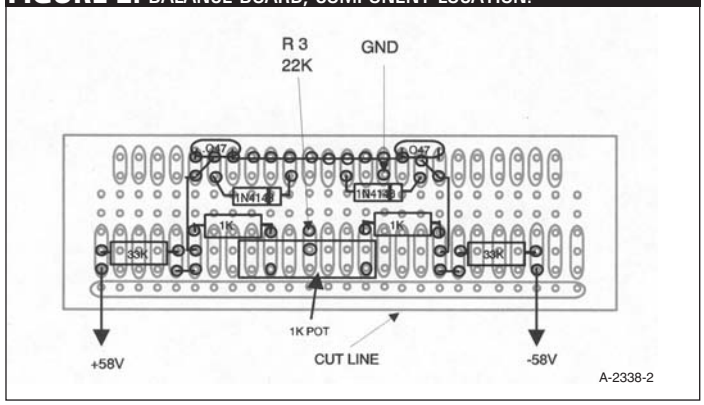


FIGURE 2: BALANCE BOARD, COMPONENT LOCATION.



MOD CONSTRUCTION

Begin construction by cutting the Radio Shack circuit board in half so that you have two pieces similar to the board shown in *Fig. 2*. Smooth the cut edge and position the board as in *Fig. 2* with the foil side down. Use *Fig. 2* as a guide to populate the board and install the interconnecting jumpers as shown. Use the facing leads of the two 0.047µF caps to construct the long ground bus between them. Ensure that the bus is soldered to each pad it crosses.

I used solid 22-gauge wire for the remaining jumpers and the wiring to the main board. When installed, the balance board is held in place by this wiring. *Photo 1* shows the component side of the board.

Turn the board over and insert a 22kΩ, ½W resistor (R3), as shown in *Fig. 2*. Leave ⅛″ to 3⁄16″ clearance between the foil and the resistor body to facilitate soldering. Cut and strip about 2″ of 22-gauge wire and insert it into the location labeled GND in *Fig. 2*. Cut two 3″ long pieces of 22-gauge wire and strip ¼″ from one end of each. Insert the stripped ends of these wires into the holes marked +58V and –58V in *Fig. 2*.

PHOTO 2: BALANCE BOARD, FOIL SIDE.

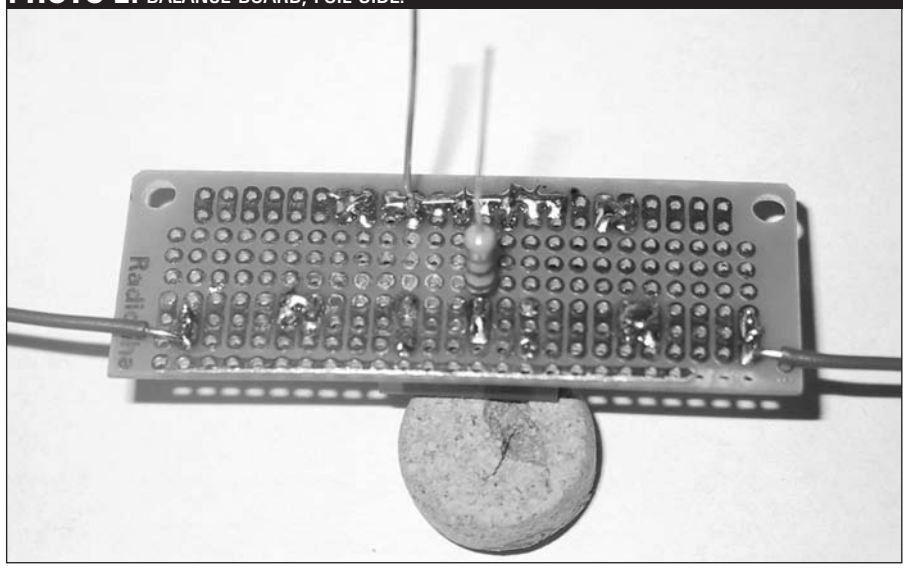
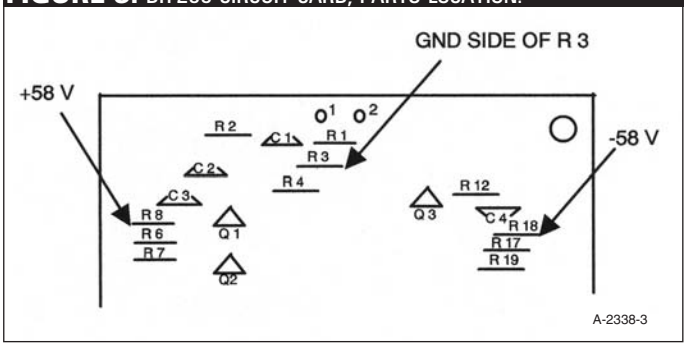


Photo 2 shows the foil side of a completed board ready to install. Inspect the board for sturdy solder joints, shorts, and so on, and ensure that all excess lead ends are clipped. Connect an ohmmeter between either end of the potentiometer and the center tap and adjust to read 500Ω, centering the potentiometer. Repeat these steps for board #2.

Disconnect all input and output connections from the DH-200 and ensure it is unplugged from the wall outlet. Remove the eight screws holding the top cover and remove the cover. Remove the four screws holding the heatsink on the left. It makes no difference whether you are facing the front or back of the unit, just remove the screws from the heatsink to the bottom chassis.

PHOTO 3: BALANCE BOARD INSTALLED.

FIGURE 3: DH-200 CIRCUIT CARD, PARTS LOCATION.



Fold the heatsink down away from the chassis. Remove the three screws holding the circuit board to the heatsink. There is a small insulating washer between the board and the heatsink on each screw. Do not lose the washers. You can fold the circuit board back toward the chassis exposing the foil side of the board. Using *Fig. 3*, locate R3, a 22kΩ ¼W resistor.

Using a small iron and solder wick, remove R3 and clean up the mounting holes for R3. The leads of R3 are a bit short to use that component, which is the reason to replace it. Position the balance board as shown in *Fig. 2*. Hold the DH-200 circuit board upright and insert the bare ground wire into the ground mounting hole for R3 and the free lead from the new R3 on the balance board into the remaining mounting hole for R3 on the DH-200 circuit board. Position the balance board parallel to the circuit board and about ⅛″ above the top of C1 (*Photo 3*).

Solder the ground lead and the lead from the new R3 in place. Locate the ends of R8 and R18 marked +58V and –58V in *Fig. 3*. There will be enough free space between the circuit board and the bent leads to slip in a wire. Cut the + and – leads from the balance board to fit with a bend and strip the ends back about 3⁄16″. Insert the ends in place and solder securely. Note that the component leads on the circuit board have been around a while, so a little cleanup may be required to secure a good joint. Check the wiring and re-attach the board to the heatsink, and do not forget the washers.

TESTING

The DH-200 can safely be run with the heatsink folded down, so you can see whether everything is set up correctly.

You will need a voltmeter that can read down to at least 10mV (most digital meters will read to 1mV on the 2V range) and a small insulated screwdriver. Connect the voltmeter to the modified channel speaker terminals and set to a range that will read 100V DC (just in case). Do not connect anything to the input. Plug the power cord in and turn it on!

If all is well adjust the voltmeter to obtain a reading, probably below 1V. Adjust the balance potentiometer to obtain a reading of less than 10mV. Note that adjusting the pot applies a step voltage to the input and causes a rather large initial change on the output, so allow the reading to settle and keep the adjustments slow. When all is well turn the DH-200 off, disconnect the line cord from the wall, and disconnect the meter. Re-attach the heatsink to the chassis and repeat the preceding for the remaining channel.

If you are unsure of the bias setting for each channel, this is a good time to check. There are two dual fuse blocks on the chassis floor set between the circuit boards and the power transformer. The fuses for the positive 60V supplies to each channel (F2L and F2R) are the ones closest to the power transformer. The output side of the fuses go to terminal 3 or 4 of each circuit card. This adjustment is done with both the input connectors and the speaker terminals open.

Make sure the line cord is unplugged; remove F2 from the fuse block of the channel being checked and connect an ammeter across the empty fuse block. Set the ammeter to read 1A and turn the DH-200 on. Let it warm up for a few minutes and adjust potentiometer P1, which is near the center of the board, for a reading of 275mA.

Turn off the power and allow the power supply to discharge until the ammeter reads zero. This may take a while as there are no bleeders on the DH-200 power supply. Replace the fuse and repeat for the other channel as required. If the bias is off by more than 10%, you should recheck the balance and touch up as needed. Replace the cover and your Hafler should be ready to go.

If you have not done the standard power-supply mod, it is simple and easy to perform and improves the high-frequency definition. Shunt each 10,000µF filter cap with a good 2.2µF polypropylene cap and a 47kΩ ½W resistor. The original filter caps have a bit of inductance. The 2.2µF cap takes care of the inductance and the resistor is a bleeder for the power supply. Enjoy the music. *aX*

PARTS LIST

Component	Quantity	Description	Source
Potentiometer	2	1kΩ, 15 turn	Radio Shack 271-342
Capacitor	4	0.047µF, 50V DC, polyester film	Radio Shack 272-1068
Diode, switching signal	4	1N4148	Radio Shack 276-1122
Resistor	4	33kΩ, ½W	Radio Shack 271-1129
Resistor	2	22kΩ, ½W	Radio Shack 271-1128
Resistor	4	1kΩ, ½W	Radio Shack 271-1118
Grid syle	1 each		Radio Shack 276-150
PC board			
About 1′ of #22 gauge solid wire, insulated			

speakers

The Kleinhorn, Part 1

By Nelson Pass

Back

With the assistance of some muscle-on-loan, our intrepid designer tackles the ultimate in big sound from a pair of exponential rear-loaded horns.

Thirty-three years ago I decided to build a big horn system. Constructed in the dorm library over Christmas break, the result was dubbed The Claw, a straight exponential horn 9′ long with a 42Hz expansion curve and a 50ft^2 mouth. We mounted a JBL LE15A woofer in the throat and used a JBL 375 compression driver and horn for about 500Hz on up. No matter where we pointed it, the cops showed up every time.

I've always wanted to build a stereo pair capable of even lower frequencies, and these days I am fortunate enough to have a 12,000ft^3 listening room isolated from civilization. This summer I was also graced with the help of a twin pair of engineering students, Chris and Matt Williams, eager to build something cool. So we tackled a set of big exponential rear-loaded horns.

They are named Kleinhorns (*Photo 1*) because they vaguely resemble a drawing of a Klein Bottle, a 4th dimensional container whose inside is the same as the outside. Perversely, Klein also means *little* in German. Horns vaguely resembling this project have been around for many years, but I believe the name at least is original.

THE EXPONENTIAL HORN

A horn is an acoustic transformer, changing high pressure and low volume at the throat to low pressure and high volume at the mouth. It does so by slowly expanding the cross section of the tube down which the sound wave travels, and it produces an acoustic load for the driver as if it had a very large diaphragm, dramatically raising its efficiency.

A horn loudspeaker is characterized by several numbers: the area of the small end known as the throat, the wide end known as the mouth, the distance from the throat traveling down the length of the horn toward the mouth, and the expansion curve of the cross section of the horn as you travel that distance. In an exponential horn, this expansion is given by the initial throat area multiplied by the natural logarithm e raised by a power factor related to the distance down the horn and the lowest frequency we want to work at.

We are very interested in the lowest frequency that the horn will work its transformation. The lower cutoff frequency is where the horn no longer "amplifies" effectively, and for an exponential horn generally the horn will be usable down to ½ octave above this point.

Figure 1 shows a picture of these elements and the formula giving the desired cross-sectional area, which is proportional to e (2.718) raised to the power of 4 * pi times X inches divided by lambda, the wavelength of the cutoff frequency. The Kleinhorn is designed around a

PHOTO 1: EVERYTHING ABOUT THE KLEINHORNS IS BIG, INCLUDING THE SOUND.

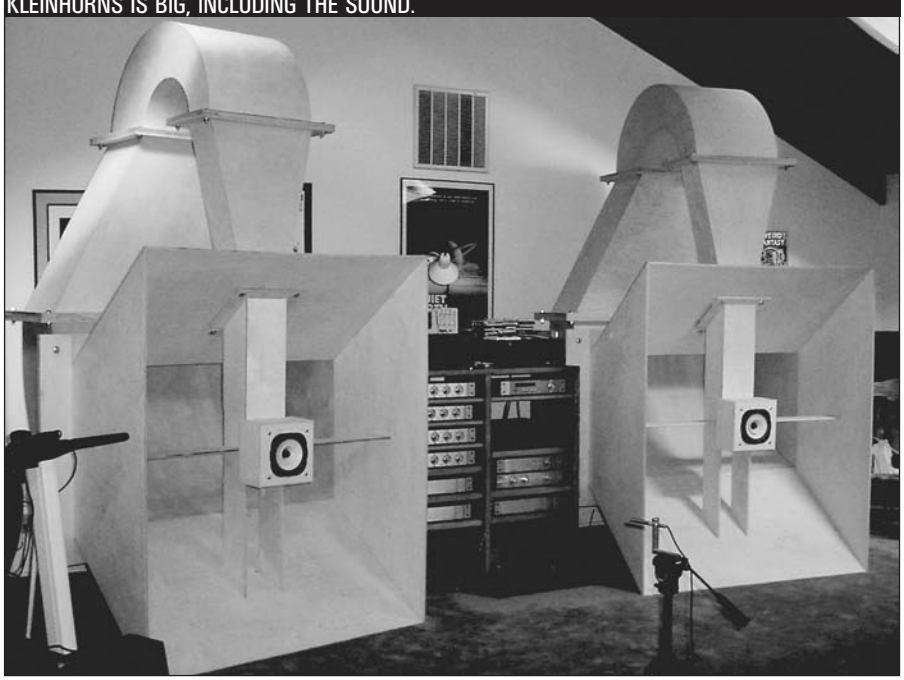
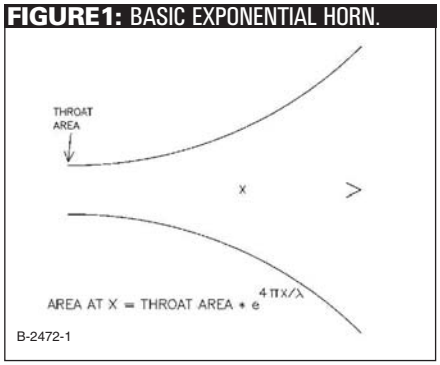


FIGURE1: BASIC EXPONENTIAL HORN.

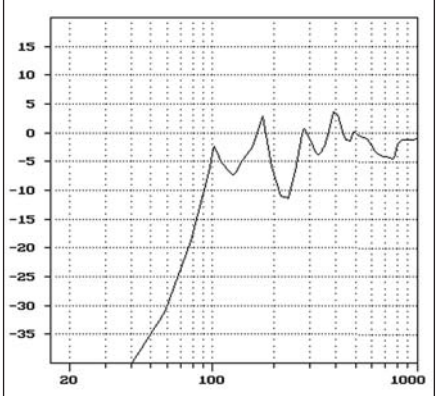


cutoff frequency of 30Hz, which gives a wavelength of 440″ (13,200″/second divided by 30/second = 440″).

The value of the exponent is 4 * pi * X /440, which equals .029 times the length X in inches. As a practical matter, we can look at this as how many inches it takes for the cross-sectional area to double, which for 30Hz is 24″. This figure is inversely proportional to the frequency, so that for 60Hz, the area doubles every 12″.

In order for the acoustic wave to be transformed, the length of the horn should be at least ¼ the wavelength of the cutoff frequency, which in this case is about 9′. Also the mouth of the horn needs to be sufficiently large, its area wanting to be about the same as a circle with a diameter about ⅓ wavelength, in this case about 12′, with an area about 117ft^2. This formula is for horns operating in free space, which will not be the case here.

