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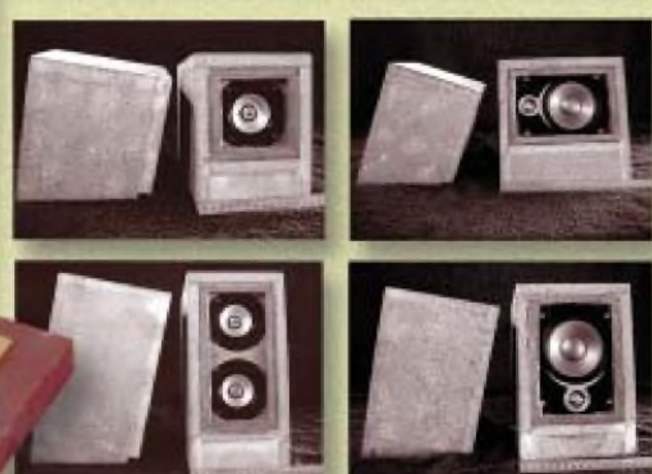

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Behind the Scene

Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2000. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

Dr. D'Appolito designs crossover, specifies cabinet design, and tests prototype drivers for Usher Audio, all from his private lab in Boulder, Colorado. Although consulting to a couple of other companies, Dr. D'Appolito especially enjoys working with Usher Audio and always finds the tremendous value Usher Audio products represent a delightful surprise in today's High End audio world.

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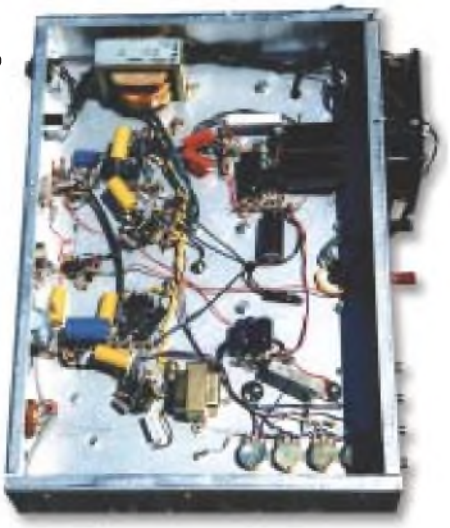

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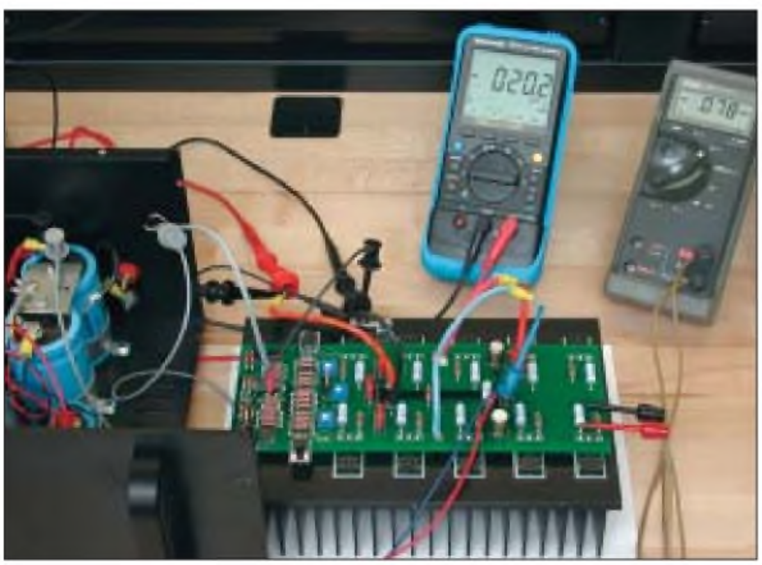

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CODE Articles and letters identified with the following symbols:
solid state tubes speakers

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

JOHN STUART MILL

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Guest Editorial

Detrashing Television

By Darcy E. Staggs

One recent Sunday night I stayed up until 1:30 AM watching TV. What was so interesting about that? It was the picture quality. During this time the electromagnetic interference permeating the 35 air miles between my antenna and the transmitter towers on Mount Wilson is at its lowest ebb, when the receiver does its best.

I saw surprising small details, natural edges, realistic colors of every imaginable hue, and subtleties within the images usually identified with good photographs. For example, reflections on the shiny paint of new cars were clear enough to identify the objects being reflected, and their color. Clothing revealed its woven texture and subtle transitions in brightness depending on its folded curvature. The makeup on the faces of newscasters could be discerned, answering the riddle why flawless complexions always seemed to accompany these images. Amazing, and not the least bit mediocre[1].

A trip to any video retailer reveals the depressing fact that nothing like these images is reproduced on consumer sets. Yet American broadcasters are sending out extremely high-quality transmissions most of the time, on traditional VHF and UHF analog TV. All this has left me both elated and strangely depressed—I can see these high-quality images, but who else can?

Due to the forces of the fundamentals driving our economy and those of other nations[2], nearly all entertainment electronics manufacturing has become the sole province of the Pacific Rim conglomerates. Zenith, for example, maintains engineering expertise in the US but is now owned by one of those conglomerates, all manufacturing done outside of the USA.

Conquering the consumer electronics market depended in part on the Asian success in taking every imaginable component and finding a way to extract all possible cost from it, often with functional quality taking a back seat to price. And so today our video reproduction is seriously compromised by these same budget components, competition being almost exclusively in the area of purchase price. And our outstanding American analog transmissions are viewed on seriously compromised sets. Scandalous.

HDTV will save us from mediocrity, we read. It is time for a major modernization effort in this area, but under the surface the benefit of going entirely digital means fewer analog components, where so much distortion has originated. The increase in the number of video lines in itself is not the biggest source of image quality improvement, although it helps. The manufacturers are being saved from themselves, at massive expense to American broadcasters, who so far have radiated signals anyone would be proud of, if they could only see them.

Just like the near total displacement of vinyl phonograph records by CDs, the shift from stunning but unseen analog TV to reception of digital video will give the sensation of better picture quality to many people. Let's say it has the capacity to add less garbage. I welcome the new technology, but the old one reminds me of the old Russian fable about the man who ran with one leg tied behind his ear to keep from outdistancing his companions.

In the Xpress Mail section of this issue (p. 66), I offer my best visible patch for underperforming electrolytic capacitors in video equipment.

REFERENCES

1. "A DVD Rescue," D. Staggs, *audioXpress*, July '02, p. 24.

2. *Economic and Social History of Medieval Europe*, Henri Pirenne.


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An Affordable Full-Range Speaker Project

The author demonstrates why you don't need to build a complicated enclosure to use full-range drivers. **By Pete Millett**

I've been interested in full-range drivers for a while. I built a small TQWP (tuned quarter-wave pipe) design I found on the Internet, using 3″ full-range drivers, and was impressed with how they sounded. And just for fun, I put some little Radio Shack drivers into ported boxes, and again was amazed at how they sounded. So, I decided it was time for a serious attempt at a full-range speaker (*Photo 1*).

I looked at many designs and quite a few different drivers. The "classic" full-range speaker looked as though it would be a Lowther driver in a big back-horn enclosure. These drivers were a bit out of my price range, though, so I focused on the more affordable drivers available from Fostex.

Most of the enclosure designs I ran across for these drivers were either back horns, or TQWPs. I don't have anything against them (I've heard many that I thought sounded wonderful) but I found the simplicity of a vented box appealing. I'm no speaker expert, and I thought that this would be the easiest configuration for me to understand.

I was happy to find that there are full-range drivers available designed to work in a simple vented box. In fact, the least expensive series of full-range drivers from Fostex are all specified to work this way.

WHY FULL-RANGE?

Full-range speakers aren't for everybody. I'll discuss some of the good and bad points about them, as compared to a conventional two-way (woofer and tweeter) speaker.

First, the advantages:

Since a full-range speaker is, well, full-range, you don't need any kind of crossover network to split the incoming audio to different drivers. In most cases, the output of the amplifier just hooks up to the two speaker terminals—nothing there but wire. This removes all the potential for phase and amplitude anomalies caused by a crossover network. I think that getting a crossover right is probably the hardest part of designing a multi-way speaker.

Since all frequencies originate from the same source, there are no temporal or spatial issues with alignment of multiple drivers. It can be difficult to blend the sound from multiple drivers correctly, since some distance separates them on the baffle of the speaker. This can cause imaging anomalies, and perhaps is the reason why many listeners find the imaging of single-driver speakers superior.

Last, partly because of the lack of any crossover, you can usually build full-range systems for less money than a comparable multi-way speaker.

Now for the disadvantages:

At high frequencies, most full-range drivers are very directional, much more so than a good tweeter. This "beaming" makes for a very definite "sweet spot" to listen from, with the sound becoming progressively duller as you go off-axis.

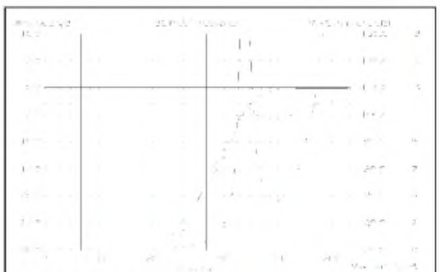

FIGURE 1: Simulated low-frequency response (using FE-164) using Boxplot software. B-2224-1

In order to reproduce high frequencies, you must make some tradeoffs that limit the amount of cone excursion the speaker can accomplish, as well as limiting the maximum cone diameter. The end result is that full-range drivers are somewhat limited in their ability to reproduce very loud, low-frequency bass notes.

For me, the advantages outweigh the disadvantages. I'm not a big fan of bass that shakes the room, and I tend to listen in a room where I can position the speakers—and my chair—exactly where I want them. For these limita-

PHOTO 1: The finished speaker (front view).


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tions, I'm rewarded with a clear, live sound and excellent imaging, for a low price.

THE DESIGN

I searched the Internet for full-range, bass-reflex designs, and didn't find very many. The most interesting was a "recommended enclosure" design published by Fostex for their FE-167E, a 6″ full-range driver. The enclosure was a floor-standing unit, which used part of the "stand" as the port.

I loaded the appropriate data into Boxplot, an excellent (and affordable) speaker-box plotting program, to examine the response of the Fostex box. It seemed too small, and had a fairly high resonant frequency. I think Fostex did this to ensure that the low-frequency power handling wasn't too limited; as noted previously, these speakers don't have a lot of cone travel, and if you design a box with a very low resonance, it would move too far with high-amplitude bass notes.

Retaining the 6″ Fostex driver, I opted for a box volume and tuning much closer to the "optimal" QB3 alignment for the speaker. I ended up with a

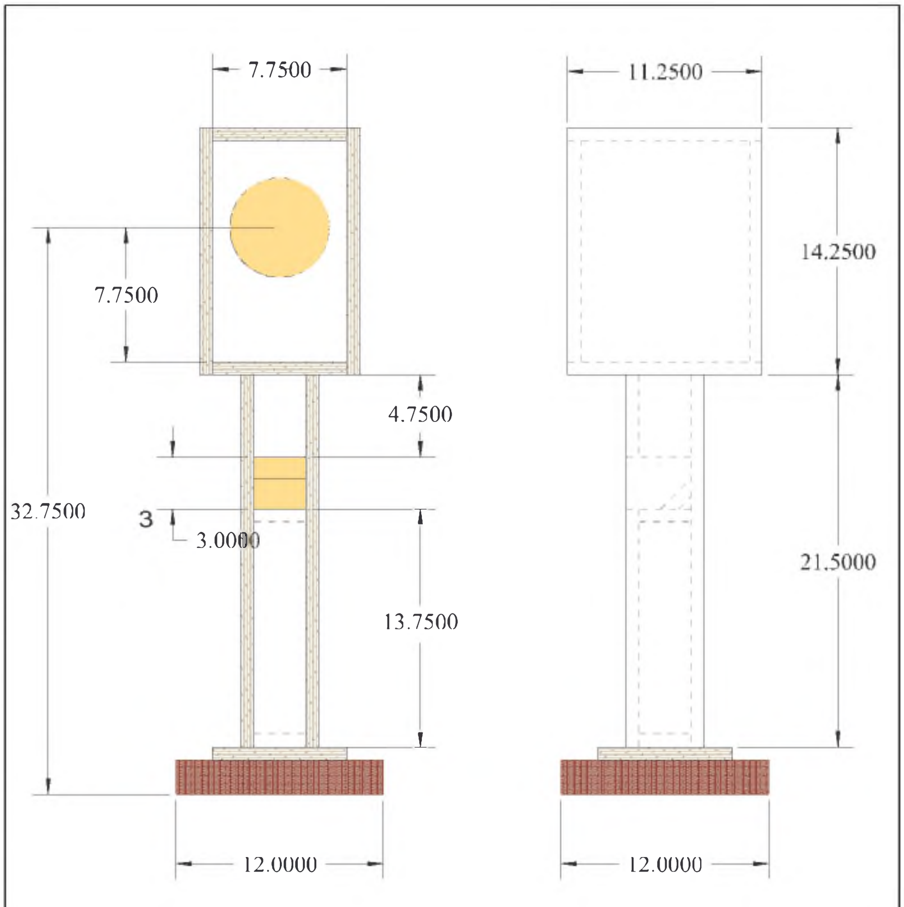

FIGURE 2: The FE-164/FE-167E speaker.