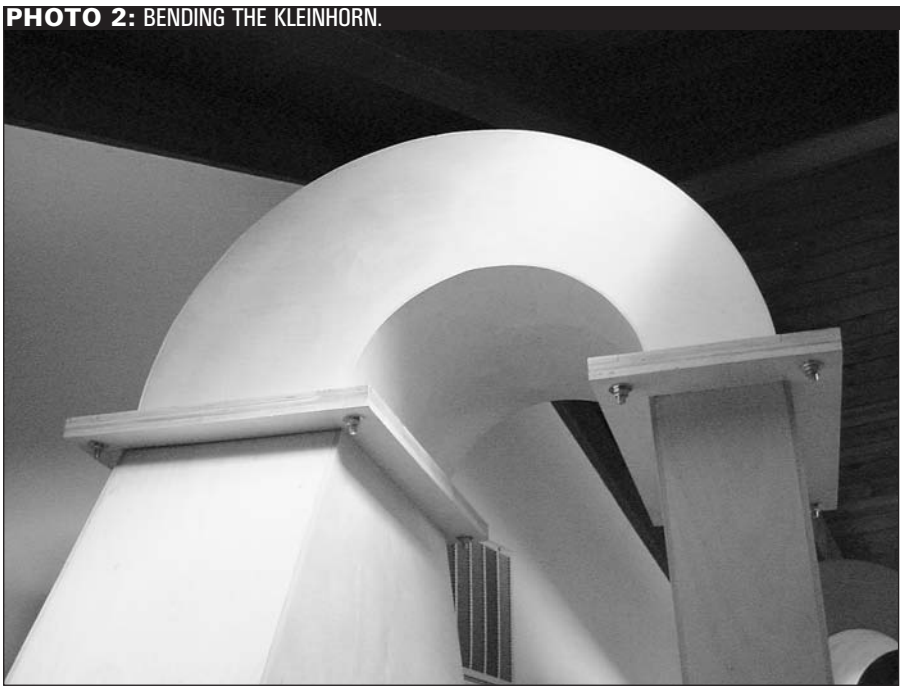
FIGURE 2: SMALL MOUTH RESONANCE.



CONSTRUCTION

For practical reasons, the Kleinhorns

PHOTO 2: BENDING THE KLEINHORN.



will each have about 30ft^2, and between them only one-half that area. Assuming essentially monaural signal at 30Hz, we can get away with this because the horns will be sitting on a floor and near a wall, producing something resembling ½ to ¼ space, which will give us some acoustic boost at the lowest frequencies. In actual measurement we see that in a large room this horn will give us usable transformation to about 40Hz, which is what we expect.

As we make the horn mouths smaller than the wavelength of the lowest frequency, we start seeing curves that resemble *Fig. 2*, an actual horn curve.

Here we see the free air response of a horn with a mouth with a diameter much smaller than the wavelength of the cutoff frequency. The output is still transformed, but becomes resonant and irregular and loses amplitude at a much higher frequency than the taper. Because most horns, including the Kleinhorns, are compromised to some degree in this fashion, it is popular to place them against a wall, and in more extreme cases, in a corner to boost the acoustic loading at the lowest frequencies.

As a practical matter, it is difficult to find the space for a straight horn whose length must be at least 10′, and so we look to fold the horn into a smaller space. In addition because this is a rear-loaded horn intended for a single driver system, we want the mouth to come out more or less pointed at the listener. So we bend the horn to achieve both these purposes.

In an ordinary duct, when we turn a corner 90°, we will experience roughly 8dB attenuation when the dimensions of the duct are comparable to ½ the wavelength. For this reason we want to avoid sharp turns, and you will see in the Kleinhorns that we try to gently ease the wave around curves when a bend is necessary (*Photo 2*).

Also as a practical matter, it is not easy to construct a nice, smooth exponential horn with a circular cross-section. Amateurs at woodworking at best, we chose to use a rectangular cross-section and break the horn into six segments, each having two parallel walls and geometric conical expansion. This way we have no complicated wood cutting, and you can fit the horn through an ordinary door.

Fortunately, horns don't need to be exact to work well. A horn that is fairly close to an exponential expansion over its length, but piece-wise has sections with a conical expansion, works nearly as well as an ideal horn. Because we have some parallel surfaces over the length of the horn, we will encounter some minor standing waves that would introduce some additional irregularities in the response.

Figures 3 and *4* show the Kleinhorns from front and side. They were fashioned from various thicknesses of Baltic (Russian) birch plywood, void free, more dense, and with thinner and more numerous grain-aligned laminations than regular plywood. Very nice stuff. The dimensions are in inches, and each curve is circular, with the radius shown. At each junction you will see the dimensions of the cross-section. From the drawings you will no-

FIGURE 3: KLEINHORNS FROM FRONT.

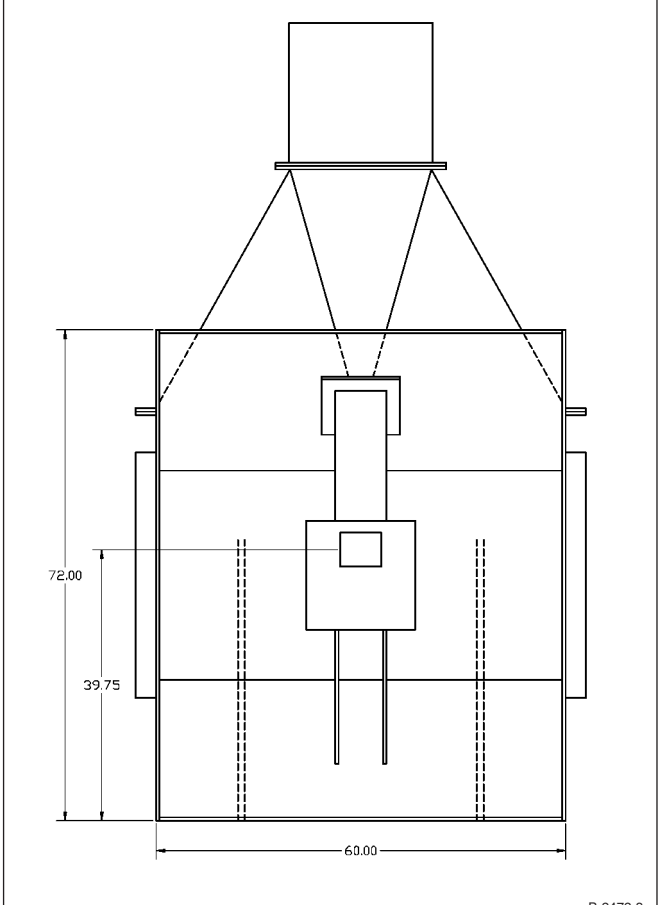
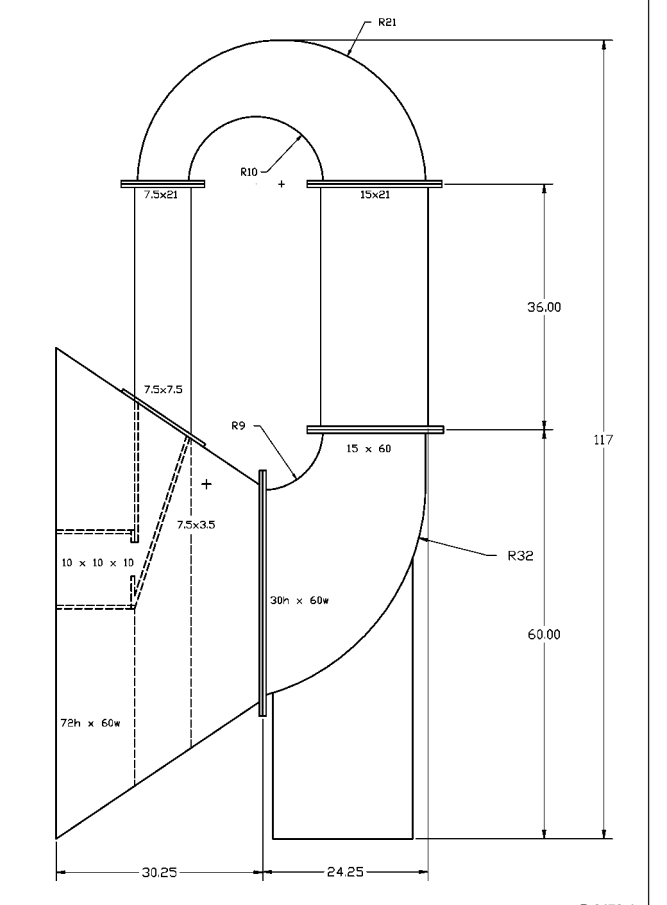


FIGURE 4: KLEINHORNS FROM SIDE.



tice the flanges at the end of each section that mate it to the adjacent piece with ½″ bolts.

These drawings are intended as guidelines, not blueprints. Use the materials you prefer, and feel free to improvise.

When we consider the effects of bending, parallel surfaces, and mouths that are slightly small, there will be a number of irregularities to smooth out. Toward that end, we will look to some acoustic filtering and damping to smooth out the response and filter out the higher frequencies going down the horn. Being a rear-loaded speaker, we want the response of the horn part of the loudspeaker to pick up where the low-frequency response of the driver starts to fall off, and we want to remove horn frequencies higher than that.

The box in which we mount the driver acoustically behaves as a capacitor. Properly stuffed with absorbent material, it attenuates the higher frequencies off the back of the driver before the throat of the horn. The Kleinhorn was designed to have a modular approach to this box, allowing for different size rear chambers to be bolted onto the throat of the horn and easy access to the stuffing, so that the filter value can be varied.

The second treatment used to control resonance is to also fill the interior of the horn itself with absorbent material. This is done starting at the throat and moving toward the mouth. The amount

FIGURE 5: RESPONSE OF LOWTHER DX55 IN .6FT^3 SEALED BOX WITH VOLTAGE SOURCE.

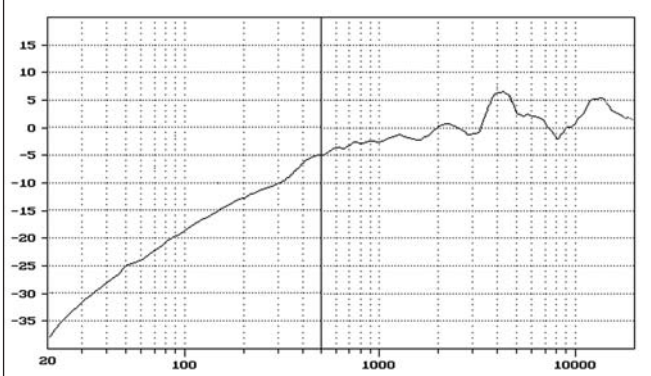


FIGURE 6: UNEQUALIZED RESPONSE OF DX55 IN HORN WITH VOLTAGE SOURCE.

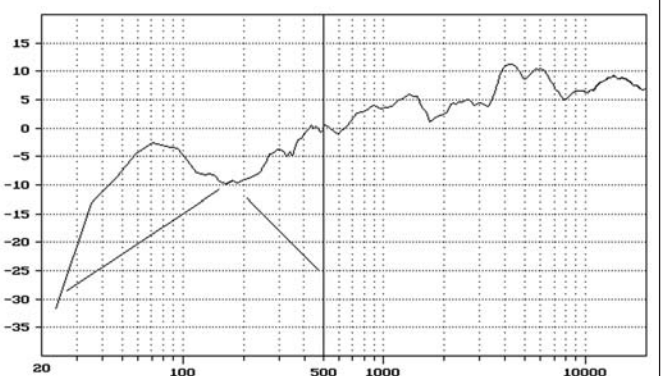


FIGURE 7: EQUALIZED RESPONSE OF DX55 IN HORN WITH VOLTAGE SOURCE.

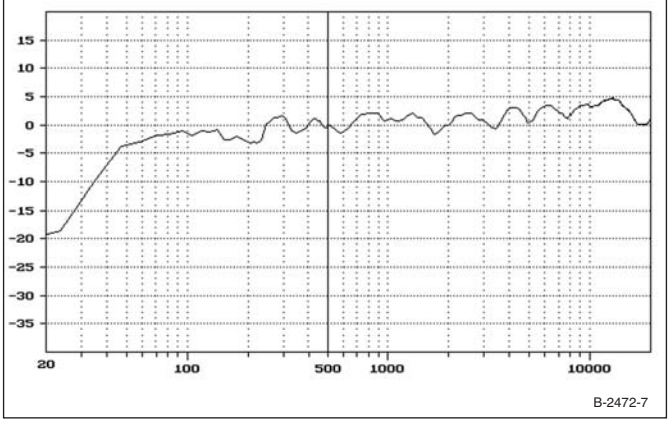
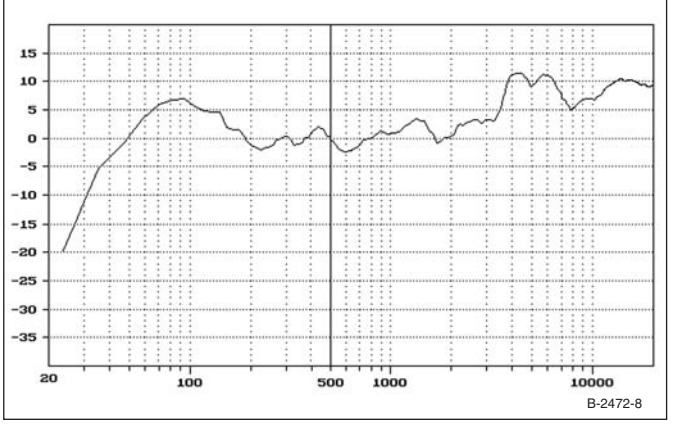


FIGURE 8: UNEQUALIZED RESPONSE OF DX55 IN HORN WITH CURRENT SOURCE.



of stuffing and its density is determined experimentally. Currently I have the front vertical part of the horn stuffed into the top arch. Sheets of absorbent material (in this case, carpeting) covered parallel surfaces in the rear vertical section. The mouth surface was left undamped.

Listening tests were the criteria used to decide on the damping. I started with the speaker box and then worked from the throat to the mouth, applying stuffing and absorbent sheets and listening to the results. In the end, I tried absorbent material on the parallel walls of the mouth area, but removed it, preferring the sound of the uncovered wood. At the time of this writing, I am still playing with the damping.

PERFORMANCE

We go to all this trouble to obtain the best performance from a high-efficiency full-range driver, such as the Lowther DX55. This is an extremely well-made and potentially very excellent-sounding driver, but its small size, light moving mass, and very efficient motor do not make for much in the way of bass response. *Figure 5* shows the response curve of this speaker in a small closed box, which is not much different in a large closed box. *Figure 5* also indicates why this driver is most often found in rear-loaded bass enclosures and not sealed boxes.

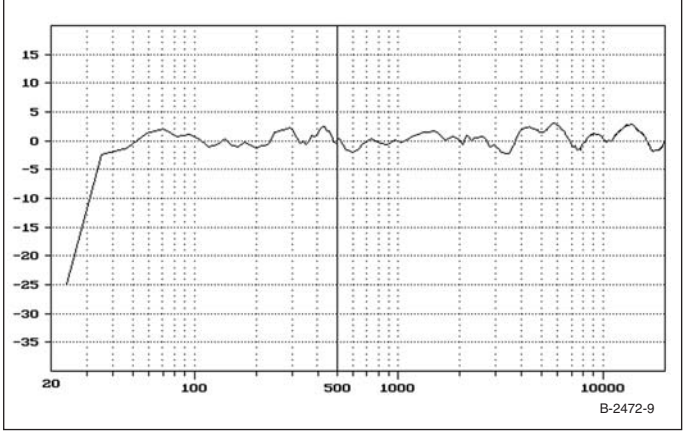
In *Fig. 5* we see the near field of the speaker measured with a 5kHz bandwidth up to 500Hz, and this curve is joined by the normalized response taken at 1 meter with a 25kHz bandwidth above 500Hz. The curve has been smoothed to ⅓ octave.

In *Fig. 6* we see the DX55 in a .6ft^3 box mounted to the throat of the horn. This curve is the combination of a curve with a 5kHz bandwidth below 500Hz, and a 25kHz bandwidth above 500Hz, both taken at a distance of 6′. Again the curve is smoothed at ⅓ octave. We cannot take a near field figure for the horn, as the operation of the horn sets conditions where there is near zero pressure but high volume at the mouth, and the pressure does not develop until you get about ¼ wavelength away at the lowest frequency, so it was measured 6′ away, which allows measurement to about 40Hz.

It is difficult to separate the contributions of the front and rear, but using the data from *Figs. 5* and *6* I estimated this and drew two lines showing the approximate relative outputs of each. We infer from this curve that the horn increases the bass energy by a factor as much as 20dB over its operating range, which is the difference between 1W and 100W.

In this case we detect a bit of a lapse between the two between 100 and 200Hz.

FIGURE 9: EQUALIZED RESPONSE OF DX55 IN HORN WITH CURRENT SOURCE.



This can be improved with a smaller box or with part of the equalization that I intended to apply. Trying both, I found that I preferred the original box and equalization, finding that it was better to limit the bandwidth of the rear horn to a lower frequency to minimize the effects of the substantial delay through the length of the horn.