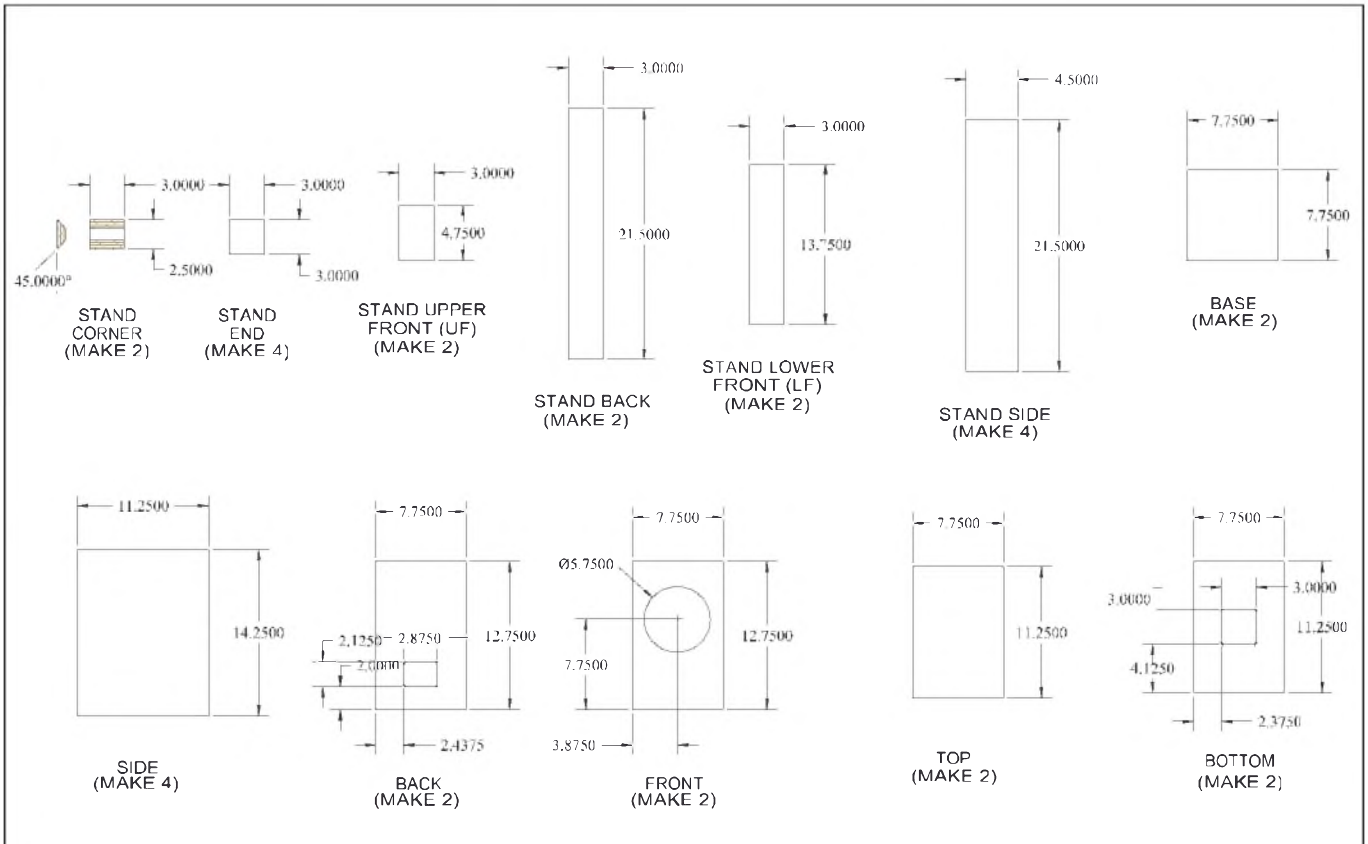

FIGURE 3: Individual parts used to make the speaker.

design that called for $0.52ft^3$ of volume and a box tuning frequency of 65Hz. The Boxplot simulation (*Fig. 1*) showed the bass response of the box to be –3dB at 70Hz.

I used the FE-164 driver instead of the FE-167E. They are very similar, and the FE-164, though recently discontinued by Fostex, was cheaper ($45.25 each at Madisound), and still available. It also had the advantage of a larger X_{MAX}, which allows it (on paper) to handle a little more power in the low bass. I think that you could use either driver in this design with equal success.

I kept the same general design as the Fostex box, using a stand to form the port. I made the stand tall enough to place the centerline of the driver at about the same height above the floor as my ears, sitting in my usual listening chair. I also added a sandstone base to the bottom of the speaker, in place of a plywood square.

Figure 2 shows the speaker assembly and overall dimensions.

CUTTING THE WOOD

I chose Baltic birch plywood for the speaker. You could also build it out of MDF or other material, but I like the looks—and sound—of the birch plywood. It costs a bit more than some other materials, but you need only one sheet. I spent about $50 for a sheet at a local store.

Figure 3 shows the dimensions of all the individual parts that you need to cut, and *Fig. 4* shows how I laid out these pieces on a 5′ × 5′ sheet of ¾″ (actually, 19mm) thick Baltic birch. Note the dotted line down the center of the sheet; I had the store where I bought the plywood cut it down the middle for me. This makes it much, much easier for one person to handle on the table saw.

I laid the parts out so that you cut the sheet the full length into seven parts. Then cross-cut these parts to get the final finished parts. I cross-cut the smaller pieces on a miter saw; I had to cut the bigger ones on the table saw.

I've found it much easier to cut the parts that are on the "outside" of a glue joint about 1⁄16″ larger than needed, and then use my router with a flush-trim bit to remove the excess. This guarantees a nice flush outside joint. This is especial-

PHOTO 2: The cut-out parts.

ly important when using plywood, because if you sand the surface of the wood to make a smooth edge, you might go through the veneer, which messes up the otherwise pretty appearance of the wood.

Photo 2 shows the parts all cut out, ready for the holes. I cut the round hole for the driver using the router and a Jasper circle jig. I used a reciprocating saw to make the rectangular holes for the input connectors and for the port at the bottom of the box, after drilling holes in the corners.

One last detail is to install T-nuts or threaded inserts to attach the driver. I just set the driver into the hole, centered it, and marked the hole locations with a pencil. I then drilled holes on the drill press and installed the fasteners. I used 8-32 threaded inserts, which install into a ¼″ blind hole ½″ deep.

ASSEMBLY

Assembly is a simple process of gluing the pieces together. I began with the stand (*Photos 3* and *4*). First, glue the bottom part together. When it is dry, mate it with the sides and upper front piece. Note that the bottom section is a closed box, while the upper section is open at the top. This upper section is the port for the speaker.

Once the glue has dried on the stand, fill it with sand, which not only helps weigh the bottom of the speaker down to make it more stable, but it also prevents it from becoming a resonant cavity. I drilled a small hole in the bottom of the stand and filled it with "play sand" from a local home improvement store (*Photo 5*). It took a lot of tapping with a hammer to settle the sand, but eventually I got as much in as I could, and sealed it closed by gluing—and screwing—the base onto the end of the stand.

With the stands finished (*Photo 6*), box assembly is next. First, glue the top and bottom (with the hole cut for the port) to the front and back (*Photo 7*). Next, before closing the box, attach it to the open end of the stand using glue and screws (*Photo 8*).

The insides of the boxes are covered in ⅛″ thick wool felt. The material that I used is called "F-13 gray felt," which I purchased from McMaster-Carr, an industrial supply house. Cut the felt to

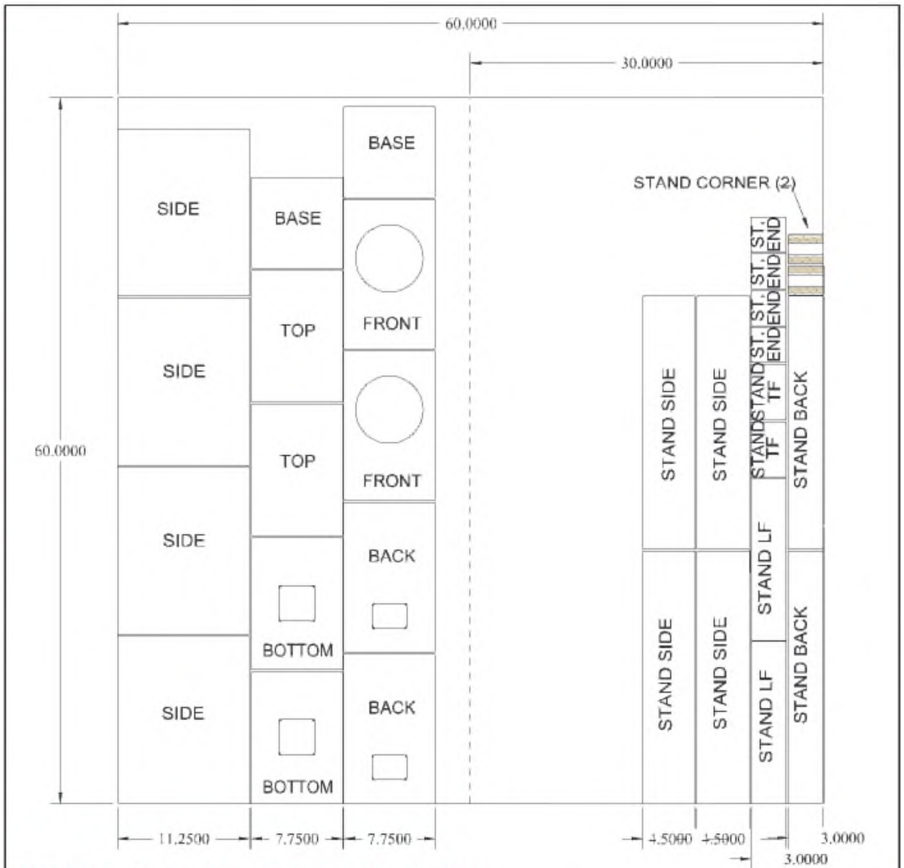

FIGURE 4: Cutting diagram from a 5′ × 5′ sheet.

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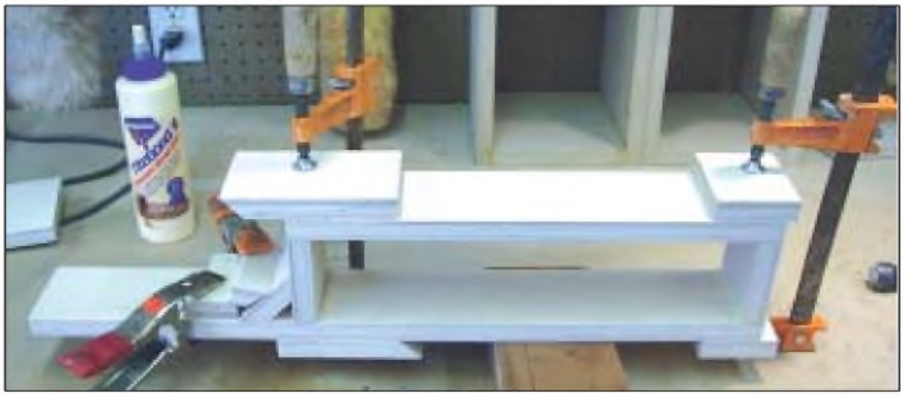

PHOTO 3: Gluing the bottom of the stand.

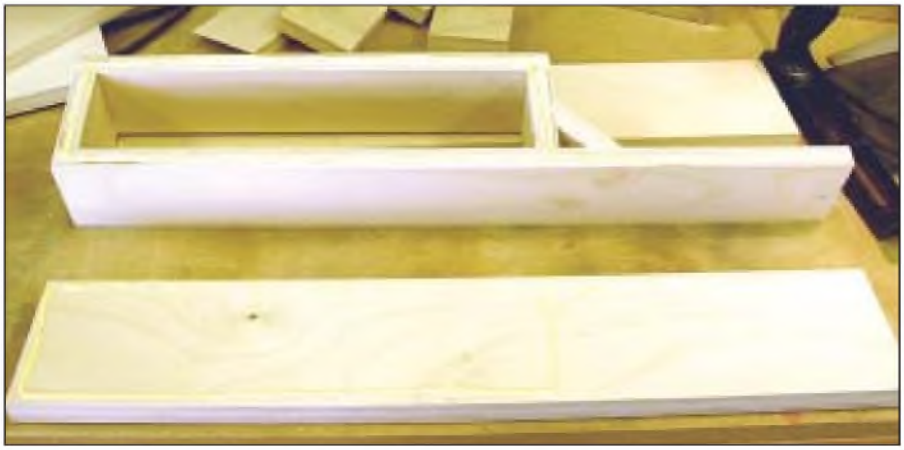

PHOTO 4: Gluing the sides of the stand.

size with a sharp knife to cover all of the inside surfaces of the box, except for the front where the driver mounts (*Photo 9*). I added a small piece below the driver (*Photo 10*), which shows the inside of the box before closing it up. I glued the felt to the inside of the box using "Liquid Nails" adhesive—very sticky, gooey stuff.

After putting felt on the inside of the sides of the box, I glued them in place, closing the box as shown in *Photo 11*. I then used the router with a flush-cut bit to remove overhanging plywood and glue from the corners.