Figure 7 shows the same conditions as *Fig. 6*, but with a parametric equalizer used to correct the response curve. It was possible to equalize the curve a little more dramatically to make it flatter, but I found that it didn't sound as good—it was described by one listener as "sucking the life out of the music," and so I actually ended up preferring the sound of curves which were not as flat as this.

I also tried driving the DX55 in the horn with a current source

FIGURE 11: RESPONSE OF CURRENT SOURCE LOAD EQUALIZATION.

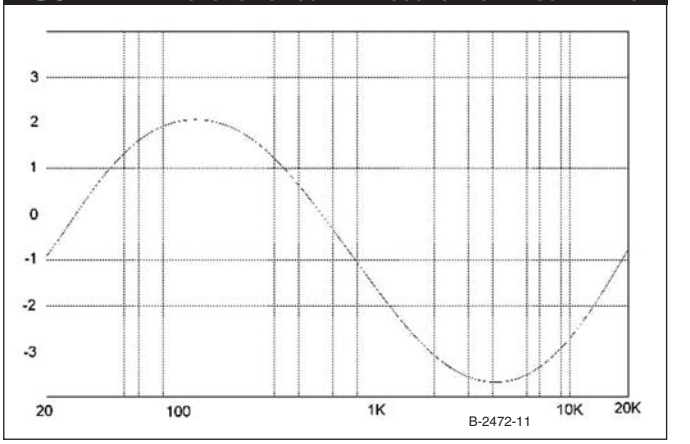


FIGURE 10: LOADING CIRCUIT FOR CURRENT SOURCE–R0, L0, R1, C1.

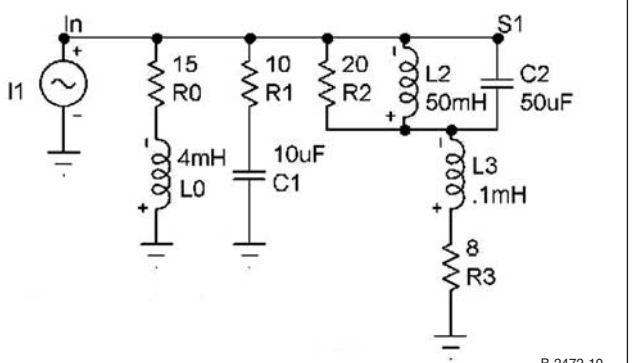
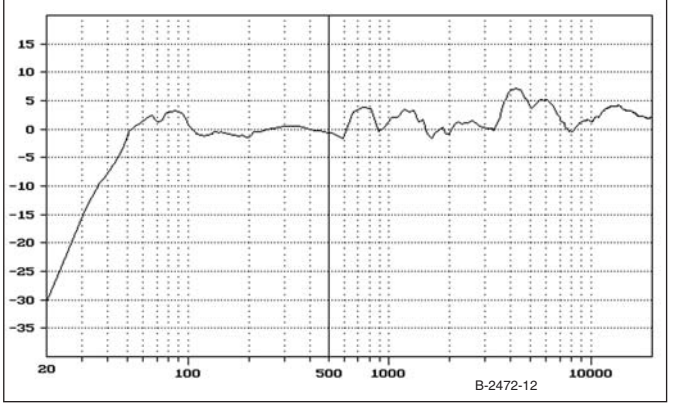
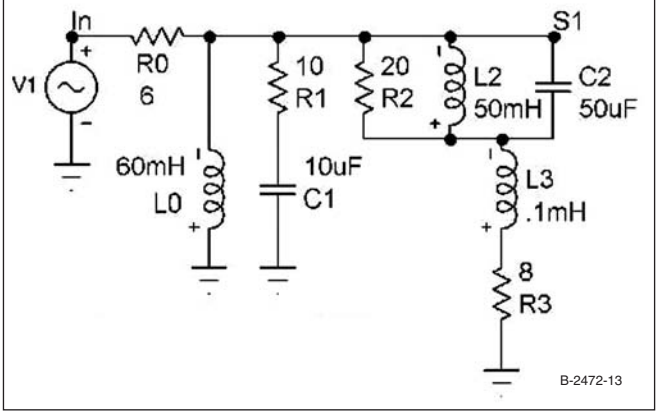


FIGURE 12: RESPONSE OF DX55 IN HORN WITH PASSIVE LOADING.



having an 80Ω output impedance and produced the curve of *Fig. 8*. It is considerably flatter than the curve of *Fig. 6*, as the current source delivers more power at the bottom end where the impedance of the driver shows its fundamental resonance at about 90Hz. Unequalized, this system sounds fairly good by itself, particularly if you are off axis about 20° or so.

FIGURE 13: EQUALIZATION NETWORK FOR VOLTAGE SOURCE.



Again I applied parametric equalization—although less of it—to the system to obtain the flatter curve seen in *Fig. 9*. The results were subjectively better than those of *Figs. 7* and *8*—I speculate that both the current source and lesser equalization contributed to this effect. Encouraged by the current source performance, I worked up a passive "Zobel" type parallel loading network to smooth out the curve of *Fig. 8*. The resulting circuit is shown in *Fig. 10*.

In *Fig. 10*, the current source has an 80Ω source impedance (that of the First Watt F1 amplifier), and the speaker impedance is represented by the components R2, L2, C2, R3, and L3, which models the DX55 in the Kleinhorn. R2 would ordinarily be the 90Ω value for a DX55 in free air, but the horn acoustic resistance damps down this resonant circuit, so that a 20Ω figure is closer to reality. The loading network is seen as R0, L0, R1, and C1, and the resulting voltage across the driver is seen in simulation in *Fig. 11*, and the measured output is seen in *Fig. 12*.

The circuit of *Fig. 11* and curve of *Fig. 12* sounded significantly better than the best previous result, that of *Fig. 9*. It had a livelier and more musical character, and became the reference system. As before, this curve was taken at 6′ on axis, with ⅓ octave smoothing, and a 5kHz bandwidth

PHOTO 3: ENGINEERING STUDENTS CHRIS AND MATT.

below 500Hz, and 25kHz bandwidth above 500Hz.

This raises the question of whether passive equalization with a voltage source would be better yet, but I did not explore this at any length. I did, however, work up a simulation of what such a circuit might want to look like, and came up with *Fig. 13*, where the voltage response across the driver is very similar to that of *Fig. 11*. If you happen to build a horn like this and use a DX55, I suggest it is a good starting point.

HORN DELAY

There are time delay issues that appear when you back-load a driver, particularly when you use a long horn. If you are not simultaneously using the front wave, some of these can be addressed with digital delays and/or placement to minimize the effect. In the Kleinhorn there is substantial delay on the rear wave due to the length. This can be reduced by shortening the length and using a larger throat, but this would be appropriate for larger drivers such as the 8″ Lowther DX4 and its cousins and the Fostex FE206E and its relatives. In any case, this would remove only a couple of feet or so.

The larger question is, how audible is this delay? My experience with this horn leads me to believe that it's not nearly as audible as you would expect. You can detect it on certain passages, but it's not particularly problematic and I don't find that it interferes with the overall enjoyment of the sound. I find that I would rather have a bottom end with a wavelength of delay than no bottom end, but part of the key to making this work is to carefully choose the driver box size, stuffing, and surface damping to filter the higher frequencies out of the horn response. Like most of the rest of the decisions on this project, this is best done through experimentation and judged by a combination of listening and measuring.

CONCLUSION

So how do they sound? Big. Very big. The articulation and detail of the DX55s have made them my current favorite in this category, but while you can coax some low end out of them in a conventional box, at reasonable levels the cone excursion introduces enough intermodulation distortion that it hides these qualities. A big rear-loaded horn is intended to alleviate this problem.

I have been extremely pleased with the results, but I must smile when I see that all this effort barely gets us below 50Hz. Still, they play loud cleanly in a manner that Lowther owners don't usually encounter, and the cones are barely moving.

Chris and Matt (*Photo 3*) learned a lot about woodworking and speaker building this summer, and we all had a lot of fun.

In addition to Chris and Matt I want to thank Desmond Harrington and Kent English—all a pleasure to work with.

But wait! There's more! In part 2, we will extend the low-frequency response below 20Hz by mating the Kleinhorn with El Pipe-O, plus some other goodies. (Copyright 2004, Nelson Pass.) *aX*

solid state

Noise Meter Amp

By Charles Hansen

Back

For low-noise measurements, this unit adds three decades of measurement to the bottom end of any high-performance AC voltmeter . . . and also features an A-weighting filter.

PHOTO 1: FRONT VIEW OF NOISE METER AMPLIFIER.

For making noise measurements in low-noise audio systems, it is desirable to be able to measure microvolts of noise. This noise meter amplifier (NMA) is designed specifically for the HP 3400A true-RMS voltmeter, but it works just as well (but with more limited bandwidth) with my HP 339A distortion test set and my Fluke 8050A true-RMS digital multimeter (DMM). All three meters have a low-end resolution limit of 1mV full-scale (FS).

Another useful feature I added is an "A-weighting" filter, which is a special filter used in sound level and audio noise meters. It is a selective bandpass filter designed to complement the subjective way we hear, because the ear's sensitivity versus frequency is not flat.

The ANSI A-weighting curve uses a wide bandpass filter centered at 2.5kHz, with 20dB attenuation at 100Hz, and 10dB attenuation at 20kHz. It provides a steep rolloff for the low end and a more gradual rolloff at higher frequencies. It is the inverse of the 30-phon, or 30dB-sound pressure level (SPL), equal-loudness Fletcher-Munson curve. By contrast, the ANSI C-weighting curve is flatter, with –3dB corners of 31.5Hz and 8kHz, respectively. The parts list cost in *Table 1* is less than $70.

HOW IT WORKS

The schematic diagram for the NMA is shown in *Fig. 1*. The main power source is two 9V batteries, which must be used in the highest gain ranges if very low noise measurements are going to be successful. S1 is a DPDT center-off switch that can select either the battery power or a supply that rectifies the 12V AC input from a plug-in AC adapter at J5.

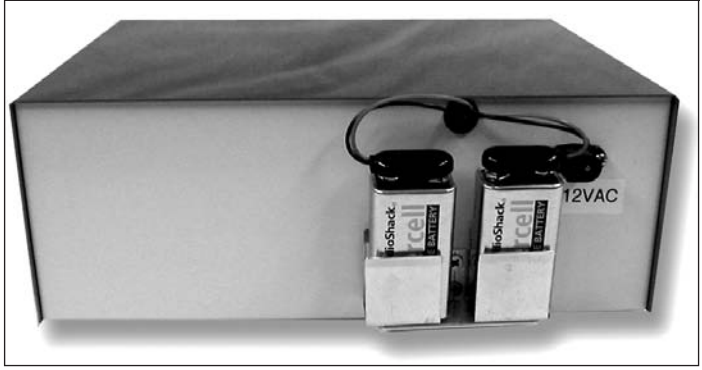
When you select battery power, C1 and C2, in conjunction with ferrite beads FB1 and FB2, low-pass filter any HF noise, while C3 and C4 are the main reservoir caps. C5–C10 provide local supply bypass for the three op amps. LED1, R28, and R29 function as the power-on indicator. In order to minimize power drain on the batteries, I chose an LED that has noticeable brightness at only 0.3mA.

When you select AC power, CR1 and CR2 half-wave rectify the 12V AC into roughly ±18V DC. C28 and C29 provide the primary ripple filtering, which is then applied to reservoir caps C3 and C4 through R25 and R26, forming a low-pass filter. R25 and R26 utilize the quiescent current of the three op amps to drop the supply to ±15V DC at the op amp rails. I did not use series IC regulators because they did not improve noise over this simple π filter.

The noise amplifier circuit consists of series gain stages U1 and U2. In order to make measurements convenient, the overall gain at the most sensitive range is Av = 1000 (60dB). This allows me to read 1μV FS on the 1mV FS range of the AC voltmeter. S2 allows me to select three gain ranges: 30dB (Av = 31.62), 40dB (Av = 100), and 60dB (Av = 1000). This makes the noise amplifier useful for other types of measurement and amplifying applications.

The noise amplifier circuitry will generate noise of its own through the input noise voltage and current of the op amps (e_n and i_n) and the thermal Johnson noise from all its resistors, so be careful when selecting components. I used two low-noise AD745JN op amps that Walt Jung graciously provided from his sample stock. The 8-pin DIP version is available from Rochester Electronics, a source of obsolescent semiconductors (www.rocelec.com). However, you can also use the AD745JR 16-lead SOIC version. Alternatively, you could use the OP-37 low-noise op amp, but there will be a bit

PHOTO 2: REAR VIEW OF NOISE METER AMPLIFIER.



higher noise, especially at the higher Rin setting, where the low 6.9fA/√Hz current noise of the AD745 outperforms the OP-37 at 0.4pA/√Hz.

The AC signal at gain input J2 is coupled by C11 and R3 to U1, making it possible to measure AC noise in the presence of a DC voltage. I used this NMA to measure the AC noise of DC audio supplies in my "DacT CT 102 Power Supply Product Review," *aX* Oct. 2003, p. 56. S5 selects between two Rin input impedances: 600Ω or 8k5. The LF –3dB point for each setting is 56Hz or 4Hz, respectively. The HF bandwidth is 225kHz. I added R1 to discharge C11 when the NMA is not in use.

Since the AD745 is only stable for gains ≥5, Walt suggested placing C12 and C13 (a value four times that of C14) across the inputs. These capacitors, in conjunction with C14, force the noise gain to 5, and provide stability over the entire operating range. I divided the gain equally between the two AD745 stages. During testing, I found that R10 needed to be a bit higher than R5 for exactly 60dB gain, probably due to the variations in the 1% tolerance of all the resistors.

The output of U2 goes through R23 to gain out jack J3 and also to S3. This switch selects between the gain stages of U1, U2, and a unity gain buffer U3B. The common pole of S3 delivers the selected signal to a passive "A-weighting" filter network[1].

I chose this particular filter circuit instead of others published earlier[2, 3] because SPICE simulations showed it to be more accurate to the A-weighting curve[4]. U3A adds the necessary gain to recover the losses through the passive network and provide unity gain at 1kHz. S4 selects either the flat response from S3 or the A-weighted response from U3A and drives J4 via R20.

CHASSIS ASSEMBLY

I built the NMA into a painted steel Radio Shack enclosure. The front view is shown in *Photo 1* and the rear view is *Photo 2*. The two 9V batteries are external to the unit to make replacement easier. *Photo 3* is the interior view. The shells of the two input jacks are grounded to the enclosure along with the circuit board ground plane. This requires that you remove the paint from the enclosure at the points of contact.

In addition, every signal wire that goes to a switch or jack uses shielded cable grounded only at one end to prevent ground loop noise from being picked up on the signal conductor. The output jacks are isolated from the chassis, again to prevent ground loops. The LED and power supply wires are twisted to minimize pickup or noise radiation.

CIRCUIT BOARD LAYOUT

I built the circuit on a ground plane PC board. This is absolutely necessary at these high gains to ensure the minimum input-to-output noise. *Photo 3* shows the view of the PC board, and *Fig. 2* shows the parts placement. I used low-profile gold-plated DIP sockets for the three op amps.

The PC board itself was a fairly simple layout, so I routed it with a 0.046″ router bit in my Dremel tool rather than use an etched PC board. This maximized the ground plane area. I drilled the board for standard T-49 vector pins so I could tweak any part values later.

Wherever the pins are not to be grounded, I used a Vector P138A pad cutter to cut an electrically isolated circular pad on the copper side around each 0.042″ (#58 drill) hole. (The pad cutter is available from Mouser as p/n 574-P138A.) Some parts, such as the power-supply components and de-coupling caps, are soldered directly in place since they would not require changing during testing.

FRONT PANEL

Figure 3 shows the panel lettering designations for the front panel. I made a full-size copy of this lettering on drafting appliqué film, which is an adhesive-backed transparent plastic. The light color base paint on the RS enclosure works well with the black photocopy lettering.

All circuit board mounting holes have clearance for a 4–40 screw with a flat washer. I used four tapped aluminum spacers to mount the board to the chassis.

MEASUREMENTS

I calculated the ideal noise level for the U1 and U2 gain stages using an Excel spreadsheet[5]. The total output noise voltage was 12.92μV/√Hz, or 91.4nV RMS over the audio bandwidth. This would be 91.4pV RMS noise referred to

PHOTO 3: INTERIOR VIEW AND CIRCUIT BOARD.

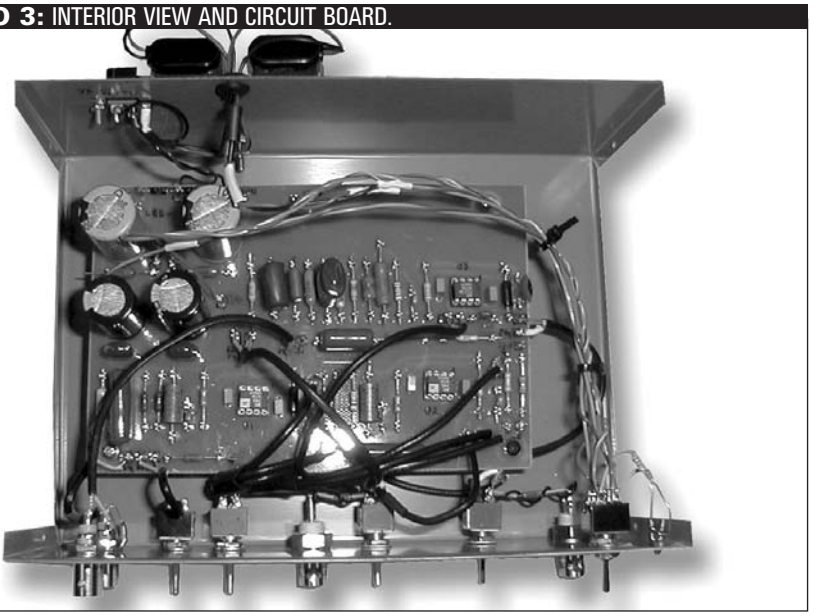


TABLE 1

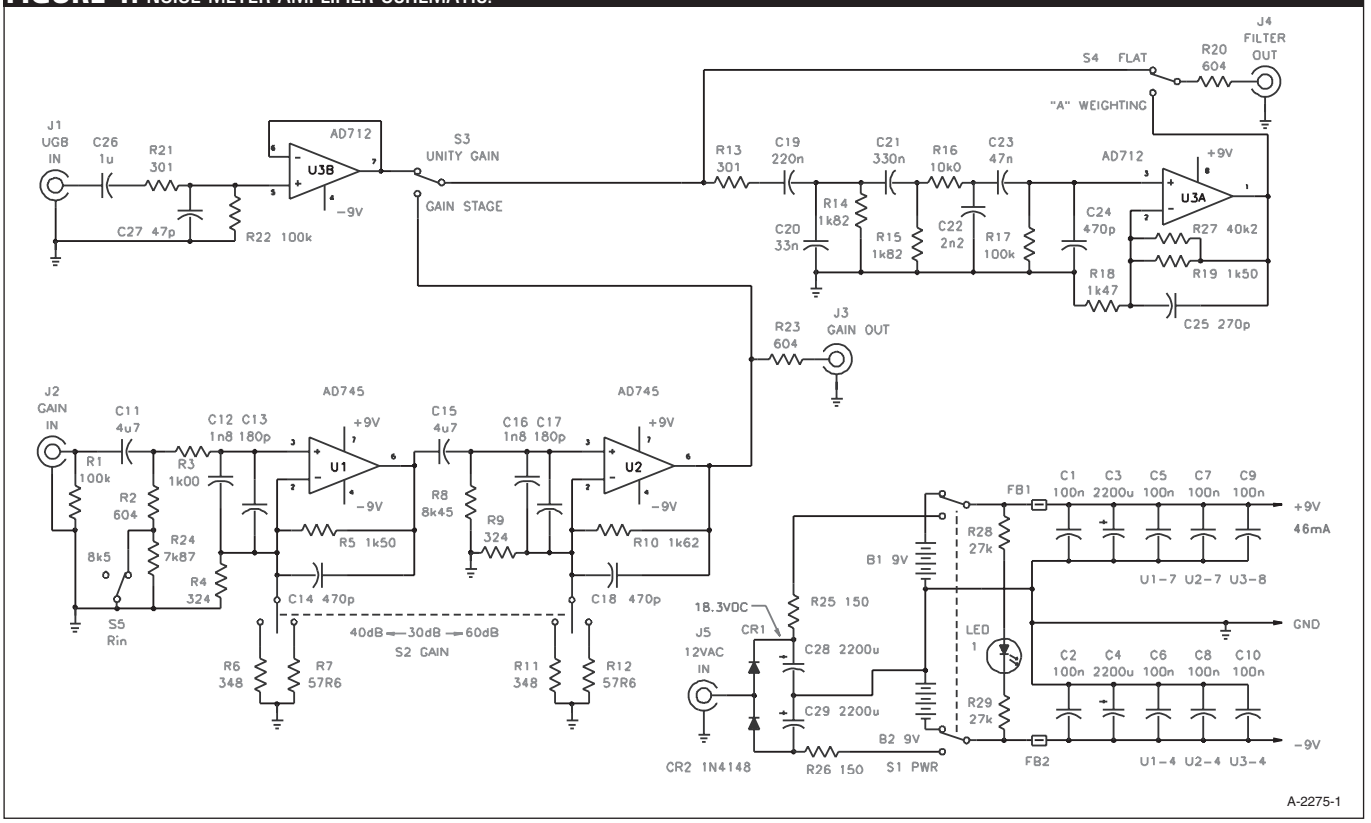
NMA Parts List

SYMBOL	VALUE	DESCRIPTION	VENDOR	PART NO.
B1, B2	9V	Battery	RS	
C1, C2	100n 100V 10%	Polyester	Mouser	140-PF2A104K
C3, C4	2200μF 35V	Aluminum	DK	P5170 ND
C5–C10	100nF 50V	Ceramic X7R	Mouser	21RX310
C11, C15	4μ7 250V	Polyester	Mouser	146-250V4.7K
C12, C16	1n8 100V 10%	Polyester	Mouser	140-PF2A182K
C13, C17	180pF 50V 5%	Ceramic NPO	Mouser	80-C315C181J1G
C14, C18, C24	470pF 50V 5%	Ceramic NPO	Mouser	80-C315C471J1G
C19	220nF 100V 10%	Polyester	Mouser	140-PF2A224K
C20	33nF 100V 10%	Polyester	Mouser	140-PF2A333K
C21	330n 100V 10%	Polyester	Mouser	140-PF2A684K
C22	2n2 100V 10%	Polyester	Mouser	140-PF2A222K
C23	47n 100V 10%	Polyester	Mouser	140-PF2A473F
C25	270pF 50V 5%	Ceramic NPO	Mouser	140-50N5-271
C26	1μF 250V 10%	Polyester	Mouser	146-250V1.0K
C27	47pF 50V 5%	Ceramic NPO	Mouser	140-50N5-470J
C28, C29	2200μF 35V	Aluminum	Mouser	140-XRL35V2200
CR1, CR2		1N4148	Mouser	78-1N4148
FB1, FB2		Ferrite Bead	DK	M2310-ND
J1–J2		BNC, Panel Mount	Mouser	523-31-221-RFX
J3–J4		BNC, Panel Mount, Ins.	Mouser	523-31-10-RFX
J5		Jack 5.5mm male	Mouser	163-5006
J6, J7		Battery Snap, 9V	Mouser	121-2224
LED1		Red, High Efficiency		
R1, R17, R22	100k 1%	Metal Film	Mouser	271-100K
R2, R20, R23	604 1%	Metal Film	Mouser	271-604
R3	1k00 1%	Metal Film	Mouser	271-1.00K
R4, R9	324 1%	Metal Film	Mouser	271-324
R5, R19	1k50 1%	Metal Film	Mouser	271-1.5K
R6, R11	348 1%	Metal Film	Mouser	271-348
R7, R12	57R6 1%	Metal Film	Mouser	271-57.6
R8	8k45 1%	Metal Film	Mouser	271-8.45K
R10	1k62 1%	Metal Film	Mouser	271-1.62K
R13, R21	301 1%	Metal Film	Mouser	271-301
R14, R15	1k82 1%	Metal Film	Mouser	271-1.82K
R16	10k0 1%	Metal Film	Mouser	271-10.0K
R18	1k47 1%	Metal Film	Mouser	271-1.47K
R24	7k87 1%	Metal Film	Mouser	271-7.87K
R25, R26	150 5% 1W	Metal Oxide	Mouser	281-150
R27	40k2 1%	Metal Film	Mouser	271-40.2K
R28, R29	27k 5%	Carbon Film	Mouser	291-27K
S1	DPDT	Mini Toggle	Mouser	10TC260
S2	DP ON–OFF–ON	Mini Toggle	Mouser	108-MS550H
S3, S4, S5	SPDT	Mini Toggle	Mouser	108-MS550A
U1, U2	AD745, OP-37	Op Amp	Newark	See text
U3		Op Amp	Newark	AD712JN
		Batt snaps, 9V	RS	270-326
		Blank 4×6 pcb	ALL	PCB-46
		Enclosure, 8×3×6	RS	270-274
	12V AC 0.4A	AC adapter	Mouser	41AC412

the input (RTI) at 60dB gain.

At 60dB gain, the NMA proved to have 3.4mV RMS residual output noise with the input shorted, and 4.7mV RMS with the input open (using the HP 3400A). The noise floor referred to the

FIGURE 1: NOISE METER AMPLIFIER SCHEMATIC.



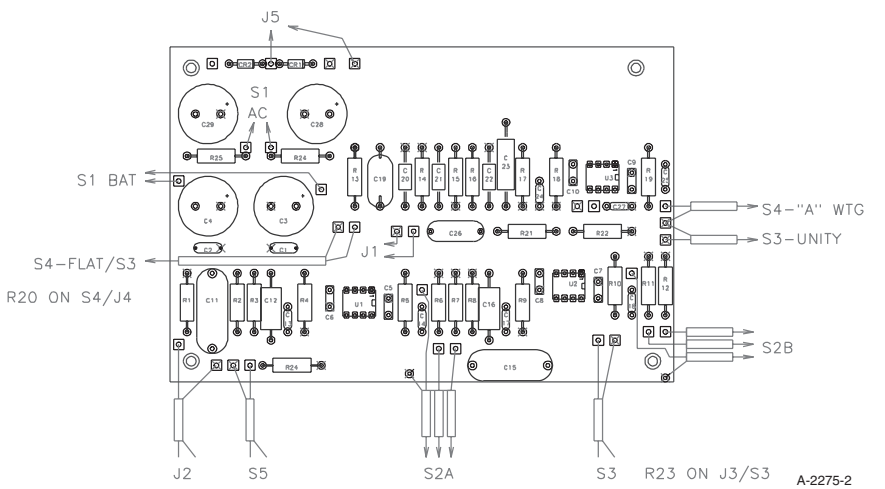
input (RTI) is 3.4µV RMS and 4.7µV RMS, respectively. This is the output noise divided by the noise gain of the amplifying circuit (quite a bit higher in the real world than the ideal calculations predict). *Table 2* summarizes the measured broadband and A-weighted output residual noise relative to 2V RMS, and the noise referred to the input (RTI), using battery power alone. *Table 3* shows the same data points using the AC adapter supply, which raises the noise floor.

Figure 4 shows the NMA frequency response relative to 2V RMS output. The unity gain input and the gain stage at 8k5 Rin curve are essentially flat. The gain stage with 600Ω Rin has more droop at the low end reflecting the interaction of the lower input resistance with the 4µ7 input coupling cap. Note that the gain stage curves have been normalized at 0dB for clarity (the dB gain has been subtracted out). The A-weighting filter response is shown, along with the published curve (dashed line).

Figure 5 shows the THD+N versus frequency for each of the gain settings, and for the unity gain input with and without the A-weighting filter. The input signal was adjusted in each case for 2V RMS output into a 10k load at 1kHz.

Figure 6 shows the THD+N versus output voltage into a 10k load at 1kHz. The solid lines represent battery power for all gain settings and unity gain with and without the A-weighting filter. The two dashed lines show the THD+N at 40dB gain and unity gain, using the AC power adapter as a source.

FIGURE 2: CIRCUIT BOARD PARTS PLACEMENT.



USING THE NOISE METER AMPLIFIER

You can use the NMA with any standard X1 oscilloscope probe. The output is connected to any broadband AC true-RMS voltmeter with a 1mV RMS full scale. The reading on the meter is then multi-

PHOTO 4: HP 3400A WITH COVERS REMOVED.

PHOTO 6: A2 BOARD AND SLIDES.

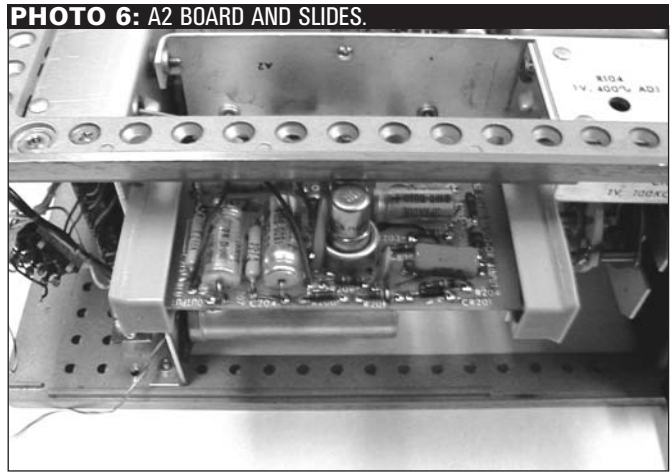


PHOTO 5: ATTENUATOR SWITCH WITH DUST.

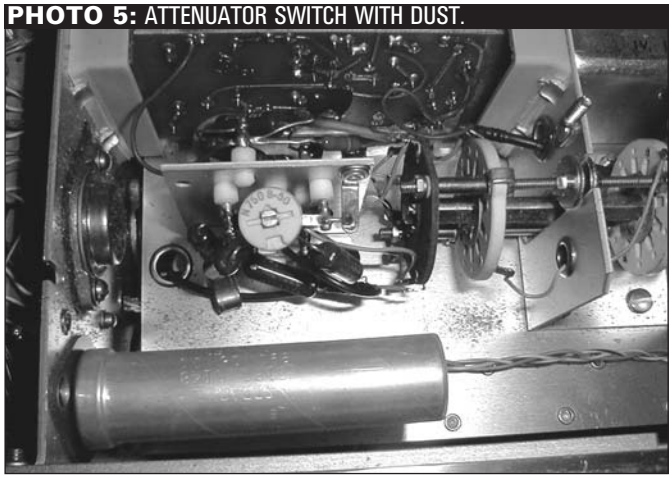


PHOTO 7: BOARD SLIDES WITH TAPE RESIDUE.

plied by the gain used in the NMA.

For low impedance sources such as power-supply noise measurements or for standard 600Ω pro audio loads, use the 600Ω Rin position. To measure the noise of higher impedance sources such as preamp line outputs, use the 8k5 Rin position to avoid overloading them; or you can use the unity gain buffer input for driving the A-weighting filter directly without any overall gain. I've listed a few more sources of noise measurement information in the References list[6–11].

FIGURE 3: FRONT PANEL LETTERING.

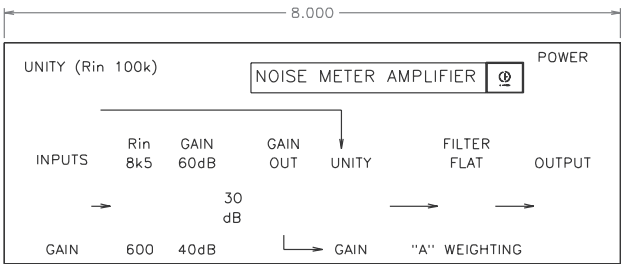


FIGURE 4: NMA FREQUENCY RESPONSE.

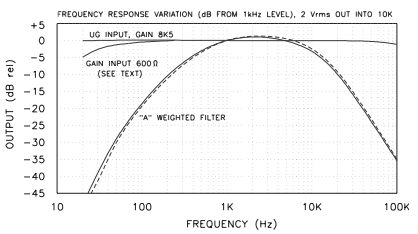
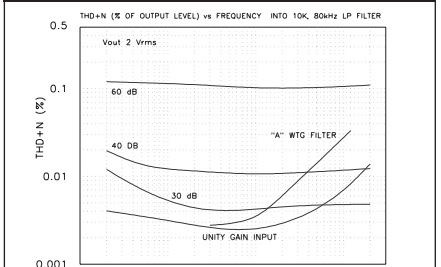


FIGURE 5: NMA THD+N VS. FREQUENCY.