FINISHING TOUCHES

Baltic birch is a little difficult to finish. It doesn't appear to take stain well, and lacquer didn't look nice either. I wound up sanding the plywood with 150 grit, then 220 grit sandpaper, on a random-orbit sander. I then applied clear oil, Watco's Danish oil finish. It gives a smooth, natural-looking finish.

With the woodwork finished, all that remains is to attach the stone base. I used some 2″ thick sandstone, quarried locally, which cost me only $4 per 12″ × 12″ square. I drilled holes through the stone using a masonry bit in my drill press, and attached the stone using long screws (*Photo 12*). The finished base looks quite nice (*Photo 13*).

If you can't obtain stone locally for a reasonable cost, you could also use a concrete stepping stone, or even several stacked layers of plywood. The idea behind the base is to make a platform that's large enough and heavy enough to keep the speaker from tipping over if somebody bumps into it.

INSTALLING THE DRIVER

The Fostex drivers come with a self-adhesive foam gasket. I stuck this gasket to the back of the driver, and soldered on wires, before installing the driver into the hole in the box. It's secured with four screws; I used 8-32 × 3/8″ black-oxide button-head screws, and nylon washers.


PHOTO 5: Filling the bottom of the stand with sand.


PHOTO 6: The completed stands.

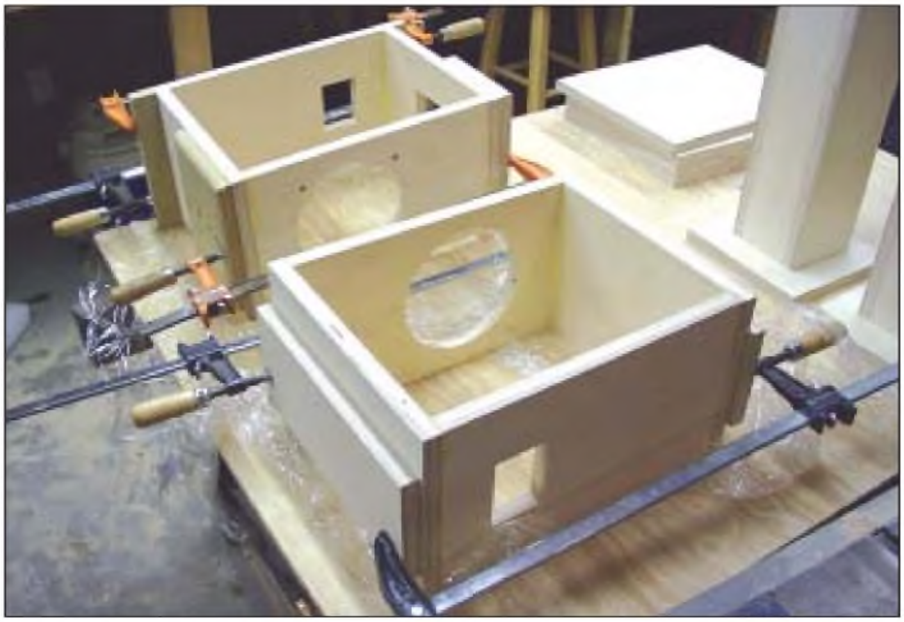
PHOTO 7: Gluing the boxes together.

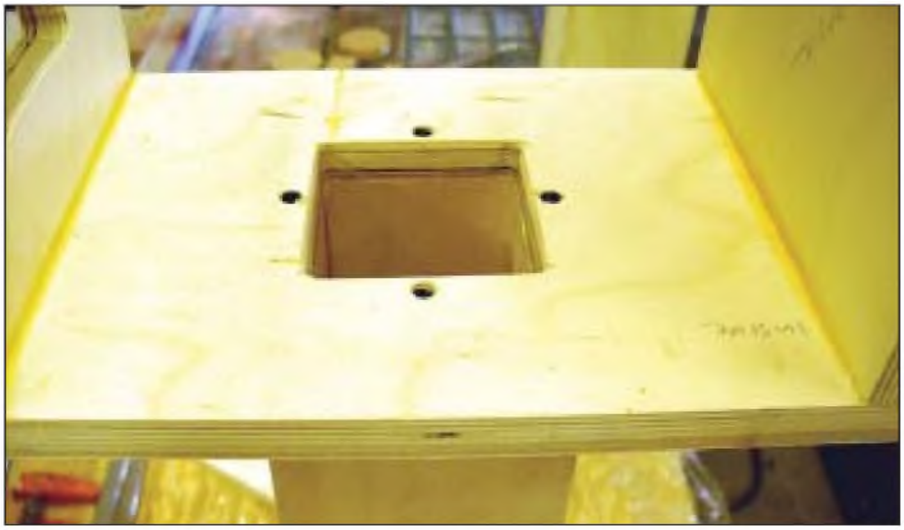
PHOTO 8: Attaching the open box to stand.

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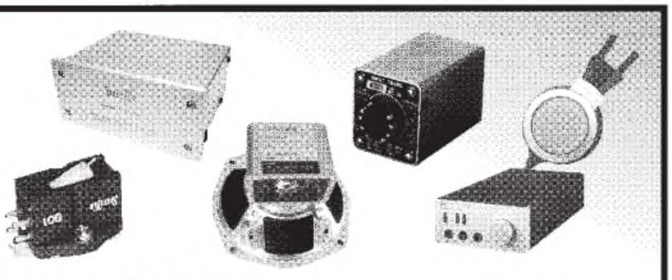

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Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750
Shelter Model 901 (CROWN JEWEL SE)	1,400

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SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
The Remembrance of Sound Post	(Susumu Sakuma)	30
Classic Valve	(Hisashi Ohtsuka)	40
MJ Selected 300B Amps	(MJ)	30
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■MC STEP UP TRANS

Model	Specifications			Price (US$)	Postage**
	Pri.Imp(Ω)	Sec.Imp(kΩ)	Response		
Shelter Model 411	3～15	47	20Hz～50kHz	980	Area I $25
Jensen JE-34K-DX	3	47	20Hz～20kHz	550	Area II $30
Peerless 4722	38	50	20Hz～20kHz	300	Area III $40 Area IV $50

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■STAX

Model	Price(US $)
OMEGA II System(SR-007+SRM-007t)	Ask
SRS-5050 System W MK II	Ask
SRS-4040 Signature System II	Ask
SRS-3030 Classic System II	Ask
SRS-2020 Basic System II	Ask
SR-001 MK2(S-001 MK II+SRM-001)	Ask

■Speaker

Model	Specifications					Price* (US $)	Postage** (US $)			
	D (cm)	Ω	Response	db	w		I	II	III	IV
Fostex FE208Σ	20	8	45Hz～20kHz	96.5	100	296	62	74	120	156
Fostex FE168Σ	16	8	60Hz～20kHz	94	80	236	42	50	73	98

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■TANGO TRANS (ISO) 〔40models are available now〕

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US $)	Postage** (US $) I	II	III	IV	
XE-20S (SE OPT)	20	2.5 , 3.5 , 5	20Hz～90kHz	300B,50,2A3	396	47	56	84	113	Price is for a Pair
U-808 (SE OPT)	25	2 , 2.5 , 3.5, 5	20Hz～65kHz	6L6,50,2A3	242	42	50	73	98	
XE-60-5 (PP OPT)	60	5	4Hz～80kHz	300B,KT-88,EL34	620	62	74	115	156	
FX-40-5 (PP OPT)	40	5	4Hz～80kHz	2A3,EL34,6L6	320	47	56	84	113	
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz～100kHz	300B,50,PX-25	620	62	74	115	156	
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz～50kHz	211,845	620	62	74	115	156	
X-10SF 〔X-10S〕	40	10W/SG Tap	20Hz～55kHz	211,845	1160	90	110	180	251	
NC-14 (Interstage)	—	〔1+1：1+1〕5	25Hz～40kHz	〔30mA〕 6V6 (T)	264	30	40	50	70	
NC-16 (Interstage)	—	〔1+1：2+2〕7	25Hz～20kHz	〔15mA〕 6SN7	264	30	40	50	70	
NC-20F 〔NC-20〕 (Interstage)	—	〔1：1〕5	18Hz～80kHz	〔30mA〕 6V6 (T)	640	42	50	73	98	
NP-126 (Pre Out)	—	20,10	20Hz～30kHz	〔10mA〕 6SN7	264	30	40	50	70	

** Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US $)	I	II	III	IV	
F-7002 (Permalloy)	10	3.5	15Hz～50kHz	300B,50	836	60	70	110	145	Price is for a Pair
F-7003 (Permalloy)	10	5	15Hz～50kHz	300B,50	836	60	70	110	145	
F-2013	40	10	20Hz～50kHz	211,242	786	70	84	133	181	
F-5002 (Amorphous)	8	3	10Hz～100kHz	300B,2A3	1276	65	80	120	160	

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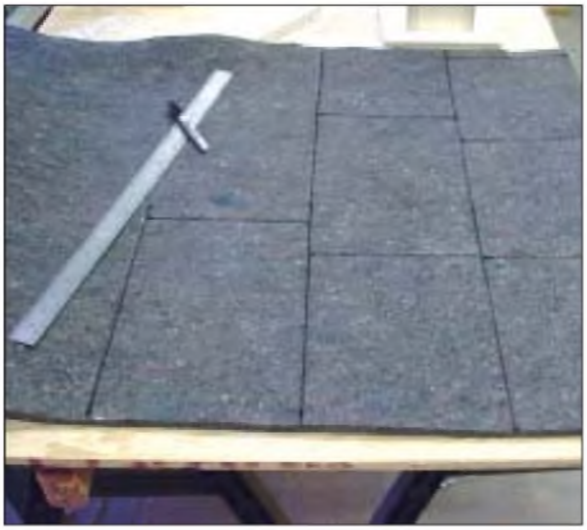

PHOTO 9: Cutting the felt.

PHOTO 10: View of inside the box before closing.

PHOTO 11: Gluing on the box sides.

Pull the wires through the hole in the back of the box, and solder them to an input terminal. You could also use solderless crimp connectors to attach the wires to the speaker and input terminal, if you prefer. I used a rectangular unit with gold-plated binding posts from Parts Express, part number 260-309. The terminal has an attached gasket, and is held in place with four wood screws. And with that, you're ready to hook it up and listen!

MEASUREMENTS AND LISTENING IMPRESSIONS

As with most speakers I've seen, the drivers needed a little time to break in. Unlike others that I've built, these speakers initially seemed a little rolled off on top, with somewhat boomy bass. After a few hours, the highs filled in and the bass calmed down.

I made some unscientific measurements of the speaker in the listening room, using a hand-held Radio Shack sound level meter. In general, the measurements appeared to follow the simulation and the driver's frequency response graph pretty well. If anything, the lowest frequencies were stronger than the simulation, probably because of room effects; I measured about –10dB at 45Hz. The overall frequency response was pretty flat from 60Hz up to almost 20kHz.

I was somewhat concerned that the limited cone motion of the speaker would lead to distorted bass. I ran a 60Hz sine wave through the system and raised the volume until I could hear distortion. This occurred at about 95dB at 1m—right in line with the simulation. I repeated this at 1kHz, and got to over 105dB, where I reached the clipping level of my amplifier.

PHOTO 12: Attaching the sandstone base.

With music, at what I consider pretty loud listening levels, there appeared to be no problem with the speaker hitting its limits. Even fairly bass-heavy electronic music sounded fine, with a lot more bass than I ever expected to get from these speakers.

I'm currently driving

PHOTO 13: Close-up of the sandstone base.

the speakers with 20W SE amplifiers, using 813 tubes connected as triodes. They also perform quite well with my 7W push-pull 6B4G amplifiers. I think that at the specified sensitivity of 92dB @ 1W (1m), you can certainly do with less power, unless you like to listen very loudly.

These speakers have the same "live" character as other good full-range speakers I have listened to. Acoustic instruments—piano and acoustic guitar—are outstanding, like they're in the room with you. The soundstage is wonderful, extending well beyond the speaker locations.

All in all, I deem this project a success—maybe the best speakers you could build for under $150! ❖

PHOTO 14: Close-up of the driver.

PHOTO 15: Finished speaker (quarter view).

CONTACTS

McMaster-Carr Supply Company
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Fostex drivers, supplies

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Parts, supplies

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Newton, NJ 07860
www.diamondcut.com/boxplot.htm
Boxplot software

60W Triode/100W UL Control Amplifier

If you're a hi-fi enthusiast looking for a great-sounding, high power amp for a reasonable cost, your search is over. **By Joseph Norwood Still**

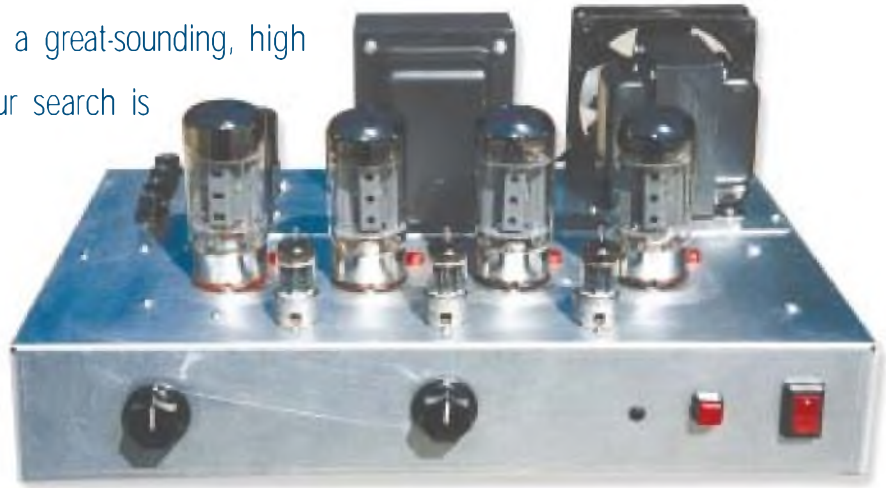

PHOTO 1: Front view of completed amp.

This amplifier is a completely self-contained mono-amp, offering five inputs to accommodate a CD player, tuner, tape, satellite audio, or other audio source (*Photo 1*). It uses push-pull, parallel 6550s to provide 60W output in triode mode or 100W output in ultralinear mode.

This amplifier features high-perveance 5687s in the line amplifier, driver, and inverter stages. The high perveance cascode (SRPP) stages and long-tailed inverter provide circuits having a wide frequency/fast time response. It is very stable and provides low distortion throughout the audio spectrum to its full output. You can build it for a reasonable amount of work at a cost of about $350.

Important: I recommend that you configure this amplifier to operate in a dedicated triode or ultralinear mode. Wire it both ways and select the mode that suits your listening preference. I recommend you first try the triode mode, which is the simplest to wire. Some reviewers claim vocals and jazz groups sound better in the triode mode, and large bands and classical music sound better in the ultralinear mode. The choice is yours!

60W TRIODE OUTPUT STAGE

The design features push-pull parallel 6550s connected as triodes, preceded by a 5687 duo-triode long-tailed phase inverter, a cascode 5687 driver, and a 5687 cascode line amplifier (*Fig. 1*). The advantages of triodes in the power output stage are improved frequency response, improved damping, and reduced distortion.

The amplifier requires 0.34V (RMS) drive signal at the input of the line amplifier and 3.0V (RMS) at the input of the first stage amplifier for 60W output. A review of the tube data in *Table 1* reveals that the dissipation level is within the recommended dissipation ratings for these tubes. The 60W triode operation is obtained with a bias on the 6550s of –60V DC (approximately), thus providing a 42V (RMS) drive signal to the control grids of the 6550s. **Note:** Clipping occurred at the 68W level.

The distortion at 60W/1kHz is 0.21%, and the frequency response at this power level is flat from 20Hz to 20kHz. At 10W (normal room listening level) the frequency response is flat from 20Hz to 40kHz. The distortion at 60W/1kHz, no feedback applied, is 0.7%. The noise of the amplifier is 2.2mV with input open and volume control fully advanced.

Note: When you operate the unit in the triode mode, you must connect the 270Ω metal oxide resistor from the screen grid to the plate, not to the transformer side of the 50Ω, 4W metal oxide resistor. Also, the 47Ω, 2W metal oxide resistors (R35–R38) are not used in the triode mode.

100W ULTRALINEAR OUTPUT STAGE

The circuit diagram of the amplifier (*Fig. 1*) features Quad matched, push-pull, parallel 6550s connected in an ultralinear configuration, preceded by a 5687 duo-triode long-tailed phase inverter, a cascode 5687 driver, and a 5687 cascode line amplifier. The ultralinear operation shares some of the best features of a triode, which is primarily improved damping, extended frequency response, and reduced distortion. It also has greater efficiency than a triode, nearly matching the efficiency of a pentode and providing five times the damping of a pentode.

The amplifier requires 0.38V (RMS) drive signal at the input of the line amplifier and 3.3V (RMS) at the input of the first stage amplifier for 100W outputs. A review of the tube data in *Table 1* reveals that the maximum signal dissipation level is within the recommended dissipation ratings for these tubes. The 100W ultralinear operation was obtained with a bias on the 6550s of –60V

ABOUT THE AUTHOR

Joseph Norwood Still, retired from the electronics industry, is designing affordable high-quality audio amplifiers for the dedicated audiophile. He thoroughly enjoys this hobby and is especially rewarded with many pleasant interchanges with dedicated, resourceful audiophiles. He lives in Bel Air, Md.


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DC (approximately), thus providing a 42V (RMS) drive signal to the control grids of the 6550s. Clipping occurred above the 100W level.

The distortion at 100W/1kHz is 0.46%, and the frequency response at this power level is flat from 30Hz to 20kHz. At 10W (normal room listening level) the frequency response is flat from 20Hz to 30kHz. The distortion at 100W/1kHz, no feedback applied, is 1.9%. The noise of the amplifier is 2.2mV, with input open and volume control fully advanced.

Note: When you operate the unit in the ultralinear mode, you must disconnect the 270Ω, 2W metal oxide resistor from the plate of the 6550 and connect a 47Ω, 2W resistor (soldered) to the free end of the 270Ω resistor. Repeat this process for the other side of the paralleled 6550. Then tie the two free ends of the 47Ω resistors to the UL screen tap of the output transformer, and the resistors and the screen tap to a terminal strip located midway between the two 6550s. This process will ensure stability of the amplifier when operating in the ultralinear mode.

LINE AMPLIFIER, DRIVER, AND INVERTER STAGE

The line amplifier amplifies the incoming signal via high perveance, medium-mu 5687 dual triodes connected in a cascode (SRPP) configuration. This arrangement guarantees a fast time/wide frequency response circuit because of the low plate impedance provided by the upper triode. The high perveance characteristics of the tube provide a high cathode bias and considerable headroom at the input and output of the tube and makes certain no clipping will occur during reproduction of music peaks. These same characteristics apply to the first stage driver.

The distortion of the line amplifier is 0.17% at 2.5V RMS output, and the frequency response is flat from 20 to 40kHz. No feedback is used in this stage, other than the self-generated feedback of the un-bypassed cathode resistor. Using no feedback in the line stage provides a lively, very realistic reproduction of the music presented to the first stage (driver) of the amplifier.

The type 5687 is used for the first stage of the amplifier (*Fig. 1*). The distortion of the cascode (SRPP) 5687 with 6.0V (RMS) output signal (no-feedback) is less than 0.5%, and the frequency response is flat from 20Hz to 40kHz.

Of interest: Reviewing the tube manual and the characteristics of 5687 and 6SN7, you note that the 5687 possesses approximately the same characteristics as a paralleled 6SN7. It is important never to exceed the design maximum 330 plate voltage of the 5687 (I recommend maximum voltage operation of 300V DC). I originally designed the amplifier using 6H30 tubes but discovered there was only a small difference in the measured test results between the 6H30 and 5687, favoring the 6H30. The high cost of the 6H30 versus the $3.80 cost of the 5687 makes the 5687 the logical choice.

Also of interest, it appears paralleling the 5687 would approximate the characteristics of the 6H30. The tube manual electrical characteristics of the 6SN7 and 5687 follow (the 6H30 mu, rp, and gm is calculated):

	6SN7	5687	6H30
Maximum plate voltage	450	330	300
Plate voltage	250	250	180
Grid voltage	–8.0	–12.5	–17
Plate current (mA)	9.0	12	14
Amplification factor	20	16	15
Plate resistance	7700	3000	1500
Transconductance	2600	5400	10000

Using the 5687 long-tailed phase inverter preceded by only a driver stage offered reduced phase shift and a stable feedback loop. Thus, only two stages are located within the feedback loop of the amplifier. The long-tailed phase inverter has unequal output voltages due to the slightly higher gain of the grid-driven section.

Balance of these output voltages is obtained by using a 33kΩ plate resistor on the grid-driven side and a 39kΩ plate resistor on the cathode-driven side. The balance was maintained with five different 5687s, so this should provide a permanent fix. An anti-ringing R/C network (C4 and R14) connects to the signal grid of the inverter. The distortion of the inverter stage, measured on either side (no external feedback), is less than 0.5% from 30Hz to 20kHz at 50V

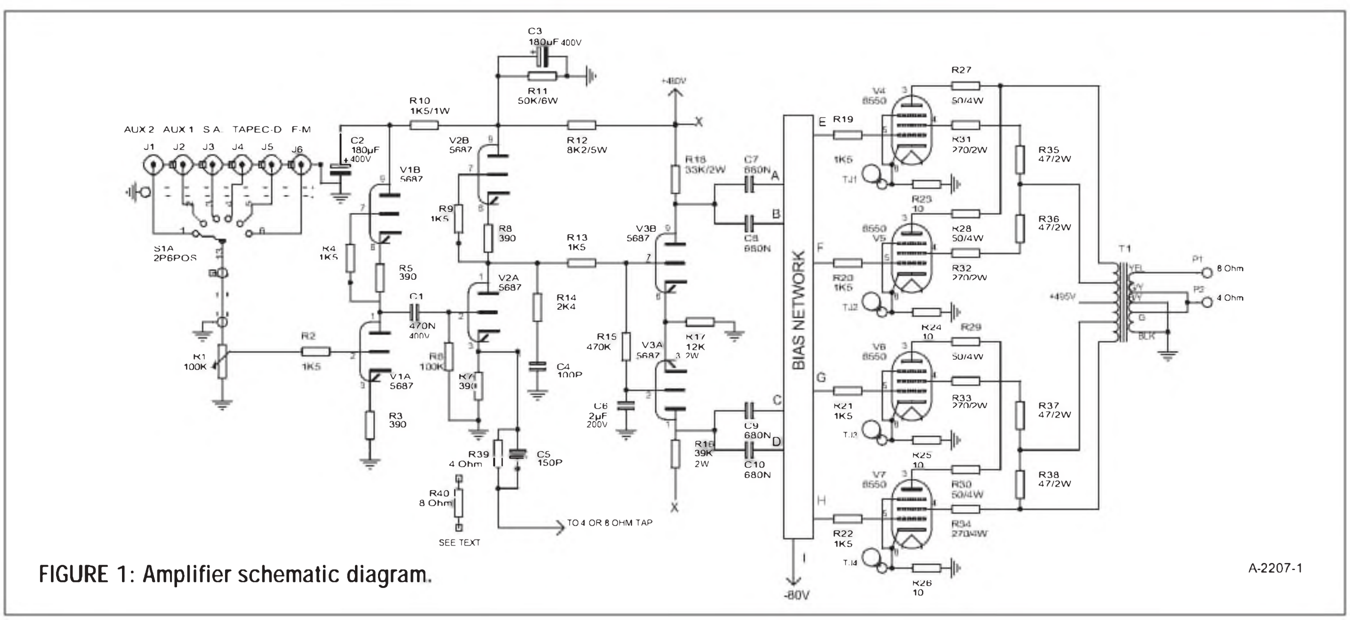

FIGURE 1: Amplifier schematic diagram.

(RMS) output (measurements made with the inverter not connected to the grids of the 6550 output tubes).

Allowing for the push-pull effect and resulting second harmonic cancellation, you can assume the total harmonic distortion for the inverter to be approximately 0.3%. The frequency response of the line amplifier, driver stage, and inverter is flat from 20Hz to 50kHz. The 42V (RMS) output of the inverter will drive the push-pull, parallel 6550s to their full-rated output of 60W triode mode or 100W ultralinear mode.

BIAS NETWORK

The bias network (*Fig. 2*) consists of a transformer (T3), bridge-rectifier (D5–D8), potentiometers (R8–R11), capacitors (C9–C12), and resistors (R12–R19). The center-tapped winding of 12V AC transformer (T3) connects to the 6V AC winding of transformer (T2) via a 5Ω, 5W wirewound resistor (R7). The 72V AC output from transformer (T3) is rectified by bridge-rectifier (D5–D8) and filtered by 1000µF capacitor (C8). The –80V DC output is tied to the top-end of 10kΩ linear potentiometers, and the bottom ends of these potentiometers are returned to ground via 15kΩ resistors.

The 15kΩ resistors and 10kΩ resistors are near the bias supply rectifier and transformer. The 2.0µF capacitors (C9–C12) and 51kΩ grid resistors (R12–R15) are near pin 5 (grids) of the 6550s. In normal operation, the potentiometers are adjusted for a typical output of –58 to –62V DC—for correct operating bias of the 6550s or, more precisely, a 70mA operating current of each 6550.

Note: To increase the operating life of the 6550s, you may bias them for 60mA operation. Operating the tubes at 60mA will cause a slight increase in the distortion and a slight decrease in the power output of the amplifier.

Caution: Before applying plate voltage to the amplifier, make certain the grid circuit is wired properly and the required –65V DC is present on the control grids of all four 6550s.

TABLE 1
TUBE DATA

6550 CLASS AB1 (PUSH-PULL, PARALLEL)	TRIODE MODE	ULTRALINEAR MODE
Plate voltage	495/475V	495/475V
Grid voltage (nominal—see text)	–63V	–63V
Grid signal voltage, RMS	44V	44V
Zero signal plate current	280mA	280mA
Maximum signal plate current	380mA	450mA
Load resistance (plate to plate)	1.9kΩ	1.9kΩ
Power output	60W	100W
Plate dissipation (per tube) no signal	34.6W	34.6W
Plate dissipation (per tube) maximum signal	30W	28.5W
Plate input power	180W	214W
Power ratio efficiency	33%	47%
Harmonic distortion at 1kHz, no feedback	0.7%	1.9%

V1 5687 Eb1 224V, Eb2 112V, Ek1 4.1V (measured across 390Ω resistor), Ek2 4.1V
V2 5687 Eb1 240V, Eb2 120V, Ek1 4.3V (measured across 390Ω resistor), Ek2 4.3V
V3 5687 Eb1 304V, Eb2 300V, Ek1 and Ek2 130 (measured across 12kΩ resistor)

OUTPUT STAGE (GENERAL INFORMATION)

The 6550 (Quad matched) push-pull, parallel output tubes operate in class AB1 in 60W triode mode or 100W ultralinear mode, under conditions shown in *Table 1*. The plate current of the 6550s is measured across 10Ω resistors in the legs of each cathode at test jacks TJ1 to TJ4 (*Fig. 1*). A 0.7V indication corresponds to a 70mA current reading.