The battery power mode with its ±9V rails has less headroom than the AC adapter mode with ±15V rails. *Tables 1* and *2* and *Fig. 6* show the noise and distortion headroom penalty for using the AC adapter.

THE DESIGN

A block diagram of the HP 3400A is shown in *Fig. 7*. The meter uses the highly accurate thermocouple method of RMS measurement. The AC input signal is applied to a 10MΩ input attenuator, which feeds the nuvistor cathode follower impedance converter. The converter drives a second 600Ω attenuator assembly that is switched in conjunction with the input attenuator to keep the nuvistor output voltage between zero and 100mV full-scale on all ranges.

The second attenuator drives a video amplifier, whose AC output is used to operate the heating element of one section of a sealed dual-thermocouple module. This thermcouple output is input to a modulator and chopper amplifier. The chopper output is demodulated, current amplified by an emitter follower, and drives the front panel analog meter.

Feedback is taken from the emitter follower back to the heating element of the second thermocouple in the module. The two thermocouple outputs are connected in reverse polarity and the

FIGURE 6: NMA THD+N VS. OUTPUT VOLTAGE.

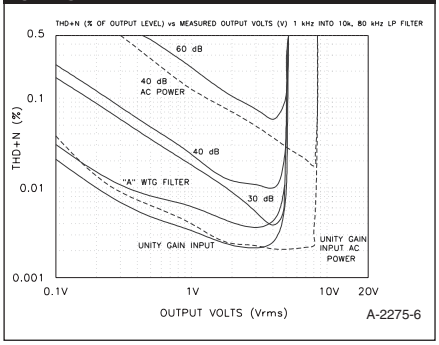


FIGURE 7: HP 3400A VOLTMETER BLOCK DIAGRAM.

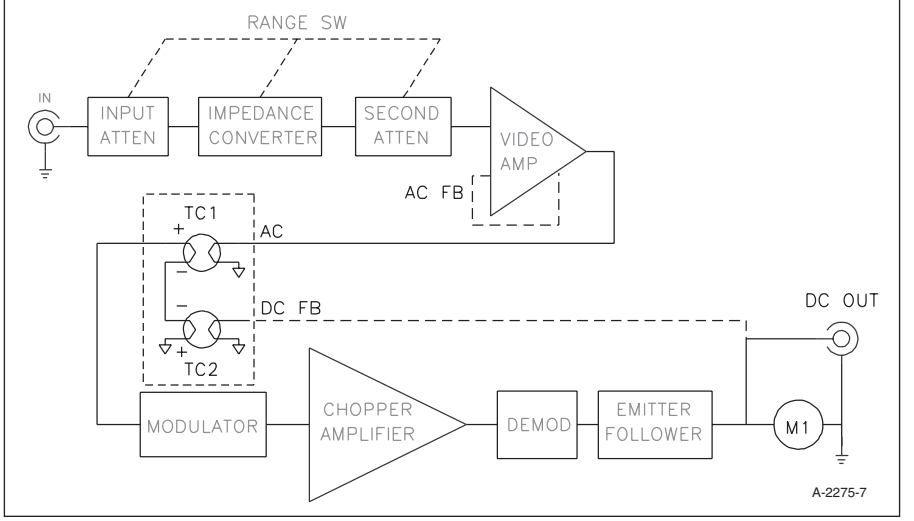


PHOTO 8: REPLACING FOAM IN BOARD SLIDES.

PHOTO 10: REMOVING REAR PANEL PLATE.

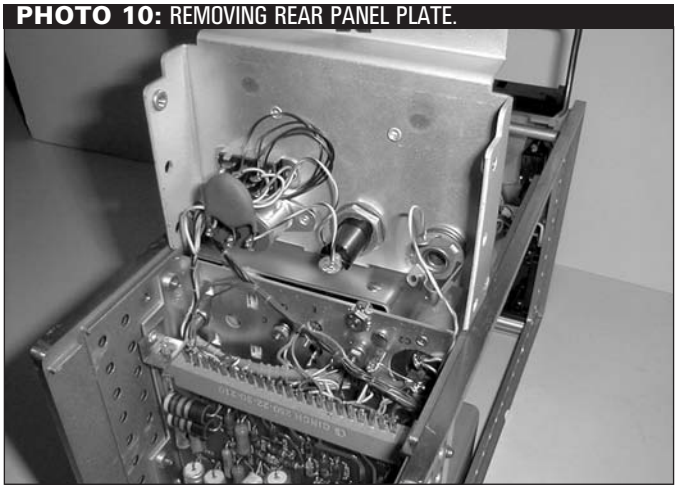


PHOTO 9: RE-CAPPED A2 BOARD.

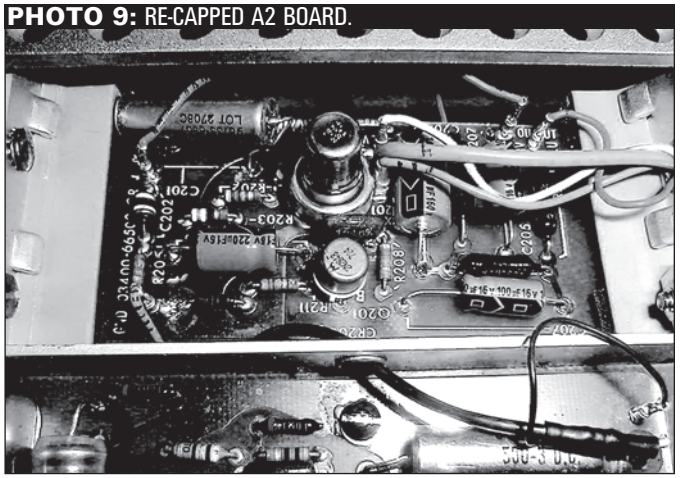
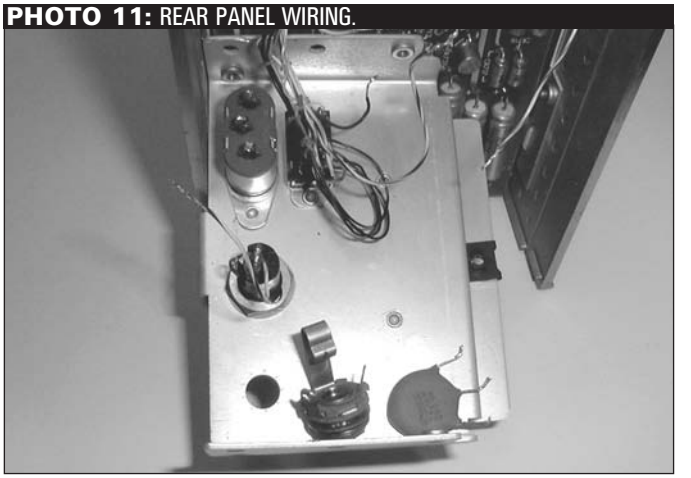


PHOTO 11: REAR PANEL WIRING.



closed loop meter circuit nulls the combined series thermocouple output to provide an accurate reading of the AC input signal regardless of its waveform or crest factor. The HP-3400A is accurate to 1% from 50Hz to 1MHz, and has 5% accuracy from 10Hz to 10MHz.

If you are selecting a 3400A, try to find one that is not "decorated" with the previous owner's lab sticker (other than calibration stickers). Ex-government agency and military units are especially likely to have painted stencils and large hard-to-remove stickers on them.

You may be able to renovate a 3400A using just my description and the accompanying photos, but troubleshooting and repairs will require the Operating and Service Manual. There have been a number of running changes to the 3400A throughout its production life. Appendix C in the latest manual covers these changes.

Units with serial number prefix 806- are the newest. Earlier prefixes were 322, 401, 528, and 714. Most of the changes pertained to the chopper amplifier A6 board, the A5 photochopper, and the thermocouple pair on A4. I find that even the older units can have very good performance.

One thing I can guarantee is that when you first remove the meter covers, it will be filled with the powdery residue of the deteriorated foam rubber used to shock-mount the A2 nuvistor board. The second most likely problem is lack of a power cord.

All-vacuum tube gear such as the HP

RENOVATING THE HP 3400A RMS VOLTMETER

The output of the noise meter amplifier must be fed to a wide-bandwidth true-RMS reading AC voltmeter. One such meter that is readily available on eBay and from surplus test equipment vendors is the Hewlett-Packard 3400A. It cost about $750 new in the 1960s, but clean working units are available on eBay for under $100.

The 3400A is all solid-state except for one RCA 7586 nuvistor triode, used as the impedance converter from the 10MΩ input attenuator network to the 600Ω output attenuator. (The later HP 400E AC voltmeter, not a true-RMS meter, was a descendant of the original HP 400 vacuum tube voltmeter, and uses a JFET impedance converter.) The HP 3400A can read from 1mV RMS to 300V RMS full-scale. The HP 3400A Operating and Service Manual and nuvistor tubes are also still readily available.

TABLE 2

Broadband and A-weighted Output Residual Noise Relative to 2V RMS, and the Noise Referred to the Input (RTI), using Battery Power.

NMA (BATTERY)	DBR OUT	DBR RTI
60dB 600Ω	–52.6	–112.6
60dB 8k5	–44.5	–104.5
60dB flat out	–44.5	–104.5
60dB "A" wtg	–61.9	–121.9
40dB 600Ω	–68.8	–108.8
40dB 8k5	–61.0	–101.0
40dB flat out	–61.0	–101.0
40dB "A" wtg	–81.9	–121.9
30dB 600Ω	–75.4	–105.4
30dB 8k5	–69.0	–99.0
30dB flat out	–75.1	–105.1
30dB "A" wtg	–92.0	–122.0
Unity Gain In	–100.0	–100.0
UG via A-wtg	–100.0	–100.0

TABLE 3

Broadband and A-weighted Output Residual Noise Relative to 2V RMS, and the Noise Referred to the Input (RTI), using AC Adapter Power.

88NMA (AC POWER)	dBr OUT	dBr RTI
60dB 600Ω	–38.6	–98.6
60dB 8k5	–40.1	–100.1
60dB flat out	–40.1	–100.1
60dB "A" wtg	–52.4	–112.4
40dB 600Ω	–60.7	–100.7
40dB 8k5	–59.6	–99.6
40dB flat out	–59.5	–99.5
40dB "A" wtg	–76.2	–116.2
30dB 600Ω	–72.2	–102.2
30dB 8k5	–68.2	–98.2
30dB flat out	–68.2	–98.2
30dB "A" wtg	–89.1	–119.1
UGB open	–100.0	–100.0
"A" wtg open	–100.0	–100.0

400D VTVM had attached power cords. The HP 3400A uses an old-style removable power cord based on the Switchcraft AC3G receptacle. These power cords are very hard to find because HP used them for only a few years in the mid-60s before the removable IEC cord became standard.

After a fruitless search for an HP power cord, my solution was to convert the meter to a standard IEC power receptacle. I have since renovated a number of 3400As the same way (this power receptacle modification also works for the earliest HP 400E AC voltmeters). The final step in my standard renovation was to replace the meter output signal phone jack with a more useful BNC jack.

GETTING TO WORK

Photo 4 shows the HP 3400A with the top, bottom, and side covers removed. There is a shield that covers the attenuator area in the bottom of the chassis and the nuvistor A2 board. While it looks complicated, it is held in place with one hex nut near the tube cutout and a second hex nut on a captured stud that fits in a slot in the plate with the power transformer mounted to it.

Photo 5 shows the attenuator switch area filled with the powdered foam from the A2 board card mounts. The first order of business is to remove the two plastic PC board holders. The A2 board and the two holders slide out of the chassis from the side (*Photo 6*). Be sure to guide the top of the nuvistor

PHOTO 12: REMOVING THE LINE VOLTAGE SELECTOR SWITCH.

PHOTO 13: PLATE WITH PARTS REMOVED.

PHOTO 14: ENLARGING THE CUTOUT FOR IEC RECEPTACLE.

PHOTO 15: INSTALLING PARTS AND BNC CONNECTOR.

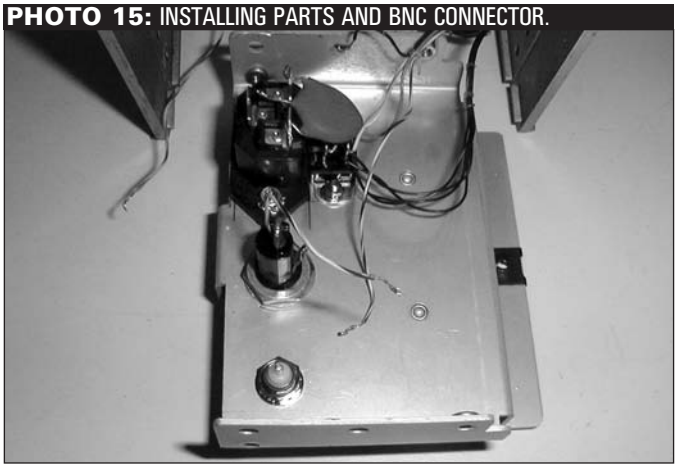


PHOTO 16: 3-LEAD CAPACITOR INSTALLATION.

PHOTO 17: ATTENUATOR SHIELD REPLACEMENT.

tube below the bottom left rail as you remove the A2 board to avoid damage.

Once the two slides are fully extended, you can remove them from the sides of the A2 board. All that will remain of the shock-absorbing foam will be the sticky back tape (*Photo 7*). Scrape the tape and the foam residue from out of each PC board holder.

Next you are going to replace the erstwhile foam with some ⅜″ wide 3⁄16″ thick self-stick high-density weatherproofing foam such as Frost King R338. Cut the replacement foam to length and use a razor knife to slit it down the middle, to within ¼″ of each end (not full length, see *Photo 8*). The slits will center the A2 board when you reassemble the unit. In order to properly hold the A2 board in position, you will need two layers of foam in the right holder to equal the total original foam thickness.

Next, use a soft paintbrush to clean all the foam dust out of the unit. It will have worked its way into every nook and cranny. Use a can of computer dust remover air spray if needed. You may even need to remove the plug-in PC boards to clean their chassis connectors.

Inspect the electrolytic caps on the A2 board. You may choose to re-cap it if they are deteriorated, as I did on one meter (*Photo 9*). You can easily remove the board by disengaging the push-on terminal wires from their pins. Be sure to make a drawing of the wires and their colors—I recommend you have the manual in hand for this.

REPLACEMENTS

The parts list in *Table 4* gives the Mouser part numbers for suitable replacements for all the aluminum caps. The level of capacitor technology in the 1960s was not what it is today. Most of those old aluminum caps had a tolerance of +100%/–10% or worse. Today ±20% is fairly standard. Replacing every aluminum cap is fairly cheap (about $15) but very labor intensive.

For the two large can caps (C1 and C2), you can connect replacement radial lead caps right across the old terminals and leave the original caps in place on the chassis. CE Manufacturing is also making replacement can capacitors for vintage audio gear (info@cedist.com).

If you don't have the HP power cord, you can easily enlarge the cutout for the original Switchcraft AC3G connector to accept a modern IEC receptacle. Remove the four flathead screws that hold the rear panel connector plate, making sure not to lose the cone-shaped lock washers. Slip the plate out of the chassis (*Photo 10*). Before removing anything, make a list of the wire colors for each of the following wires (*Photo 11*).

Unsolder the white/green/gray wire to the phone jack and remove the jack. Unsolder the white/brown/gray harness wire to the fuse holder (no need to remove the one to the power socket). Unsolder and remove the 3-wire ceramic capacitor on the back of the power socket. Finally, unsolder the three wires from the socket (white/brown/gray, white/gray, and black).

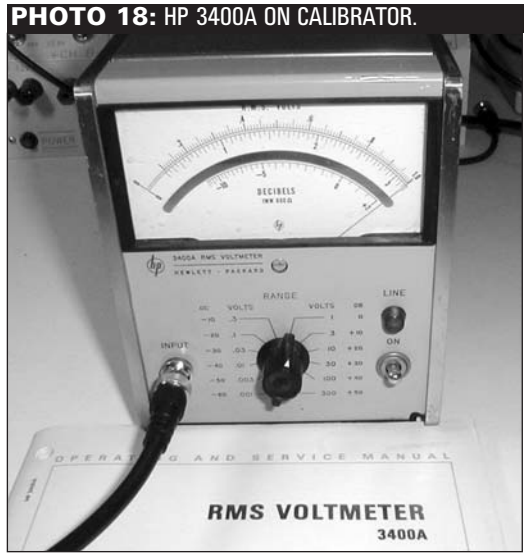
The line voltage selector switch is riveted to the back plate and must be drilled out so you can remove the switch from the plate without having to remove all its wires. I used a 13⁄64″ drill (*Photo 12*). It turned out to be easier to also remove the fuse holder for better access. Be very careful and support the rear plate with care. The wiring used in

the 3400A is fragile #24 and #26 and can easily be broken.