The carbon resistors connected to the control grids and the metal oxide resistors connected to the screen grids and plates of the 6550s are required to prevent oscillation of the output tubes. The plate resistors are paralleled 2W metal oxide resistors. (These resistors cannot be the wirewound type. They must be the large physical-size types that are available either from Radio Shack dot Com or NTE Electronics. Most distributors stock 2W resistors that are small in size and have a 250V rating; the larger resistors are rated for 500V operation.)

To ensure stable operation of the amplifier, you must connect the inverse feedback loop to the output taps in use. For example, when using the 4Ω tap, connect resistor R39 (2400Ω) and capacitor C5 (150pF) to the 4Ω tap. When using the 8Ω tap, connect resistor R40 (4700Ω) and capacitor C5 (150pF) to the 8Ω tap.

If the amplifier requires the use of either 4 or 8Ω outputs, incorporate a switch in the amplifier. If the amplifier is used with only 4 or 8Ω output, you can omit the switch. You must connect the secondary windings of the Hammond output transformer in a series configuration as shown in *Fig. 1*. If oscillation occurs when the amplifier is "turned on," reverse the primary plate leads of the output transformer.

AMPLIFIER POWER SUPPLY

The schematic diagram of the power supply is shown in *Fig. 2*. The Hammond plate supply transformer (T1) has a secondary winding rated at 200V AC and 870mA and is available from Antique Electronics Supply for $40. The transformer is used in a full-wave bridge, voltage-doubler circuit; thus its output current capability is reduced to 435mA. This transformer, although small in size, runs cooler than the conventional Hammond transformer that is rated for full-wave operation at 800V AC C.T./465mA.

A 120V AC, 32 or 54 cfm fan is used to cool the transformer, and my experience with plate transformers and their high heat levels suggests that all these transformers should have a fan to dissipate the heat. The fan should be located directly in front of the transformer. Radio Shack has a 1″ wide, 32 cfm fan for $20, and All Electronics has a 1.5″ wide, 54 cfm fan for $14.

The bias supply (–80V DC) uses transformer T3 and a full-wave bridge rectifier. The potentiometers R8–R11 are adjusted to set the bias to the 6550s. Capacitor C8 provides final filtering, limiting the AC ripple to 0.016V AC. The filament transformer (T2) provides the 6.3V AC required to operate the four 6550 and three 5687 heaters.

The high voltage winding of the transformer T1 "feeds" a full-wave, voltage-doubler rectifier (D1–D4). The DC output of the supply is 495V (no load), and 475V DC with the amplifier producing 60W-T/100W-UL output. Resistors R3–R6 are bleeders primarily for protection of the electrolytic capacitors. Capacitor C3 eliminates AC hash.

Capacitors C1 and C2 function as filters and voltage-doubler devices, and capacitors C4–C7 are the final filters of the supply. Resistors R1 and R2 provide isolation between the filter capacitors of the supply. The AC ripple of the 495V DC output of the power supply is 1.4V AC.

AMPLIFIER PERFORMANCE CRITERIA

The performance data of the amplifier is presented at its maximum operating levels. At 1W, from 30Hz to 15kHz, the distortion is less than 0.17% in the triode and ultralinear modes. The harmonic distortion data is presented for 10, 20, 30, 40, 50, and 60W, triode mode

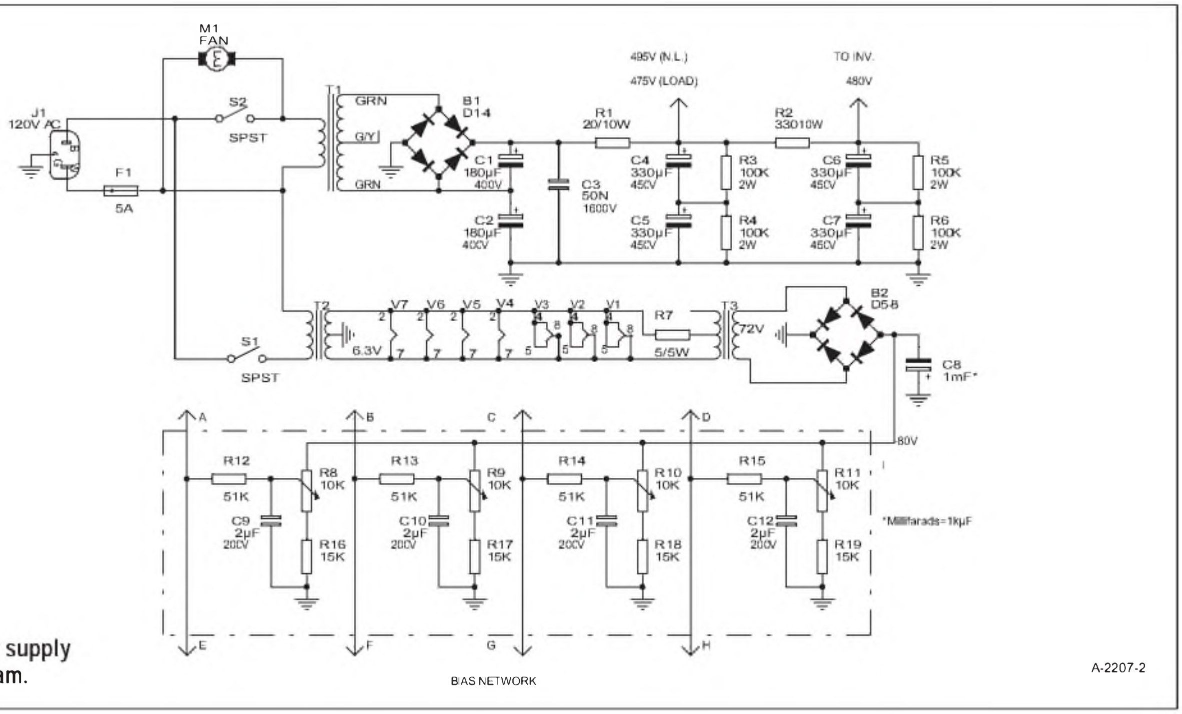

FIGURE 2: Power supply schematic diagram.

(*Table 2*), or for 10, 20, 40, 60, 80, and 100W, ultralinear mode (*Table 3*). I made the amplifier distortion measurements using the 4Ω and 8Ω output taps, but show only measurements for the 8Ω taps, because the 4Ω tap measurements mimic the 8Ω readings.

The square-wave oscillograms are shown for the 60W-T/100W-UL outputs of the amplifier (*Fig. 3*). The 100Hz and 10kHz square waves exhibit slight low- and high-frequency rolloff, while the 1kHz square wave is reproduced almost perfectly, and no peaking is evident in any of the square waves. No high-frequency ringing occurred with the amplifier reproducing the 10kHz square wave, even with a 0.1μF capacitor shunted across the transformer secondary—indicating good high-frequency stability. (The amplifier is also stable at low frequencies.)

The 100Hz and 10kHz square-wave measurements are made with the square-wave signals "fed" to the input of the line amplifier. The square waves of the 60W triode and 100W ultralinear amplifier are virtually identical.

Of interest: To obtain "textbook" quality nearly perfect square waves, the amplifier output stage should have low impedance. See author's "100W Triode Amplifier," *Glass Audio* 3/00 and Michael Burrows' "6B4G Stereo 90 Power Amp," *audioXpress*, August 2002.

The frequency response of the triode mode amplifier at 60W output is flat from 20Hz to 20kHz; at 10W it is flat from 20Hz to 40kHz. The negative feedback of the triode amplifier is 18dB.

The frequency response of the ultralinear amplifier at 100W output is flat from 30Hz to 20kHz; at 10W it is flat from 20Hz to 40kHz. The negative feedback of the ultralinear amplifier is 11dB. The noise of the amplifier in the triode or ultralinear modes, measured at the 4 or 8Ω taps with the input open and the volume control fully-advanced, is 2.2mV. In conclusion, the performance specs speak for themselves. This is an amplifier that you can build for moderate cost with few construction problems.

HUM (NOISE) PROBLEMS

The amplifier is free of 60Hz hum and noise problems, and when connected to a loudspeaker absolute quiet is assured. I recommend that during the initial tryout you have only the amplifier and speaker connected. After you determine that the system is quiet, connect

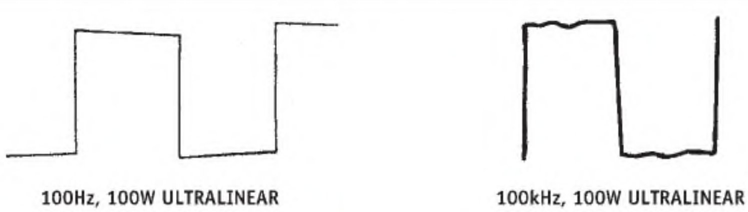

FIGURE 3: Square-wave oscillograms.

TABLE 2
TRIODE MODE TOTAL HARMONIC DISTORTION

OUTPUT FREQUENCY	10W	20W	30W	40W	50W	60W
30Hz	0.15	0.16	0.18	0.2	0.22	0.54
100Hz	0.10	0.11	0.12	0.12	0.15	0.32
1.0kHz	0.08	0.09	0.10	0.10	0.14	0.21
10kHz	0.17	0.30	0.44	0.54	0.78	0.95
15kHz	0.24	0.46	0.66	0.86	1.2	1.4

TABLE 3
ULTRALINEAR MODE TOTAL HARMONIC DISTORTION %

OUTPUT FREQUENCY	10W	20W	40W	60W	80W	100W
30Hz	0.34	0.36	0.44	0.62	0.82	1.0
100Hz	0.15	0.17	0.26	0.36	0.48	0.54
1.0kHz	0.06	0.10	0.16	0.28	0.4	0.46
10kHz	0.08	0.12	0.28	0.36	0.5	0.6
15kHz	0.10	0.14	0.38	0.46	0.6	0.7

the CD player to the amplifier. If it is still quiet, connect the FM tuner to the system. Repeat this process of adding additional items.

If you add a phono preamplifier to the system, make sure the ground wire from the phono motor is connected to a ground post near the input of the phono preamplifier. Make sure all stereo audio input cables are standard shielded types. Also, make sure that all audio runs within the amplifier over 3″ use microphone shielded cable, grounded at the input end. Do not use professional low-impedance (600Ω) studio shielded audio cable.

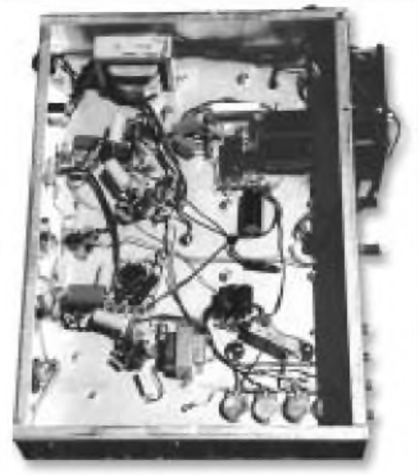

PHOTO 2: Inside the unit.

Note: If you are currently using a solid-state amplifier, be aware it has a low-impedance input and is much less prone to hum pickup than a high-impedance input vacuum tube amplifier. So, be prepared to eradicate any ground loops you may have in your system.

CONSTRUCTION TIPS

"Point-to-point" wiring is used in the amplifier (*Photo 2*). I do not recommend a common ground bus with all components tied to this bus. Instead, use an aluminum chassis to simplify the chassis ground connection process; also use single ground lug terminals in addition to the center post of the five-post terminal strip for all grounds. (When using a painted metal chassis, it is necessary to scrape the chassis to bare metal to make a ground connection.)

Most components are "tied" to the five-pin terminal strips, the center terminal is ground, and components requiring a ground are tied to this ground post. You must use a shielded microphone cable, grounded at the input end, for all audio signal leads of 3″ or more. You may cut the primary and secondary windings of the transformer to shorten the plate leads and speaker leads.

The power and audio output transformers are secured with ¼″ fasteners or similar large fasteners. You should move the two electrolytic capacitors mounted above the chassis to below the chassis; use a five-pin terminal strip to secure the capacitors. The bias supply and the heater supply require the use of one terminal, each. Three terminal strips are required for the high voltage rectifier circuit. One terminal strip is required at the 5687 inverter, and another is located near the 5687 line amplifier. A terminal strip is located near the output transformer to accommodate the secondary leads.

The heaters require no. 18 AWG wire, which is the size used throughout the amplifier. Each side of the paralleled 6550 heaters has its heater wires returned to the 6.3V output winding of the transformer. A short ground path is provided for the ringing, grid, and cathode circuits of the 5687.

On the bottom of the amplifier chassis use large mounting "feet" to give good under-chassis "air-flow." You can see in the photos of the amplifier a 32 cfm fan located directly behind the power transformer to cool the trans-

AMPLIFIER PARTS LIST

THE FOLLOWING PARTS ARE AVAILABLE FROM RADIO SHACK:		
J1–J5	Female, phono jack, gold, chassis mount	274-852
P1, P2	Nylon binding posts	274-662
R1	100k, audio taper, ½W	271-1722
S1	Rotary switch, double section. 6 position	275-1386
THE FOLLOWING PARTS ARE AVAILABLE FROM ANTIQUE ELECTRONICS SUPPLY, 6221 SOUTH MAPLE AVE., TEMPE, AZ 85283, 602-820-5411:		
C1	0.47µF, 400V, polypropylene	C-RD47-400
C4	100pF, 500V, Silver mica	C-SM100
C5	150pF, 500V, Silver mica	C-SM150
C7–C10	0.68µF, 600V	C-UD68-630
R2, R4, R9, R10, R13, R19–R22	1.5k, 1W, C.F. (carbon film)	RB-1.5K
R3, R5, R7, R8	390Ω, 1W, C.F.	RB-390
R6	100k, 1W, C.F.	RB-100K
R14	2.4k, 1W, C.F.	RB-2.4K
R15	470k, 1W, C.F.	RB-470K
R39	2.4k, 1W, C.F.	RB-2.4K (only used with 4Ω speaker)
R40	4.7k, 1W, C.F.	RB-4.7K (only used with 8Ω speaker)
T1	Hammond, 120W, 1900Ω Ct., Sec. 4, 8, 16Ω	P-T1650T
V1, V2, V3	5687 (NOS)	
V1–V3S	9 pin, subminiature tube socket with shield	P-ST9-213MX
V4S–V7S	Octal socket	P-ST8-209MIP
TJ1–TJ5	Insulated tip jack (4 red, 1 black)	S-H260/S-H261
THE FOLLOWING IS AVAILABLE FROM PART EXPRESS, 800-338-0531:		
V4–V7	6550, Sovtek (matched Quad)	
THE FOLLOWING IS AVAILABLE FROM ALL ELECTRONICS, 800-826-5433:		
C6	2.0µF, 200V, metallized polyester	RMC-220
C2, C3	180µF. 400V, 180/400VR	
THE FOLLOWING IS AVAILABLE FROM RADIO SHACK DOT COM:		
R11	50k, 6W, M.O. (parallel three 150k, 2W resistors)	900-0870
R12	8.2k, 5W, wirewound	900-0990
R16	39k, 2W, M.O. (metal-oxide)	900-0856
R17	12k, 2W, M.O.	900-0844
R18	33k, 2W, M.O	900-0854
R23–R26	10Ω. 1W	900-0642
R27–R30	50Ω. 4W (two 100Ω, 2W paralleled resistors)	900-0794
R31–R34	270Ω, 2W	900-0804
R35–R38	47Ω, 2W (Note: only used in UL mode)	900-0786
Cable	shielded, 6-conductor, Belden 9536, (6′)	910-1605
Cable	shielded, microphone, Belden 9394, (6′)	910-1650

WARNING

Lethal voltages are present; exercise extreme caution when constructing and testing the amplifier and never leave the amplifier upside down when children are present.

TEST EQUIPMENT USED

Distortion Analyzer, H-P 331 A
Sine/Square Audio Generator, Heathkit 1C-5218
DMM, Radio Shack 22-168A
Oscilloscope, Proteck 6502

former. You must mount the 6550 octal tube sockets so that the centers of the sockets are at least 3″ apart. I ordered the parts for this amplifier from distributors offering the best prices.

RECONFIGURATION

You may configure the two mono amplifiers to operate from a remote location via a control unit. This requires you to produce a new schematic. You must move the 5687 line amplifiers and associated components, heater switch, plate power switch, and the audio input selector switch to the new schematic. The new configuration requires the following changes:

1. Add chassis box (10″ × 6″ × 2″)—aluminum.
2. Line amplifier (5687) and associated components—move from power amplifier schematic to control unit schematic.
3. Replace power supply switches S1 and S2 in both amplifiers (four switches) with two new switches, D.P.D.T, R.S. 275-691 in the control unit.
4. Replace the two mono amplifier rotary selector switches (S1) with one rotary selector switch R.S. 275-1386 in the control unit.
5. Add eight-conductor 16 AWG cable, Belden 9405 (length as required—suggested minimum, 12′), R.S. dot Com 910-1640.
6. Add two sockets R.S. 910-5492 (attach to amplifier cables) and two plugs R.S. 910-5496 (attach to control unit cables).
7. Place the ten female RCA phono sockets designated for amplifiers on the back chassis of the control unit.
8. Add input and output RCA phono sockets to the control unit and mono amplifiers.
9. Move 100kΩ potentiometer (R1) from amplifiers to control unit. ❖

POWER SUPPLY PARTS LIST

B1	115V AC, 3½ × 1″, fan	900-2518, R.S.
T1	200V CT, 0.87A—Hammond	P-T167H200, A.E.S.
T2	6.3V CT, 10A— Hammond	P-T166S6, A.E.S.
T3	12.6V CT, 450A	273-1365, R.S.
C1, C2	180µF, 400V, 180/400 VR	ALEL
C3	0.05µF, 1600V	C-SD05-1600, A.E.S.
C4–C7	330µF, 450V,	EC-3345, ALEL
C8	1000µF, 100V	1000/100VR, ALEL
C9–C12	2.0µF, 200V	RMC-220, ALEL
D1–D8	2.5A, 1000 P.I.V.—(order 10)	900-2875, R.S. dot Com
R1	20Ω, 10W, w.w.	900-1056, R.S. dot Com
R2	330Ω, 10W, w.w.	900-1085, R.S. dot Com
R3–R6	100k, 2W, metal-oxide	900-0866, R.S. dot Com
R7	5Ω, 5W, w.w.	900-0914, R.S. dot Com
R12–R15	51k, ½W, carbon film	900-0420, R.S. dot Com
R8–R11	Potentiometer, 10k, linear, 0.5W	271-1715, R.S.
R16–R19	15k, 1W, metal-oxide	900-0718, R.S. dot Com
FH-1	Fuse holder	P.E. 071-510
F-1	Fuse 3AG, 5A, fast-blow	P.E. 071-706
S1	Switch, S.P.S.T, 3A, 115V	275-617,R.S.
S2	Switch, S.P.S.T, 10A, 115V	275-694,R.S.
AC cord	6′, 18 AWG, 1250W	910-1730, R.S. dot Com
Terminal strip	5 lug (4 packs)	R.S. 274-688
Chassis	aluminum, 17 × 10 × 3	P-H1444-32, A.E.S.
Hook-up wire	3 spools, black, red, green, #18	RS-278-1223
Ring tongues		R.S. 64-3040
Solder-rosin core		R.S. 64-006
Cable clamps		R.S. 64-3028

DISTRIBUTORS:
R.S.—Radio Shack
Radio Shack dot Com 800-442-7221
A.E.S. (Antique Electronics Supply) 6221 South Maple Ave. Tempe, AZ 85283
P.E. (Parts Express) 800-338-0531
ALEL—All Electronics 800-826-5432

Digital Measurement of the Thiele/Small Parameters

Here's some groundbreaking work in determining the all-important T/S figures for speaker design. **By Oliver H. McDaniel**

The Thiele/Small parameters[1] describe the electrical, mechanical, and acoustical properties of a dynamic loudspeaker. These parameters are invaluable in the design of loudspeaker systems. They are essential for predicting the low frequency response of woofers and midrange speakers and for designing crossover networks. The loudspeaker's impedance, from which the other parameters can be derived, is central to the design of crossover networks. For example, choosing crossover frequencies for midrange and tweeter units sufficiently above their resonant frequencies and avoiding units with high impedance magnitudes at resonance will greatly enhance the possibility of a good system design.

You can use three key parameters to theoretically predict the low frequency free-field response of the speaker in a sealed or vented enclosure. These speaker parameters are

f_S = the free air resonant frequency
Q_{TS} = the total free air Q
V_{AS} = the volume of air with the same compliance as the speaker compliance.

The low frequency response in a vented box[2] is given by

$$R_V = f^4/((f^4 - Cf^2 + A)^2 + (Bf - Df^3)^2)^{1/2} \quad (1)$$

where

f = frequency
f_B = box tuning frequency
$A = (f_B/f_S)^2$
$B = A/Q_{TS} + f_B/Q_B f_S$
$C = 1 + A + f_B/Q_B f_S Q_{TS} + V_{AS}/V_B$
$D = 1/Q_{TS} + f_B/Q_B f_S$
V_B = the enclosure volume
Q_B = the box $Q \cong 7$

You can find the closed box response by setting f_B to zero in equation 1 or by the simpler expression

$$R_C = f^4/((f^4 - (1 + V_{AS}/V_B)f^2)^2 + (f^3/Q_{TS})^2)^{1/2} \quad (2)$$

A number of programs are available that calculate and display the frequency response of a loudspeaker in sealed and vented enclosures using equation 1 with the key Thiele/Small parameters as inputs. You can vary the enclosure volume and tuning to obtain the desired results. These programs include BoxPlot[3], Abacus[4], and WinISD[5]. Each program has its own unique features.

Boxplot allows continuous adjustment of α and H, where $\alpha = V_{AS}/V_B$ and $H = f_B/f_S$. Abacus will display up to four response curves with different enclosure specifications. WinISD has a feature that allows the data to be displayed in text format that can be saved as an ASCII file for plotting with graphics software.

BASIC MEASUREMENT TECHNIQUES

These measurement techniques are primarily a digital adaptation of the methods described by Weems[2] and Dickason[6]. In this method the impedance of the loudspeaker is measured in free air, in a sealed enclosure, and optionally in a vented enclosure. The vented enclosure measurement provides an additional calculation of V_{AS} and is a good overall check of the measurement accuracy. Also included is the added mass

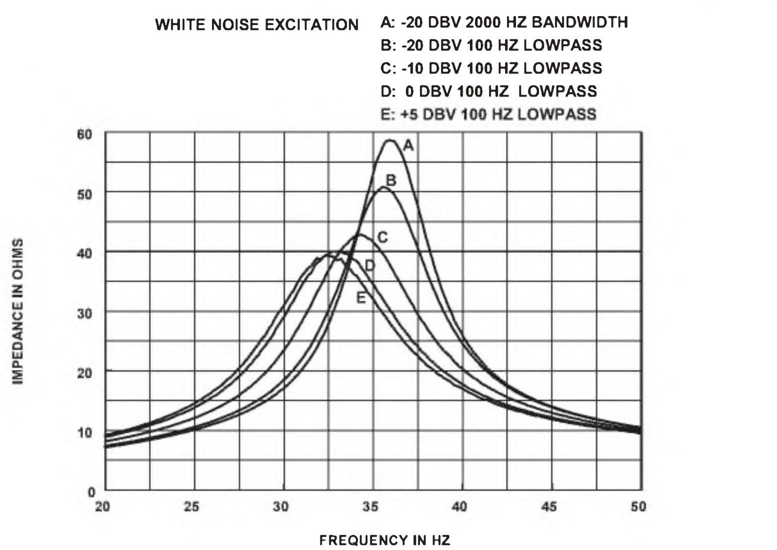

FIGURE 1: Loudspeaker nonlinearity.

B-2071-1

ABOUT THE AUTHOR

Oliver McDaniel has a PhD in Engineering Acoustics. He taught and did research in underwater acoustics and noise control for 30 years, until his retirement in 1995. He has taught courses in experimental methods in acoustics, noise control engineering, environmental noise, and fundamentals of acoustics.

method, which provides an independent check of V_{AS}.

In a recent article[7], I described digital methods for measuring impedance using software such as AtSpec and Spectra Plus. These programs use the computer's soundcard as an analog to digital converter and use the Fast Fourier Transform (FFT) to compute the transfer function necessary to calculate the impedance. AtSpec[8] is the software of choice used here due to its ease of use and low cost.

The software is set up to compute the complex transfer function and display it in terms of the magnitude and phase. My article described the experimental setup for obtaining the necessary input signals and the equations for calculating the impedance. The phase and amplitude data are exported from AtSpec as ASCII text files for use in the calculations. You can perform the calculations with any spreadsheet program such as Excel or Lotus 1-2-3, which can also graph the data.

SIGNAL GENERATOR AND WOOFER NONLINEARITY

Woofers can exhibit significant nonlinearity at low excitation levels that can result in errors in impedance measurements. The impedance magnitude and phase can vary as a function of the input power. This is illustrated in *Fig. 1* for a Madisound 6102 6.5″ woofer.

The impedance was measured with the speaker driven by white noise with a 2kHz bandwidth at –20dB re. 1V and with white noise low-passed at 100Hz from –20dBV to +5dBV. As the drive level increases, the impedance approaches a constant value indicating linear behavior. Decreasing the noise bandwidth from 2000Hz to 100Hz at the same voltage level as in curves A and B increases the power to the speaker in the frequency of interest (20 to 100Hz), which is equivalent to increasing the level for the 2kHz bandwidth signal by 10 log(2000/100) = 13dB. Band-limited noise will therefore increase the excitation of the speaker in the frequency range of interest.

This nonlinear behavior has probably not been observed with analog techniques using sine wave excitation since all of the energy with this method is concentrated at a single frequency for each data point. For digital methods the problem is solved by driving the speaker with band-limited noise at sufficiently high levels. I have observed this nonlinear effect only with woofers with foam surrounds. I have tested several speakers with rubber surrounds and the impedance results appear to be independent of drive level.