Once the switch is removed, the plate is free from the chassis. Now drill out the two rivets holding the AC socket, and the plate will be clean of all components (*Photo 13*).

Now you must enlarge the AC socket hole to accept the IEC power receptacle. Scribe an accurate horizontal line on the inside of the rear plate exactly 1/16″ up from the bottom of the plate. This will be the bottom of the new opening for the IEC receptacle. Next lay out scribe lines for the IEC socket outline, using the outside mounting hole to locate its horizontal position. I used a Radio Shack 64-823A nibbler tool to quickly enlarge the opening in the thin aluminum (*Photo 14*). Be careful not to bend or damage the thin 1/16″ web that is left on the bottom of the opening.

PHOTO 18: HP 3400A ON CALIBRATOR.



I used 6–32 screws to mount the new IEC receptacle. You will need to enlarge the old outside rivet hole and drill a new hole for the inside screw. Use flat washers under the heads of the screws and lock washers inside, and install the IEC receptacle with its center ground pin at the bottom of the rear plate. Next install the line voltage switch with 4–40 screws, flat washers, and lock washers. Finally, replace the fuse holder and install the BNC connector in place of the phone jack (*Photo 15*).

Now you need to install the 3-lead ceramic cap onto the IEC receptacle terminals, without soldering the leads yet. The cap should point upward from the bottom of the chassis. It must fit between the receptacle and the green connector for the A7 board at the back of the unit. If this cap is damaged, you can replace it with two individual 10nF 250V AC ceramic disc caps.

Next solder the wires you removed earlier back onto the IEC receptacle, the fuse holder, and the BNC jack. Carefully route the wires so they are not pinched between the green connector and the 3-lead capacitor (*Photo 16*). Replace the four screws that hold the rear panel in place.

Before replacing the attenuator shield (*Photo 17*), coat the rotary switch contacts with Caig deoxiT or other good contact cleaner. *Photo 18* shows the 3400A alive and well on the calibrator. Calibration is fairly

TABLE 4

HP3400A Renovation Parts List

SYMBOL	ORIG. VALUE	DESCRIPTION	NEW VALUE	MOUSER PART NO.	QTY
C1	40+40µ 200V	Alum can	47µ 160V	647-UVZ2C470MHH	2
C2	890µ 15V	Alum can	1000µ 16V	647-UVZ1C102MPH	1
C612	880µ 1V	Alum	1000µ 10V	140-XAL10V1000	1
C205, C412, C609, C613	100µ 12V	Alum	100µ 16V	140-XAL16V100	1 each
C207	100µ 15V	Alum	100µ 16V	140-XAL16V100	1
C204, C702, C704	10µ 150V	Alum	10µ 160V	140-XAL160V10	1 each
C411, C606	10µ 10V	Alum	10µ 16V	140-XAL16V10	1 each
C602	10µ 12V	Alum	10µ 16V	140-XAL16V10	1
C610	1µ 25V	Alum	1µ 100V	140-XAL100V1.0	1
C203, C407	200µ 3V	Alum	220µ 16V	140-XAL16V220	1 each
C427	200µ 15V	Alum	220µ 16V	140-XAL16V220	1
C413, C415	20µ 25V	Alum	22µ 25V	140-XAL25V22	1 each
C605	2µ 25V	Alum	2µ2 50V	140-XAL50V2.2	1
C712	35µ 250V	Alum	33µ 250V	140-XAL250V33	1
C402	500µ 3V	Alum	470µ 10V	140-XAL10V470	1
C708, C709	50µ 25V	Alum	47µ 25V	140-XAL25V47	1 each
C706, C707	40µ 50V	Alum	47µ 50V	140-XAL50V47	1 each
J2		1/4″ phone jack BNC		523-31-221-RFX	1
J3	AC3G	AC receptacle IEC		161-0707-1	1

straightforward, but you will need the manual to perform all the steps. Depending on the IEC receptacle you select, you may need to use the nibbler to notch some clearance in the bottom cover where it wraps around and sits against the receptacle flange. *aX*

REFERENCES

1. Elliot, R., Fehringer, J., "'A' Weighting Filter for Audio Measurements," Elliot Sound Products, Project 17 update 21 May 2000, sound.westhost.com/project17.htm.

2. Trolliet, P., Audio Aids, "'A' Weighting Filter," *TAA* 3/81, pp. 25–26.

3. Stamler, P., "The Tape-Tweaker's Pal, Part I," *TAA* 5/82, p. 19.

4. Metzler, B., Weighting Filter Responses; *Audio Measurements Handbook*, Audio Precision, 1993, Figure 5, p. 16.

5. Clarke, B., "Find Op Amp Noise with Spreadsheet," Application Note AN-253, Analog Devices Inc.

6. Metzler, B., Noise Measurements, Audio Measurements Handbook; Audio Precision, 1993; pp. 15–22.

7. Bohn, D., Signal-to-Noise Ratio, Equivalent Input Noise, Audio Specifications, RaneNote 145, Rane Corp. pp. 3–4.

8. Thompson, B., "Calculate and Measure Noise Values," *Test and Measurement World*, August 1, 2001, pp. 65–71.

9. Gruchalla, M. "Measure Wide-Band White Noise Using a Standard Oscilloscope," *EDN*, June 5, 1980, p. 157.

10. Motchenbacher, C., Connelly, J., "Noise Measurement," Chapter 15, "Low Noise Electronic System Design," Wiley-Interscience, 1993.

11. "Op Amp Applications," Analog Devices Seminar Series, edited by Walt Jung.

SOURCES

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cd review

Back

By Gary Galo,
Regular Contributor

***A Matter of Coincidence**. Recorded by Brian Preston. (1 CD, 71:51).* Old Colony Sound Lab, Peterborough, NH 03458. $14.95. Part # CDBP1.

***Boston Audio Society Test CD—1**. Recorded by members of the BAS. (1 CD, 71:40) Boston Audio Society, PO Box 211, Boston, MA 02126; 603-899-5121; http://bostonaudio.home.att.net, E-mail: dbsystems@attglobal.net.* $30. Part # CDBAS1.

Both CDs available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458-0876; 603-924-9464; 603-924-9467 (FAX); www.audioXpress.com; E-mail: *custserv@audioXpress.com*.

These two CDs offer a wide variety of "purist" recordings that depart radically from most of those made by the large commercial record labels. *A Matter of Coincidence* (*Photo 1*) was recorded in the UK by Brian Preston, and offers 23 tracks of recordings made, almost entirely, with two microphones in coincident or quasi-coincident pairs. For those unfamiliar with this terminology, there are basically two methods for making stereophonic recordings with the fewest number of microphones, both dating back to the earliest days of experimental stereophonic recordings.

PHOTO 1: A MATTER OF COINCIDENCE.

PURE COINCIDENT TECHNIQUES

Coincident recording involves using two directional microphones with the capsules aligned vertically, but angled in the horizontal plane (usually 90°). The stereophonic information is derived entirely from amplitude differences between the left and right microphones. Vertical alignment of the microphones ensures that there are no phase differences between the two channels. Coincident recording originated with the experiments of the British engineer Alan Blumlein in the late 1920s and early 1930s, and many Europeans pursued variations on the coincident theme as stereophonic recording evolved and eventually became a commercial reality. Blumlein's coincident method consisted of a pair of crossed figure-8 microphones at 90° angles.

Beginning with the stereophonic ex-

periments conducted by Bell Labs in the early 1930s, the majority of American recording engineers have opted for the spaced microphone approach to stereophonic recordings. At its simplest, this technique involves two spaced microphones, usually omnidirectional. The spacing means that the stereophonic information is derived from both amplitude and phase difference between the two channels. With two spaced microphones, it is often difficult to attain a precise image in the center of the soundstage.

To overcome this problem, the omnidirectional microphone array is often expanded to 3, with the center microphone fed to both the left and the right channels. The "3-spaced-omni" approach was made famous by Mercury in the 1950s and early 1960s. In the late 1970s Telarc adopted this same approach.

Recording with two microphones, regardless of the configuration, means a simpler signal path, since a simple 2-channel microphone preamp is all that is needed between the microphones and the recorder. Expanding the number of microphones to three means that you must use some type of microphone mixer.

The subtitle to *A Matter of Coincidence* tells us that the CD contains "A wonderful collection of selections recorded using a variety of two-microphone coincident methods." The disc contains a variety of fine recordings made using various coincident methods. Schoeps microphones crossed at 90°, both hypercardioid and cardioid, were used for most of these recordings.

SELECTIONS

Most of the recordings are classical, and Preston was fortunate to have a number of very fine musicians at his disposal. The first selection on the disc is a performance of Handel's Suite in D Major (with material from the familiar *Water Music*), with Crispian Steele-Perkins' stylish Baroque trumpet accompanied by organist Gerald Gifford. The recording is actually what should be called "near-coincident," made with spread Schoeps hypercardioids at 90°.

There are several recordings on the disc made with the spread microphones, all angled at 90°. This one is extremely clear and detailed with a solidly placed stereo image. Unfortunately, there is no information on just how far the microphones were spread. There are no examples of the ORTF method, which is two cardioids spaced 17cm, and angled at 110°.

Preston seems to favor a close perspective on his pure coincident recordings. Many of these are too close for my taste. As an example, in a movement from Beethoven's Piano Trio in C-minor, Op. 1, no. 3, each instrument sounds as though it is in its own acoustic cubicle. The recording doesn't allow the instruments to blend into the acoustic space before being captured by the microphones. This recording was made with spread Schoeps hypercardioids.

An all-acoustic Irish specialty group called The Shortwaveband is featured on several selections, most made with Schoeps hypercardioid microphones crossed at 90°. Most engineers would, by force of habit, put a microphone on every player in the group—Preston's purist recordings show how unnecessary multi-miking often is with acoustic instruments. A couple of the selections

are overdubbed, but the overdubbing was done in the same venue, with the same microphones, so the sense of the original space is preserved.

Despite the reference to Blumlein, only two out of the 23 recordings were made with crossed figure-8 microphones (Schoeps). These, in my opinion, are the best recordings on the disc. Two recordings made in the Church of Saint Mary de Lode, Gloucester, feature organist Gerald Gifford, and offer a very realistic perspective on the organ (and the anonymous flute soloist on one of the tracks). They are not too close, and capture a palpable sense of the recording venue.

Beyond the microphones, no details are offered on the rest of the recording equipment, which is unfortunate. Preston doesn't offer any orchestral recordings on the disc, but finding an orchestra to record is not always easy. *A Matter of Coincidence* does offer a number of insights into the possibilities available to the recording engineer using only two microphones. (Note: A portion of the royalty paid on this recording goes to the Royal Society of Musicians of Great Britain Reg. Charity No. 208819.)

BAS RECORDINGS

The Boston Audio Society's *Test CD—1* contains 11 musical recordings made by members of the society, many of them well-known in the audio field. This disc is particularly interesting since it features recordings made using a variety of purist techniques, with both near-coincident and spaced microphones. None of the BAS recordings were made with pure coincident techniques.

A choral recording by David Hadaway, the long-time proprietor of db Systems, features "Jabberwocky" by Sam Pottle performed by University of Massachusetts Chamber Choir conducted by E. Wayne Abercrombie. This recording was made with a pair of spaced Shure SM80 omnis. The recording is noteworthy for the extremely wide soundstage.

After the undoctored recording is heard, the next track features the conclusion of the same recording with Lexicon digital reverberation added. The "enhanced" recording is not nearly as good as the original—the precision of soundstage localization and the sense of air and space in the recording disap-

pear, and the natural sound of the recording venue is replaced by a strangely "dry"-sounding digital reverb.

A more successful Lexicon enhancement is a movement from Soler's Sonata in G-minor, performed by harpsichordist Irma Rogell. The recording was made in a living room with a pair of spaced Schoeps CMC-56 omnis. The recording is clean and detailed, without obvious side effects from the digital reverb.

Engineer Micha Shattner offers an excerpt from the 2nd movement of the Saint-Saëns *Organ Symphony*, with the Boston Civic Symphony conducted by Max Hobart and organist James David Christie, recorded in Jordan Hall. This recording was made with the ORTF method, using a pair of AKG C-414 microphones in the cardioid position. The stereo image is very precise, and the low end of the organ is most impressive. The booklet warns the listener not to play the recording too loud with vented or planar loudspeakers.

Shattner also recorded the excerpt from Mahler's *Das Lied von der Erde*, in a chamber arrangement. This recording, made with two Schoeps CMC-56 omnis, is both spacious and detailed, with a good center image.

Two other orchestral recordings are included on the BAS disc. David Hadaway has been recording the New Hampshire Symphony Orchestra since 1977. The recording featured here is the conclusion of Bruckner's Symphony No. 4 in the original version, conducted by James Bolle, the orchestra's founder. The recording was made with analog equipment and Dolby-A noise reduction, using a pair of Nakamichi CM-1000 cardioid microphones. Although the booklet doesn't specify, the recording sounds as though the microphones were spaced.

The NH Symphony is a respectable provincial orchestral but their hall—the Palace Theatre in Manchester—is very dry. The recording is much like listening to a large orchestra in a recording studio, but Hadaway's recording offers a seemingly realistic picture of the ensemble in that acoustic space. I disagree with his comment that "one hardly notices (the dry venue) since the music never stops."

As a tribute to the late Peter Mitchell, founder of the BAS, *Test CD—1* includes the "Dies Irae" and "Mors stupebit" from the Verdi *Requiem*, with the Boston Philharmonic Orchestra conducted by Benjamin Zander. The recording was made by Mitchell and E. Brad Meyer in 1981 using four Nakamichi 700-series microphones: two omnis for overall pickup and two cardioids at lower levels to accent the chorus. The recorder was Sony's PCM-1, a 14-bit digital encoder that used videotape as the storage medium. To overcome the low-level linearity problems inherent in 14-bit recording, they used a custom-built dbx I noise reduction unit with a mild 5:1 companding ratio.

The recording is impressive for the uncompressed dynamics and the impact of the bass drum. Otherwise, the recording shows its age. It is much drier than I would expect from Symphony Hall, with no air in the treble. I suspect that these problems were exacerbated when the original was transferred to a 16-bit PCM-F1, which required a 14-bit D/A conversion, dbx decoding, and 16-bit A/D conversion in the F1.

One choral selection, "At Christmas All Be Merry," was recorded using Lex-

icon's Logic 7 system, which allows a 5-channel recording to be mixed down to two while retaining the original balances. No details on microphones are given for this recording made by David Griesinger, but the balances and the image are very successful in the 2-channel mix down. The recording has a bit of treble hardness.

DOCUMENTATION

E. Brad Meyer is the engineer on a wonderful organ recording, an arrangement of Bach's Concerto in A-minor (after Vivaldi), performed by James Johnson at the Busch-Reisinger organ at Harvard University. This recording was made with four Nakamichi CMC-700 microphones: two cardioids in the center, in an ORTF configuration, and two omnidirectional outriggers. Meyer notes that the recording was made during warm and humid weather, which increased the high-frequency reverberation time in the hall. I find the perspective just right, an excellent mix of detail and natural hall acoustic. The BAS booklet does not give details of the microphone array—I needed to consult the *Stereophile Test CD* (STPH-002-2) for this information (Meyer and Peter Mitchell also made the recording on the *Stereophile* CD, with the same microphone setup as Meyer mentions in the notes to the BAS CD).

There are some other problems with the documentation supplied with the BAS CD. Track 10 is billed as Mahler's *Das Lied von Der Erde* arranged by Arnold Schoenberg. It would be difficult to perform this work in the 4½ minutes included on this disc. In fact, we are given the conclusion of the last movement, "Der Abschied." Schoenberg abandoned his chamber arrangement in 1921 and never returned to it. The scoring was completed in the early 1980s by Reiner Riehn, who should have been credited by the BAS. They probably should have also credited Universal-Edition, Wien, the copyright holder of this arrangement.

Track 12 is a 10-minute excerpt from the program "Shop Talk," which originated from WBUR-FM in 1984, featuring E. Brad Meyer, Peter Mitchell, and Richard Goldwater. The booklet tells us that the program begins with a 1932 stereo recording, and Meyer tells us that it was made by Bell Labs on March 12 of that year. No mention is made anywhere of the work performed, or the musicians involved. It is, in fact, an excerpt from Scriabin's *Prometheus—The Poem of Fire*, a live performance by the Philadelphia Orchestra conducted by Leopold Stokowski, issued on LP in 1979 by Bell Laboratories, BTL-7901.

This record, and its companion BTL-8001, contain experimental high fidelity and stereophonic recordings made by Bell Labs in 1931 and 1932, all with Stokowski and the Philadelphians, and are well worth seeking out on the used market. The excellent transfers put Ward Marston on the map as one of the world's foremost engineers in the field of historical recordings (Marston is also uncredited).

The chorus and bass soloist for the Verdi *Requiem* excerpt are not identified. None of the performers, or the recording venue, are given for "At Christmas All Be Merry."

My other quibble with the notes is admittedly a matter of personal opinion. The booklet states that the conclusion of Bruckner's Fourth Symphony was performed in the original version, "before the revisions by well-meaning friends." In the case of the Fourth Symphony, the extensive revisions were done by Bruckner himself, including replacement of the Scherzo and two wholesale rewrites of the last movement.

In its original version, this work has substantial structural problems, with long stretches that meander rather aimlessly. The composer's final thoughts resulted in a far more coherent work than the original, earning it a place in the standard repertoire. Bruckner's final version is not to be confused with the first published version of 1889, edited by Loewe, that admittedly wreaked havoc on the work and has long been discredited.

TEST TRACKS

The rest of the *Boston Audio Society Test CD-1* consists of test tracks for equipment and system evaluation. Most seem to be intended for ear-only evaluation, such as stereo and mono wideband pink noise tracks (one "out-of-phase"), and ⅓-octave mono pink

noise. Sweep tracks are included for subjectively evaluating your system/room response and/or your hearing. The "Low-Level Crackle Tests" consist of a background noise, like someone fiddling with their program at a concert, repeated five times, each time 10dB lower than the previous. The notes say that if you can hear the fifth repetition, which is at the lowest level that can be recorded on a CD, you may need more dynamic range than a 16-bit system provides.

Two of the tests are liable to confuse the user. A "picket fence test" contains "A sum of sine waves at all ⅓-octave centers from 20Hz to 20kHz, except for omitted 1.25kHz and 3.15kHz. It is a severe test for bit-reduced systems (like MP3) since it contains energy spread across the spectrum." There's no hint of how to use it or what to listen for.

The "Multitone Test," suggested by Robert Cordell, has three tones of equal amplitude, 20.00, 10.05, and 9.00kHz. The distortion components are difference tones at 950 and 1050Hz. The booklet says that these tones "allow easy measurement of both even- and odd-order distortion without the use of a spectrum analyzer. The suggested test fixture comprises two 3rd-order low-pass filters at 2kHz followed by a 4th-order bandpass filter centered at 1kHz, with gain in between each pair." Beyond that, there is no information on how the test is to be used, and what might constitute acceptable results.

The musical recordings on the Boston Audio Society *Test CD-1* complement those on *A Matter of Coincidence*. Audiophiles interested in some of the results possible with purist stereo recordings are likely to find both of these discs interesting and informative, and a refreshing change from the excessive number of microphones used on many commercial recordings. Both discs are CD-Rs, rather than manufactured CDs. *aX*

TEST TRACKS

What music selections do you use to test your audio setup? Simply describe your seven favorite pieces (not to exceed 1,000 words): include the names of the music, composer, and manufacturer (and number), and send to TEST TRACKS, audioXpress, Box 876, Peterborough, NH 03458, e-mail editorial@ audioXpress.com.

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OUTPUT BIAS CIRCUITS

In Dan Stanley's otherwise well-written article ("A High-Quality MM IC Preamp," July '04), there is an error in Fig. 3, which purports to show an "FET Cascode output bias" circuit. As connected in the drawing, the 100Ω resistor does essentially nothing, as it appears in series with the much higher impedance of the upper FET drain. Its current will be fairly ill determined based on the properties of the lower FET at low drain-gate voltage, and that voltage will depend on the characteristics of the upper FET.

A more appropriate circuit would place the resistor in series with the source of the lower FET, and the upper FET would be selected to have a larger magnitude of pinch-off voltage and saturation drain current compared to the lower FET. As well, a resistive voltage divider between the negative rail and ground could be provided for the gate of the upper FET, to ensure that its source would be sufficiently positive with respect to the negative rail to provide some room for operation of the lower device. The gate of the upper FET should be biased as close to the negative rail as provides proper operation, lest the operation of the circuit cease at large negative output voltage excursions.

Actually, a current sink based on bipolars would be a generally more satisfactory solution, given the availability of well-regulated rail voltages.

Brad Wood
Chatsworth, Calif.

Dan Stanley responds:

The brief reference that I made in my article concerning the use of a FET-based class A output bias was primarily intended to inform prospective builders that better methods than a pulldown resistor exist for biasing. To this end, I included a reference to a very informative Internet article on the subject of biasing for op amps. The 100Ω resistor in my article circuit isolates the capacitance of the FETs from the op amp, but I do not believe that this circuit necessarily represents the highest performance approach to output biasing.

Since I did not actually evaluate the sonics of the FET configuration illustrated in my article, I strongly suggest that readers review the referenced Internet article and search several of the well-known audio forum sites before deciding exactly how they might choose to approach using an improved output biasing circuit. I am sure that there would be welcome interest among readers in seeing diagrams of FET and bipolar op amp output bias circuits that you believe would offer higher performance for this type of application. Even using just the pulldown bias resistors, careful builders will find the sonic performance of this design to be quite good.

AUDIO EVALUATIONS

I know it's foolhardy to go anywhere near a dustup between two silverbacks of the audio world such as Reg Williamson and Ed Dell (Xpress Mail, 9/04 *aX*, p. 56). But I'll risk being crushed by one or both of them, to sug-

gest that perhaps they're both right about objective and subjective evaluation of audio equipment, if you take into account the audience for each. Subjective evaluation is important to the prospective *user* of the equipment, while objective evaluation is important to the *designer* of it.

As a prospective purchaser, I want to know *what* a piece of equipment will sound like, so I read the opinions of careful, experienced people who have listened to it. Designers need to try to figure out *why* the device sounds the way it does, and what to do to make it sound different, so they take measurements and try to correlate the measurements with what they hear, and with design parameters they could change.

Here's an example. On page 14 of the 9/04 *audioXpress*, I read:

"Prior to final testing, I lightly stuffed the enclosure with 14 oz of poly fiber fill, which reduced the closed box Q from 1.5 to 1.4 and closed box resonant frequency from 40.2Hz to 37.3Hz. Directionally, this should improve system performance since another basic servo design principle is to develop the best open-loop response possible before closing the loop."

This paragraph is accompanied by graphs showing the subwoofer closed- and open-loop waveforms at 25Hz.

Here, the author used measurements (of Q, resonant frequency, the mass of the added poly fiber, and waveforms) to help decide what to do (add or remove poly fiber) to make the device sound more like he wants it. The measurements were very helpful to him as a designer, as well as to readers who design. But if I were thinking about buying the subwoofer from him, I'd rather have the subjective opinion of someone who had listened to it than the objective measurements of Q, mass of fiber, and so on.

I suppose we could do away with subjective reviews if we knew exactly what to measure and how to interpret the measurements. But we don't, at least not yet.

I remember years ago when *Con-*

sumer Reports evaluated loudspeakers. They hooked each speaker to a sine-wave generator and measured the response over the audible range of frequencies. They then came up with a single number for each speaker that showed how close its response was, on average, to flat. You just needed to go down the list and pick the one with the highest number you could afford. No subjective evaluations required. Heck, no ears were even required.

But people who dinked around with audio knew that frequency response wasn't the only factor to consider. You could make the speaker sound better if you tilted it back a little—same frequency response, better sound. Designers started measuring and modifying their designs for proper time adjustment. In his letter, Mr. Williamson gives an excellent example having to do with amplifiers. People didn't used to know how to measure phono input stage response until they figured out that you needed to include a phono cartridge in the test circuit.

Thanks to people such as Mr. Williamson, we can now design better products more easily. But we still don't know everything that needs to be measured, how to measure it, or how, exactly, to correlate the measurements with what people hear.

Until we do, I'll continue to rely on expert subjective opinion, and let designers make use of objective measurements. If I care to know how comfortably a car rides, I'll read the opinion of someone who's ridden in it. I don't care about the Q and resonant frequency of the suspension, but the designer had better give one.

I don't buy wine based on lab analyses, either; I read Robert Parker. But vintners certainly use measurements to design and produce their products. And until someone can look at one of those waterfall plots and tell me exactly what a speaker will sound like, I'll keep reading reports from listeners.

Ken Craven
ax@kcraven.com

DSP TECHNOLOGY

My thanks to Rune Aleksandersen for his article on using DSP technology to produce loudspeakers with flat frequency and phase response ("Room Correction, Part 1," 8/04 *aX*, p. 6). I have been trying to achieve the same results, but with analog technology, and his approach seems better. However, I do have some comments on his article:

The summed response of the high-pass filter and the low-pass filter is not truly flat, having about 1dB of ripple. This may be due to either or both of the following:

1. The filters are not symmetrical; that is, the low-pass has a much steeper cutoff than the high-pass.
2. The filters are said to have a 500Hz cutoff, but from Fig. 8 the low-pass seems to have a 6dB cutoff closer to 550Hz.

What is desired is a set of filters whose sum adds up to unity over the audio band. One example of this would be the Linkwitz-Riley filters with phase correction.

Also, there is a discrepancy in the THD curves. Figure 12 shows the THD as being generally below 0.01%. Figure 14 shows the 5th and 7th harmonics peaking at about 0.02%. If the 5th and 7th harmonics are each 0.02%, then the THD should not be less than that. 0.02% is very good performance, but I would like to know which measurements to trust, or else I cannot trust any of them.

Dick Crawford
dickcrawford1934@aol.com

Rune Aleksandersen responds:

Thanks to Mr. Crawford for his letter. Regarding the 1dB ripple, my evaluation of the high- and low-pass filtering was a very simple and quick one. I did not spend much time optimizing the amplitude response; the cutoff frequency of the filters were moved around a bit until the amplitude became reasonably flat. Most important is the perfect time response obtainable from the digital filters, an impossible task for any analog filter but the first-order filter.

The THD measurement shows 0.02% around the 4–7kHz range. I've been a bit enthusiastic. My text should have used the

words "mostly around 0.01%." This would be a more correct statement.

HANDS-ON AUDIO

The sentiments expressed in Mr. Fullmer's letter to the editor ("Electronic Sentimentalists," *aX* 8/04, p. 66) echo my own. I, too, have been involved in the design of audio equipment, beginning with amplifiers built out of old radios while in junior high school back in the mid-1950s.

The nonsensical hysteria over the "sound" of cables and capacitor dielectrics is sad; after all, the bottom line is generally one's enjoyment of reproduced program material . . . seemingly impossible to spoil with any amount of silly clock jitter short of that which would generate an appalling bit-error rate.

audioXpress is one of the few remaining "hands-on" publications that address our interests in hands-on audio. Gone are *Radio & Electronics*, *Popular Electronics*, and the like; and what used to be quite interesting under the name of *Wireless World* seems about ready to bite the dust. Even the AES *Journal* isn't much fun to thumb through anymore.

I attribute much of this to the immediate availability of inexpensive, good-quality, ready-made gear. It's no longer necessary to wire tube sockets with twisted filament leads (grain-oriented and oxygen-free, of course!) to drive a pair of speakers properly. Indeed, the area in which *audioXpress* excels, in my opinion, is in the speaker-building department, where innovative ideas are not duplicates of off-the-shelf items.

Perhaps some of us old-timers ought to pen the occasional article for something truly useful, rather than for some esoteric unity-gain, class-A buffer stage. I'm confident that such efforts would be welcomed by the *audioXpress* editorial staff, and might even foster a renewed interest in useful and practical DIY projects. As an employer in the industry, I often wonder where the next generation of engineers with practical, "tinkering" experience will come from (India seems likely).

I hope you will reconsider renewing your subscription; you will be missed as a reader and as a possible and, I'm sure, welcome, contributor.

Jim Wood, Pres./Chf.Engr.
INOVONICS, INC.
Santa Cruz, Calif.

Richard K. Fullmer responds:

I am flattered to see that Mr. Jim Wood of Inovonics was motivated to reply to my comments. He is absolutely correct in his description of the current "audiophile" hard-pressed to achieve any real audible improvement from home-built electronics, assuming he is willing to spend a reasonable fee for good-quality factory-built equipment. Forty plus years ago there was very little high-performance audio electronics available, without spending well over five-hundred real 1950 dollars.

When Heath, Eico, and later Dynakit made very good equipment available for under $100, there was a ready market of people who wanted to get the best out of the new LP recordings, some of which were quite good. Not only did most of the kit builders succeed in getting their rig working, a new consumer hobby market was produced, in which people were look-

ing for, and buying, upgrades that were audibly superior to anything on the consumer market.

As always, the weakest link was the loudspeaker. It was here that creativity reigned supreme, and anyone with a sheet of ¾" plywood and a copy of Audiophile, Radio-Television News, Popular Science, *and so on, could indulge in speaker building. Individuality and "the sound" were discussed with gusto, and there were very few speaker systems of this era that would be considered respectable by today's standards. It was great fun! Hardly anyone published any meaningful data. What measurements that were taken were mostly locked up in a file at the labs of the few manufacturers who owned an anechoic chamber.*

In about 1960, with the advent of Villchur's AR-1, and Thiele's papers in the early '70s, the low-frequency performance of moving coil speakers became somewhat predictable. The individual "sound" of different manufacturers' loudspeakers was becoming less of an art, or an asset.

At the same time, solid-state audio electronics started to appear. After an oversupply of really awful "transistorized" power amplifiers, the kit builders had another heyday, and could build a dependable, and very respectable sounding power amp for less than $300. Kit building was still fun. However, the design-it-yourself era was drawing to a close for most amateur audiophiles. Some good loudspeakers could now be had for less than $600/pair (1980 dollars), and home-designed transistor equipment tended to become smoky. We could still massage our records and cartridges with various emollients, and a quiet LP recording was a rare gem worth seeking.

But the inscrutable future was in the wings. The hobbyist was suddenly attacked on both flanks by digital recording, followed by the CD, and the dreaded DSP. The consumer-level implementation of all this was made possible by large-scale circuit integration. It is hard to become very sentimental about a lump of black potting compound, which is the ultimate in inscrutability. And the final nail in the coffin for the creative audiophile is that good factory stuff is relatively low in cost, and 99.99% dependable.

Try that with your Mark III Dynaco, which sounds great and glows in the dark wonderfully well, but will use up a pair of $100 KT-88s in six months. Ditto for the Marantz power amplifiers.

Fortunately, really good reproduced sound is still elusive, but not due to mysterious electron encounters with copper molecules in the wire. The acoustics of rooms is a major factor, made even more apparent with the relatively uniform performance of most good equipment. Unfortunately, the home decoration factor is probably responsible for more bad listening experiences than all of the metaphysical murmurings and nasty electrolytic capacitors now in circulation. It is still an engrossing hobby.

Please, let us search after truth. In this climate of "diversity" and universal acceptance of any and all philosophies, we are better served by some serious objectivity and common sense.

CLEVER CIRCUIT

At the risk of being drummed out of the corps, I would like to direct readers to an article that appeared in another publication, *EDN Magazine*, a "qualified circulation" biweekly that is sent without charge to design professionals in the mainstream electronics industry.

While hi-fi audio is not a subject typically covered by *EDN*, the magazine does incorporate a regular Design Ideas section that showcases circuit hints and shortcuts sent in by readers. Every so often an article will appear that has application in our domain.

The September 16 issue contains a design idea by a fellow from a university in Moldova. He shows that the simple addition of a diode to a classic Class-A junction transistor audio stage reduces second harmonic distortion by more than two-thirds in a maximum-gain, open-loop situation. Doubtless none of us would ever consider a gain stage without at least local feedback, but I can't see any reason that the diode trick would not be applicable to designs that utilize "global" feedback, helping to improve overall linearity with a minuscule investment in additional parts.

The article can be accessed at: http://www.reed-electronics.com/ednmag/article/CA450599?pubdate=09%2F16%2F2004&spacedesc=designideas.

Jim Wood
Brea, Calif.