Signal generation can be easily accomplished using the signal generator included in Wavetools[9]. This software can generate signals by looping any desired Windows PCM (.wav) file that is in the Wavetools subdirectory. It can also gen-

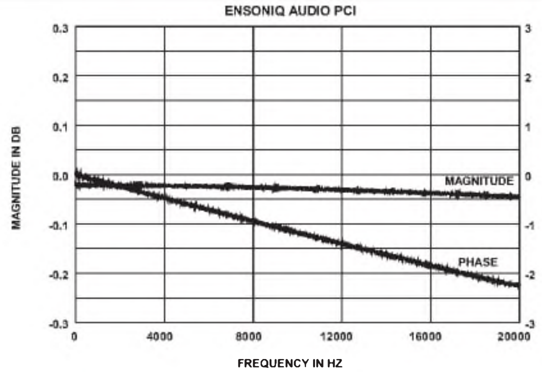

FIGURE 2: Transfer function test for Ensoniq Audio PCI soundcard. B-2071-2

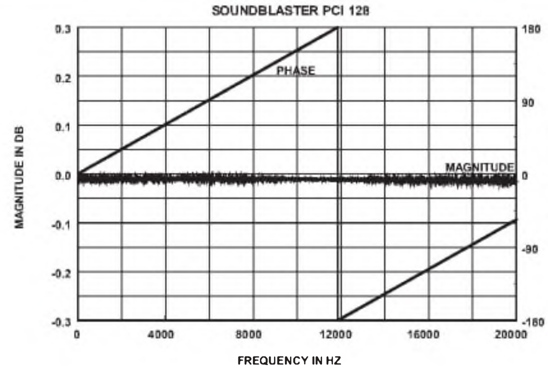

FIGURE 3: Transfer function test for Soundblaster PCI 128 soundcard. B-2071-3

erate sine, square, and pulse signals.

In order to use it for random noise generation for impedance measurements, the required Wave files are made and placed in the Wavetools subdirectory. The noise files can be generated with Cool Edit[10] at the desired sample rate, which should match the sample rate to be used in the FFT analysis. Noise files of 15-minute duration work well for obtaining smooth data. The file is then low-pass filtered using Cool Edit or Gold-wave,[11] which is much easier to use for simple low-pass filtering.

The resulting file will be reduced in amplitude but can be amplified with Goldwave or Cool Edit. The resulting level should be at about half of the maximum .wav amplitude to prevent clipping from random peaks. The desired signal can then be generated by selecting it in the Wavetools signal generator menu.

SOUNDCARD CONSIDERATIONS

The soundcard used should be checked for amplitude and phase matching between channels. This can be done by obtaining a transfer function with both soundcard channels connected to the same white noise input. Example transfer functions thus obtained for two soundcards are shown in *Figs. 2* and *3*.

The Ensoniq has a maximum amplitude error of 0.04dB and phase error of 2.2°. The Soundblaster has similar amplitude error, but the phase error exceeds 180°. This is due to a time delay between channels.

The latest version (2.2) of AtSpec can correct for time delay if the delay is a multiple of the sample rate interval, but this is not the case with this Soundblaster card. This could be corrected for as a part of the impedance calculation, but the best solution here is to avoid using a soundcard with this type of delay, for example, a different SoundBlaster card (the AWE 64), which does not have this problem.

Figure 4 shows impedance measurements of a Madisound 6102-4 woofer for free air and in sealed and vented 17.8 ltr enclosures. For these measurements the data was recorded digitally as .wav files at a 4kHz sample rate using Cool Edit. Goldwave has the same recording capability, but the elapsed time display in Cool Edit is convenient. The sound recorder furnished with Windows is not suitable because it is limited in sample rate and total sample time. Sixteen-minute samples were recorded, each producing 15MB files. The data was analyzed using AtSpec.

Similar results can be obtained by directly analyzing the data with AtSpec; however, recording the data for later analysis has some distinct advantage. Recording the data produces a permanent record which can be analyzed many times, and the FFT analysis of wave files is much faster than by the direct real-time method. The transfer functions were obtained using 16,384 point FFTs and 200 averages. This results in a frequency resolution of 0.122Hz and very smooth data.

FREE AIR MEASUREMENTS

From the impedance data we first determine the free air mechanical and electrical Qs, Q_{MS} and Q_{ES}. The S in the subscripts denotes a free air measurement. The resonant frequency f_S, the impedance at resonance R_{ES}, and the frequencies f_1 and f_2 are shown in *Fig. 5*.

TABLE 1
IMPEDANCE OF SPEAKER AROUND RESONANCE

DATA POINT	FREQUENCY	AMPLITUDE	PHASE
263	31.98	53.52	13.15
264	32.1	54.54	11.14
265	32.23	55.4	9.24
266	32.35	56.18	6.83
267	32.47	56.65	4.98
268	32.59	56.91	2.56
269	32.71	57.1	0.4
270	32.84	57	−1.73
271	32.96	56.69	−3.71
272	33.08	56.47	−5.89
273	33.2	56.26	−7.87
274	33.33	55.92	−10

TABLE 2
IMPEDANCE AROUND F_1

DATA POINT	FREQUENCY	AMPLITUDE	PHASE
199	24.17	12.82	58.95
200	24.29	12.98	58.74
201	24.41	13.16	58.79
202	24.54	13.32	58.81
203	24.66	13.46	58.77
204	24.78	13.64	58.74
205	24.9	13.85	58.63
206	25.02	14.03	58.48
207	25.15	14.25	58.21
208	25.27	14.44	58.18
209	25.39	14.65	58.11
210	25.51	14.87	57.93

TABLE 3
IMPEDANCE AROUND F_2

DATA POINT	FREQUENCY	AMPLITUDE	PHASE
344	41.87	14.57	−61.5
345	41.99	14.45	−61.66
346	42.11	14.3	−61.74
347	42.24	14.16	−61.83
348	42.36	14.04	−62.01
349	42.48	13.93	−62.15
350	42.6	13.79	−62.22
351	42.72	13.68	−62.27
352	42.85	13.56	−62.35
353	42.97	13.46	−62.38
354	43.09	13.36	−62.32
355	43.21	13.26	−62.44

The best way to determine these quantities is to read them from an impedance data sheet generated from the impedance calculation[7]. The free air data around resonance for the Madisound unit of *Fig. 4* is shown in *Table 1*.

The free air resonant frequency, f_S, is 32.71Hz, and the impedance at that frequency, R_{ES}, is 57.1Ω. The voice coil resistance, R_E, for this speaker is 3.35Ω. Note that the phase goes through zero at resonance. The quantity Z_{12} is calculated from:

$$Z_{12} = (R_{ES}\,R_E)^{1/2} = 13.83 \qquad (3)$$

The next step is to find the two frequencies, f_1 and f_2, on either side of f_S, where the impedance is equal to Z_{12}. A view of the data sheet below f_S is shown in *Table 2*.

The impedance closest to Z_{12} is 13.85, which is at 24.9Hz. f_2 is found similarly from *Table 3*. You can see from the data that f_2 is 42.6Hz. The Qs are calculated from

$$Q_{MS} = f_S(R_{ES}/R_E)^{1/2}/(f_2 - f_1) = 7.629$$
$$Q_{ES} = Q_{MS}/(R_{ES}/R_E - 1) = 0.475$$
$$Q_{TS} = Q_{ES}\,Q_{MS}/(Q_{ES} + Q_{MS}) = 0.447$$

SEALED ENCLOSURE MEASUREMENTS

This process is repeated exactly for the speaker mounted in a 17.8 ltr sealed enclosure to obtain the following data:

f_C = 55.54Hz
R_{EC} = 45.62Ω
Z_{12C} = 12.36Ω
f_{1C} = 46.14Hz
f_{2C} = 65.8Ω
Q_{MC} = 10.425
Q_{EC} = 0.826
Q_{TC} = 0.765

The C subscript denotes that the measurements are in a closed box. You can then calculate the quantity V_{AS} from

$$V_{AS} = V_B[(f_C\,Q_{EC}/f_S\,Q_{ES}) - 1] = 34.71 \text{ ltr} \qquad (4)$$

where V_B is the volume of the enclosure.

VENTED ENCLOSURE MEASUREMENTS

A typical impedance curve for a vented enclosure is shown in *Fig. 6*. Two resonant peaks occur at f_H and f_L. You can calculate V_{AS} from

$$V_{AS} = V_B(f_C^2 - f_L^2)(f_H^2 - f_C^2)/f_H^2 f_L^2 \qquad (5)$$

and the frequency to which the enclosure is tuned, f_B, is

$$f_B = (f_L^2 + f_H^2 - f_C^2)^{1/2} \qquad (6)$$

For the Madisound speaker in a 17.8 ltr vented box,

f_L = 17.46Hz
f_H = 62.99Hz
V_{AS} = 36.12 ltr
f_B = 36.12Hz

The value here for V_{AS} is within about 5% of the value obtained by the free air closed box method. This additional vented box measurement provides a good

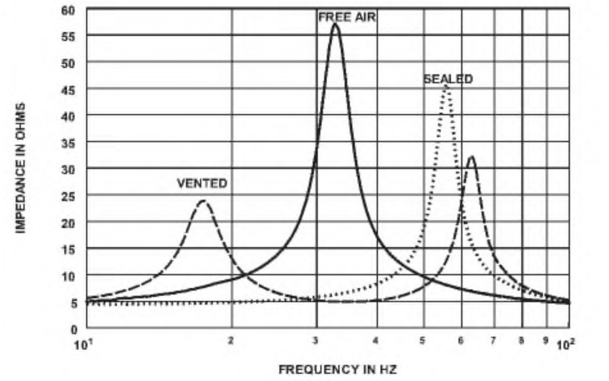

FIGURE 4: Madisound 6102-4 woofer impedance measurements. B-2071-4

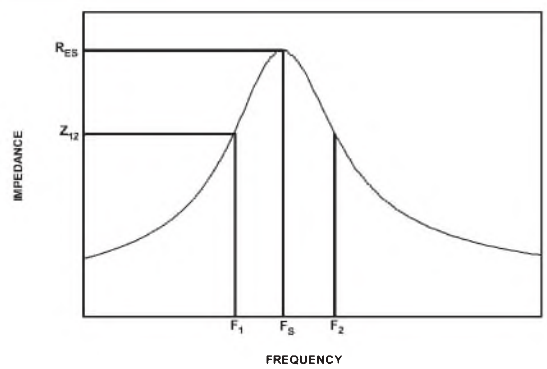

FIGURE 5: Quantities required to calculate Q values. B-2071-5

check on the measurement process. Any errors or mistakes will show up with vastly different numbers for V_{AS} for the two measurement methods.

THE ADDED MASS METHOD

The added mass (or delta mass) method has received less attention in the literature. Thiele and Small don't even mention it. Beranek[12] and Dickason[6] explain its use for measuring the total cone mass M_{MS}, which includes the acoustic mass loading. D'Appolito[13] provides an excellent treatment of the method for obtaining the total mass, the compliance, and V_{AS}. He includes an example calculation and a comparison with the other methods.

In this method the free air resonant frequency f_S of the speaker is first measured. An added mass M_A is then attached to the cone. I like to use US coins for the added mass because the mass is known and closely controlled by the US Mint.

I attach the mass to the dust cap with double-sided carpet tape. The resonant frequency is then re-measured with the attached mass to obtain f_{SA}. D'Appolito provides some good guidelines on the amount of mass to add.

The total mass is given by

$$M_{MS} = M_A/((f_S/f_{SA})^2 - 1) \qquad (7)$$

The acoustic mass loading M_{MA} and the cone assembly mass M_{MD} are

$$M_{MA} = 2.67\ \rho_0\ a^3 \qquad (8)$$

$$M_{MD} = M_{MS} - M_{MA} \qquad (9)$$

where ρ_0 is the density of air (1.21kg/m^2) and a is the cone radius in meters. The compliance of the loudspeaker suspension C_{MS} is given by

$$C_{MS} = (1/M_A(2\pi f_S)^2)((f_S/f_{SA})^2 - 1) \qquad (10)$$

You can obtain V_{AS} from the compliance by

$$V_{AS} = \rho_0\ c^2\ S_D^2\ C_{MS} \qquad (11)$$

where c is the velocity of sound (343m/s) and S_D is the cone area (m^2). There is some uncertainty in determining the cone area. The outer edge of the surround doesn't move and the inner edge moves with the same displacement as the rest of the cone. A good estimate of the true radiating area can be determined from a diameter measurement from the center of the surround.

C_{MS}, M_{MD}, and V_{AS} were measured by this method for the Madisound 6102-4 woofer used in the previous examples. Added masses of 5.67, 11.34, and 17.01 grams (one, two, and three US quarters) were added and each resonant frequency was measured. The results were

ADDED MASS (G)	CMS (M/N)	MMD (G)	VAS (LITERS)
5.67	1.82×10^{-3}	12.33	36.56
11.34	1.84×10^{-3}	12.20	36.90
17.01	1.79×10^{-3}	12.59	36.82
Madisound data	1.81×10^{-3}	14.5	39.0

This method is easy and measures the Thiele/Small parameters without the need of an enclosure. It also provides the cone assembly mass, the compliance (which can also be obtained by the free air, sealed box method), and a good accuracy check for the other two methods.