FIGURE 2: STOCK TANG BAND W3-881S, SERIAL NUMBER 206, FREQUENCY RESPONSE GRAPH. THE FREQUENCY RESPONSE IS DRAWN IN BLACK, THE DECAY RESPONSE IS DRAWN IN RED, AND THE ONSET RESPONSE IS DRAWN IN BLUE.

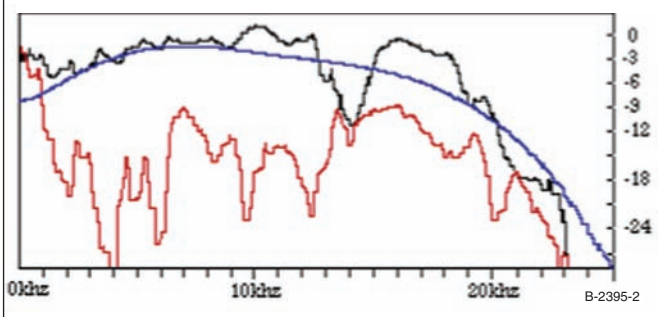


FIGURE 6: STOCK TANG BAND W3-881S ONSET RESPONSE COMPARISON GRAPHS. THE ONSET RESPONSE OF DRIVER SERIAL NUMBER 206 IS DRAWN IN GREEN; THE ONSET RESPONSE OF DRIVER SERIAL NUMBER 495 IS DRAWN IN BLUE.

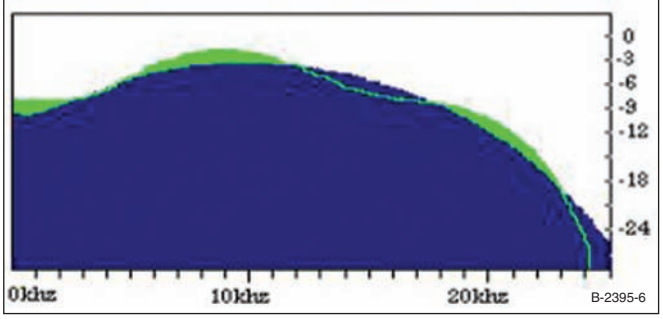


FIGURE 3: STOCK TANG BAND W3-881S, SERIAL NUMBERS 206 AND 495, FREQUENCY RESPONSE COMPARISON GRAPHS. THE FREQUENCY RESPONSE OF DRIVER SERIAL NUMBER 206 IS DRAWN IN GREEN; THE FREQUENCY RESPONSE OF DRIVER SERIAL NUMBER 495 IS DRAWN IN BLUE.

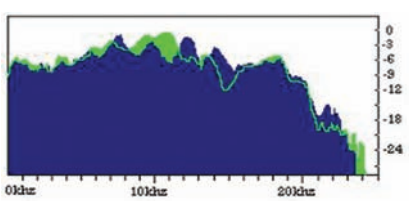
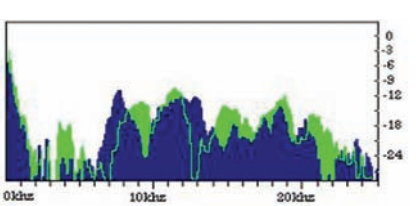


FIGURE 5: STOCK DECAY RESPONSE COMPARISON OF W3-881S DRIVERS SERIAL NUMBER 206 AND 495. SERIAL NUMBER 206 DECAY RESPONSE IS DRAWN IN GREEN, AND SERIAL NUMBER 495 DECAY RESPONSE IS DRAWN IN BLUE.



EDN is an excellent periodical with whom we have always had cordial relationships over the years. Several of the editors are aX *readers.—E.T.D.*

CORRECTION

Several graphs accompanying Mark McKenzie's "Remaking Tang Band's W3-881S (Dec. '04)" were inadvertently placed in the black and white section of the magazine, but make more sense when published in color. The relevant graphs are reprinted here in color. *Xm*

solid state

Back

Passive Preamp: Transformer Implemented

By Jack Elliano

An introduction to amplification with passive devices.

Line amplifiers are most commonly used to increase the level of a CD player or phono preamp into the amplifiers. Line amps do not need to amplify very much to accomplish this. The average overall gain required is about four. Normally the job is done with an active device such as a tube with its conventional means to offer a low impedance output to the power amplifiers with standard interconnects.

A not-so-popular means to increase gain is to use a step-up transformer plus some sort of variable gain control. This usually fails because the output impedance is too high for the interconnect cable capacity and attenuation of high frequencies. However, step-up transformers using a low impedance source signal can offer a usable stepped-up impedance with very little effect from interconnect cable capacity.

PHOTO 1: INTERIOR VIEW FROM FRONT.

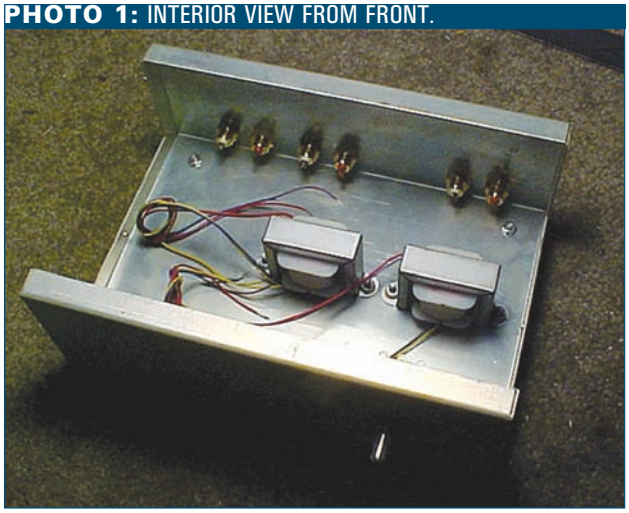
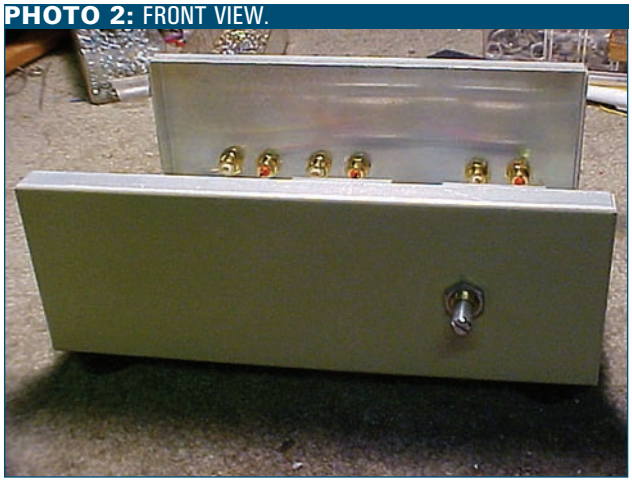


PHOTO 2: FRONT VIEW.



Many newer music source devices show very low output impedance, frequently 50 to 100Ω. These computer-based, hard drive music delivery systems use a pro audio sound card that develops an extremely linear sound and has standard 50Ω balanced and unbalanced outputs.

TRANSFORMER DESIGN

Step-up transformers that will work with op-amp outputs must have specific characteristics to ensure full bandwidth, constant load, and low distortion. Its primary winding must have a low direct current resistance (DCR) and enough inductance to react with the low end. This parameter is necessary because the

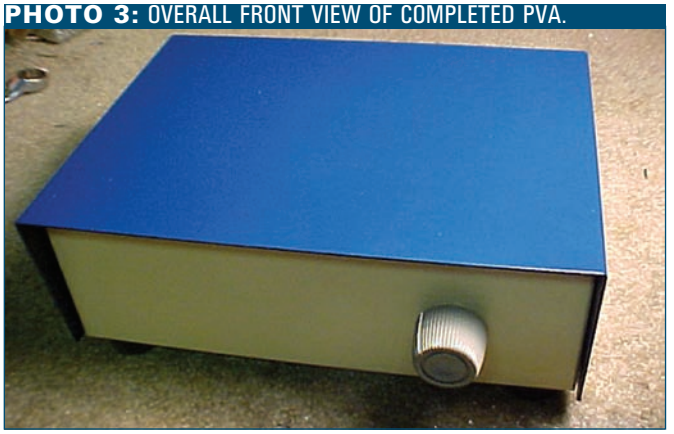
DCR is added to the total impedance and is a loss.

Due to the very low voltage and power at this impedance, you must use a core material that will increase inductance and move fast over zero crossing at low levels. Nickel laminations work very well at these levels and offer very high permeability. This transformer also needs a high coupling, low capacity wind.

We at Electra-Print Audio tried a few arrangements of step-up and settled on a 1 to 8V gain, which offered an impedance of 10kΩ from 150Ω source. This worked well with a 10kΩ high-quality audio taper as a constant load and a means to vary the gain for the amplifiers. We calculated that 80kHz is the –3dB point using a 50pF interconnect from a 10kΩ source. This seemed safe enough. Note that a 400pF interconnect from a 10kΩ source at 20kHz will give a reactance of 19kΩ, so you would measure a rolloff of about –2dB. When reactance is equal to source impedance, this is –3dB.

The transformer we built and used in this circuit had an overall bandwidth –1dB from 6Hz to 55kHz. We achieved bandwidth with 49% nickel core and proper longitudinal balanced windings. The 10k secondary load forced a constant impedance back to the source with which to operate. This offered good operational characteristics.

PHOTO 3: OVERALL FRONT VIEW OF COMPLETED PVA.



WHAT WORKS AND WHAT WON'T

The input impedance of a common tube amplifier is mostly the value of its input tube grid resistor. This impedance does not need to be matched—it is not transferring power. If it is too low (less than 20k), it may lower the total gain of the passive voltage amplifier (PVA) output. The input sensitivity of the amplifier is the most important parameter needed. Sensitivities of 0.25 to 1V work well and the higher input levels may not give full power output. The output impedance of the PVA is the sum of the resulting transformer secondary and resistance used within the volume control.

Some music sources may not work with this PVA design due to their higher output impedances. One remedy for this is to wind a different ratio step-up to offer lower gain levels. The CD players with cathode-follower outputs most likely are about 3kΩ, so the voltage level should be sufficient to drive an attenuator only, to amplifier input. It would benefit the waveform to pass it through a 1:1 ratio transformer and its nickel core.

PHOTO 4: BACK VIEW.

You can custom-wind these type transformers to accommo-

date any ratio. Due to the small size, nickel core, and levels involved, a wide bandwidth is always a result.

MEASURING OUTPUT IMPEDANCE

One means to measure output impedance of a CD player—with close results—is to play a test CD with a 1kHz signal and measure the open unloaded, output signal on a good AC voltmeter. Then add a variable resistance across it, adjust it until the value is half of the open measurement, remove this resistance, and measure its resistance with a standard ohmmeter. That value should be very close to the source impedance, at least at 1kHz.

SIMPLE CIRCUIT DESCRIPTION

As the circuit diagram (*Fig. 1*) shows, the volume control wiper then goes to the output, and, if needed, to subwoofer outputs. The 1k resistor between these outputs serves to isolate any low-pass filter effects from the normal output. The circuit arrangement here ensures that the bandwidth will not change, no matter what volume level you use. It will not add or subtract inductance and capacity as other tapped ratio devices will. The output or source impedance of this device is within 10k or so. It is best to keep the output to the amplifier interconnect capacity, around 50pF.

If component isolation is necessary due to a ground loop, or if a hum develops while interconnecting the unit, you must insulate the input jacks from the case. Remove the ground from the input winding or jack. Grounds on the output and bass output must be common and cannot be isolated from each other.

LISTENING NOTES

The audible result of this device is very interesting and different. One noticeable difference is the increase of amplitude range from low to high levels as well as increased detail. With a nonactive or passive means, amplification has an extremely accurate transfer characteristic result. A tube, as many people experience, will add its signature to the sound. Obviously, with no power supply, this design will have no power supply noise residue despite having gain. Normal listening level ranged from 11 to 2 o'clock on the control, so it has plenty of gain.

The silver secondary version of the transformer sounded the same as the all-copper, save for a slight but notice-

FIGURE 1: CIRCUIT DIAGRAM, ONE CHANNEL SHOWN.

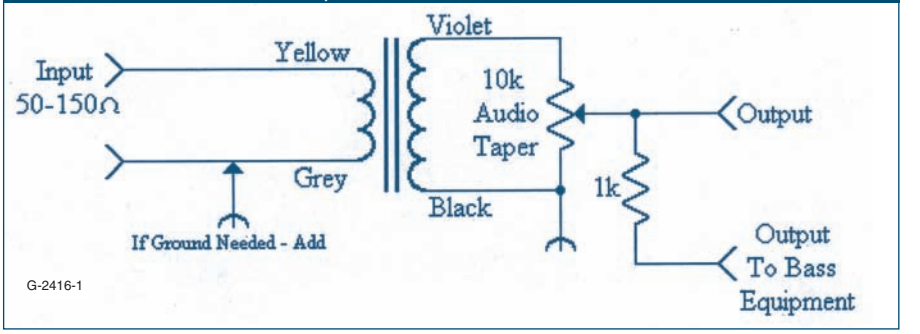


PHOTO 5: INTERIOR TO REAR RCA JACK.

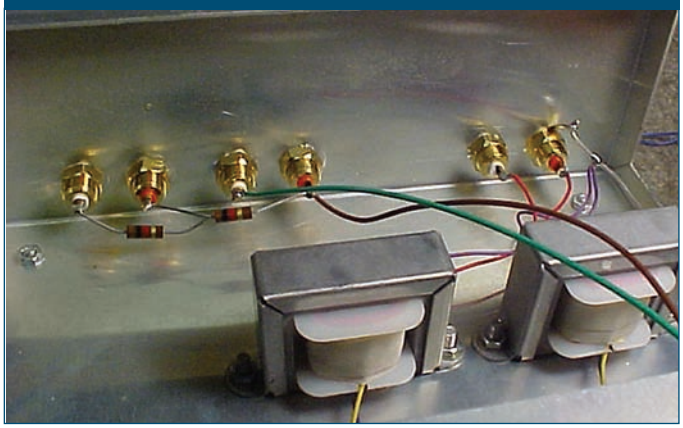
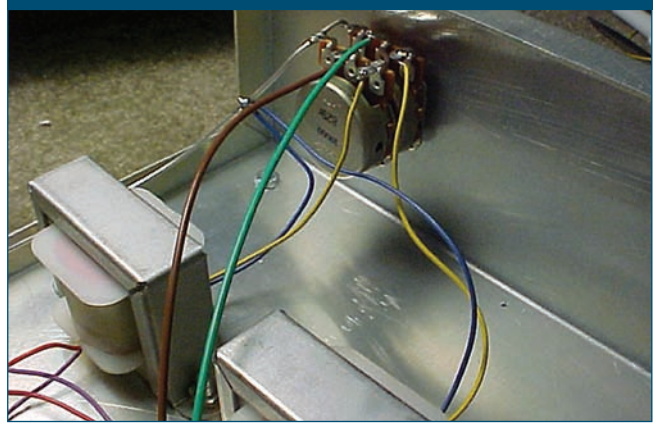


PHOTO 6: VIEW OF VOLUME CONTROL.



able added softness or silky sound. Silver is more sensitive to flux variations than copper. It may reveal a very high number of harmonics.

A nickel core low-level audio transformer seems to have an interesting resulting sound. The first thing you notice is a wide swing in dynamic range—very different from the original CD player op-amp output. The second is a clean, rich delivery of harmonics of strings and brass. Also note that it will bring out a poorly recorded CD. We know little at this point about why these effects are happening, but our theories are piling up.

CONSTRUCTION

You should build the PVA with a steel chassis for best results. Steel will absorb stray magnetic fields from power transformers and chokes in other equipment located nearby. You can add switching to other inputs to the design in order to accommodate other music sources provided source output impedances are between 50 and 150Ω. Do not exceed 3V input or the nickel core will saturate and distortion will occur at low frequencies (patent applied for). *aX*

PARTS LIST

- 2 PVE-3 Electra-Print transformer (nickel laminations with copper wire-$56 each) or PVA-3S Electra-Print transformer (nickel laminations with silver secondary-$250 each)
- 1 10k audio taper volume potentiometer or equivalent
- 1 7″ × 5″ × 2″ steel chassis with bottom plate or equivalent
- 6 RCA jacks
- 2 1k ½W resistors

(Transformers available from Electra-Print Audio Co., 4117 Roxanne Drive, Las Vegas, NV 89108, electaudio @aol.com, 702-396-4909, FAX 702-396-4910.)