THEORETICAL PREDICTIONS

The Thiele/Small parameters required to predict the response of the measured Madisound woofer are

f_S = 32.71Hz
f_B = 34.46Hz
Q_{TS} = 0.447
V_{AS} = 34.71 ltr (36.12 for the closed box-vented box method)
V_B = 17.8 ltr

Theoretical predictions for the response of the Madisound woofer in sealed and vented enclosures are shown in *Figs. 7* and *8* for the measured and manufacturer's Thiele/Small parameters. The results were obtained using WinISD. Madisound's published sensitivity of 90dB was used. Madisound's Thiele/Small parameters are

f_S = 30Hz
Q_{TS} = 0.33
V_{AS} = 39 ltr

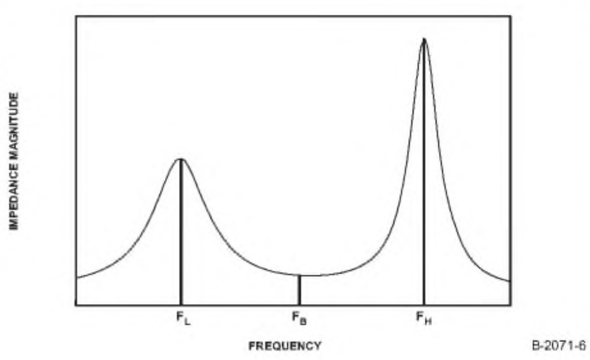

FIGURE 6: Loudspeaker impedance in a vented enclosure.

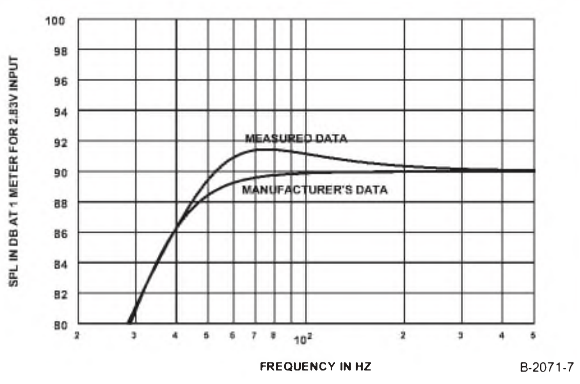

FIGURE 7: Predicted woofer response in a vented enclosure.

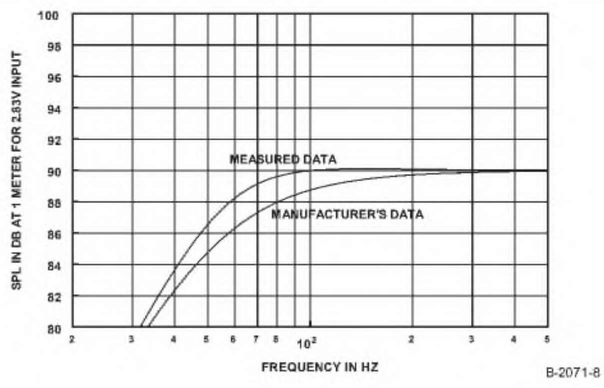

FIGURE 8: Predicted woofer response in a sealed enclosure.

The vented enclosure response shows a slight low frequency boost for the measured parameters, but the low frequency cutoff (f_3) is 42Hz in both cases. For the sealed enclosure the two f_3s are substantially different, with f_3 = 52Hz for the measured data and f_3 = 67Hz for the manufacturer's data.

CONCLUSIONS

The techniques described here provide an accurate and rapid method for obtaining the Thiele/Small parameters of a loudspeaker using inexpensive hardware and software and without external test equipment, except for a power amplifier and an enclosure. The example shown here demonstrates that the actual parameters for an individual speaker can vary from the manufacturer's published data. It has been my experience that these variations are usually higher than those shown here for the Madisound unit. The largest parameter variation is usually with f_S, and I have observed that the measured f_S is always higher than the published value even after extensive break in. Several examples include:

SPEAKER	MEASURED F_S(Hz)	MANUFACTURER'S F_S(Hz)
Audax (sample 1) HT080MO	103	80
Audax (sample 2) HT080MO	94	80
Madisound 6102-4 (sample 1)	32.7	30
Madisound 6102-4 (sample 2)	32.8	30
Madisound 6204R (4W)	34.7	26.8
Vifa M10MD-39	133	110
Vifa D27TG-35	914	650Hz

The Audax unit is a 3″ woofer and shows great promise for a very compact system using the published data, but the measured data indicates rather poor low frequency performance. The Vifa tweeter looks like a great candidate for two-way systems at f_S = 650Hz, but is not as attractive at 914Hz. The differences here could substantially affect crossover design due to the drastically different impedance around the crossover frequency. ❖

REFERENCES

1. *Audio Engineering Society Loudspeaker Anthology. Vol I.* The pioneering work on loudspeakers by A. N. Thiele and R. H. Small is documented in this excellent series with reprints of two papers by Thiele and four papers by Small. The four volumes of the series are available from Old Colony Sound Lab (PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail custserv@audioXpress.com).

2. David B. Weems, *Great Sound Stereo Speaker Manual.* TAB Books, division of McGraw-Hill, 1990.

3. BoxPlot by Rick Carlson. You can download a functioning demo version from www.speakerbuilding.com. The full version is $25 and includes a driver database and extended crossover design capability (the demo does only first order crossovers). Information on ordering this software is available at www.diamondcut.com/boxplot.htm.

4. Abacus software by Vivek Mehta. You can download this freeware from www.speakerbuilding.com.

5. WinISD software by Juma Hartikainen contains an extensive driver database of manufacturer's published Thiele/Small parameters. You can download it as freeware from www.linearteam.dk.

6. Vance Dickason, *The Loudspeaker Design Cookbook*, Audio Amateur Press, 1991.

7. Oliver H. McDaniel, "Digital Impedance Measurement," *audioXpress* July 2001.

8. AtSpec software by Paavo Jumppanen. The current version (2.2) has some significant improvements over version 2.0 described in reference 7. These include soundcard delay error correction, fractional octave band analysis (1/1, 1/3, 1/6, and 1/9 octave bands), and a wider selection of sample rates (4, 8, 11.025, 22.05, 44.1, 48, and 96kHz). The actual sample rates available to the user depend on the capabilities of the soundcard in use. The cost is $95. Details are available at the author's website at www.taquis.com.

9. Wavetools software by Paul Kellett. This very well written and free collection of programs includes a signal generator, dual channel oscilloscope, meter for monitoring soundcard input levels, and a 512 point single channel FFT analyzer. You can download it from the author's website at www.abel.co.uk/~maxim.

10. Cool Edit software by David Johnson. This program has excellent recording capability with an elapsed time display and VU meters for input monitoring. It can also do filtering, down-sampling, and many other audio editing functions. The full version of Cool Edit 2000 costs $69 and the free trial version is available at www.syntrillium.com. The free version has the save feature disabled so that the noise file generation described here cannot be accomplished with this version. An earlier free version (Cool Edit 96) does not have this limitation and can be downloaded from www.hhs.csus.edu/Downloads/. The file name is c96setup.exe. An Internet search using the file name as the key word will produce other sources for the software.

11. Goldwave software by Chris Craig. This program is similar to Cool Edit with the same recording capabilities. It has easy to use low-pass, bandpass, and high-pass features. It costs $40. You can download a fully functional demo from www.goldwave.com.

12. Leo L. Beranek, *Acoustics*, Chapter 8, McGraw-Hill, 1954.

13. Joseph D'Appolito, *Testing Loudspeakers*, Audio Amateur Press, 1998.

An SE Class A Headphone Amp

Achieve high-quality sound and low impedance with this dynamic headphone amp. **By Aren van Waarde**

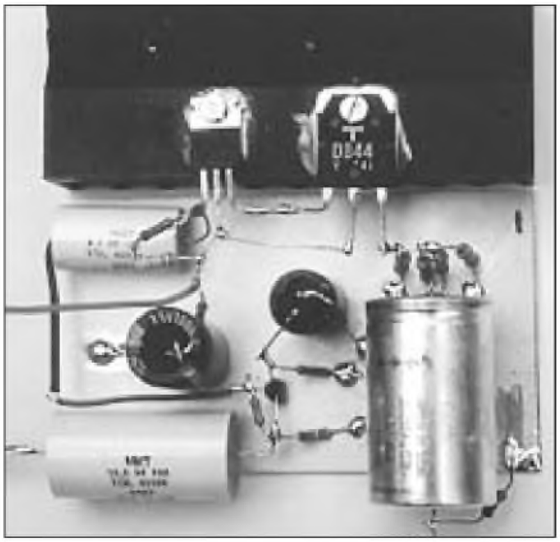

PHOTO 1: First prototype of a single channel of the headphone amplifier, hardwired in dead-bug style using a piece of unetched PC board.

I started this project after I inherited two pairs of very low-impedance headphones from my parents (a Sansui SS-2 and a Best SH-50BV, both 8Ω). It proved surprisingly difficult to drive the old headphones with modern equipment, so I thought: Why not build a dedicated amplifier . . . and use some old parts to clear the junkbox? I checked my collection of electronics magazines, but could not find a circuit that I really liked.

However, I stumbled across an interesting website on the Internet: the homepage of Rudi Stor[1]. Rudi is an experienced home constructor who has built several tube and solid-state amplifiers, using a minimalist approach. I prefer such an approach because even inexperienced builders can reproduce minimalist schematics, which you can easily upgrade by replacing parts, since the number of parts is relatively small.

One of Rudi's schematics caught my attention: a single-ended Class A headphone amplifier with only three transistors per channel—one BC109 (voltage amplifier with local negative feedback) and two 2N3055s (modulated current source). At first, I thought that this circuit could hardly sound good. Just for fun, I entered the circuit details into a simulation program (Simetrix, PSpice-based) and ran some tests (frequency response, square-wave response, maximal power output, harmonic distortion, and so on)[2].

To my surprise, the simulations indicated low THD and a high quality of sound reproduction. Moreover, the pattern of the harmonics was nice: mainly second and relatively few, weak higher-order components; in effect, a tube-like spectrum, which in my opinion is good! This encouraged me to give the circuit a try[3].

However, I didn't have old-fashioned transistors such as BC109s or 2N3055s. So, I determined whether Rudi's schematic was sensitive to component changes. The simulation program indicated that most transistors with similar voltage and power ratings would work well.

Thus I decided to use more modern, Japanese transistors: A 2SC1327 instead of the BC109 (equivalent, although the pinout is different), followed by a Darlington made of a 2SD1138 and a 2SD844 rather than two 2N3055s (*Fig. 1*). Apparently, the h_{FE} of this novel Darlington is much higher than the original one based on 2N3055s. Therefore, I needed to increase the value of the resistor at the base of the 2SD1138 from the original 220k to 1.5M. After this modification, the novel circuit drew the same quiescent current as the 2N3055 version; i.e., 100mA/channel.

Some other part values in my circuit (*Fig. 1*) also differ from Rudi's original. I replaced the 100µF electrolytic capacitor at the input with a 10µF foil cap (could be sonically advantageous), and

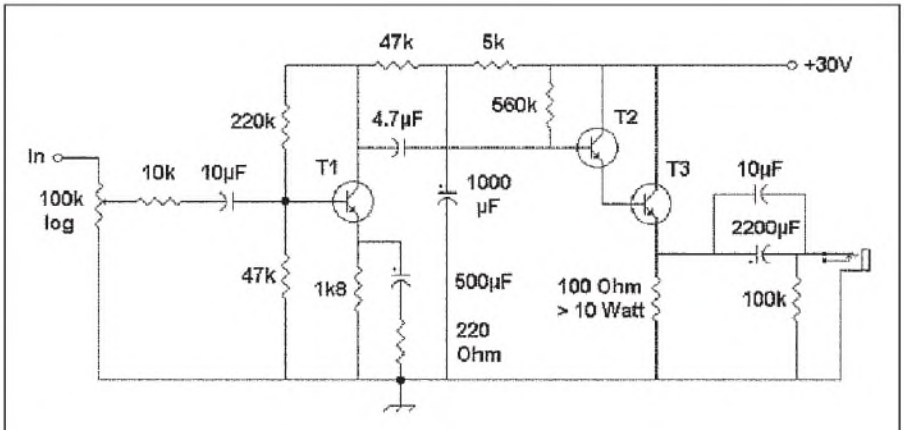

FIGURE 1: Amplifier schematic (one channel). Part values shown are of the final version of the amplifier.

ABOUT THE AUTHOR

Aren van Waarde is a biochemist working in the field of medical imaging (positron emission tomography). He has worked as a Ph.D. student and a postdoc at several universities (including Leiden University and Yale University) before accepting tenure at the University of Groningen. His passion for audio started at his ninth birthday when his parents gave him a Philips kit. Most of his current audio equipment (loudspeakers, radios, tuners, pre- and power amps, both tube and solid-state) is homemade.

the 6.8μF electrolytic cap between input stage and Darlington with a 4.7μF foil cap. Rudi used a 5000μF electrolytic to connect the headphone to the Darlington, whereas I employed a 2200μF electrolytic bypassed with a 10μF foil cap. The contents of my junkbox mainly dictated these modifications.

I built the whole amplifier without a printed-circuit board, gluing the components in "dead bug style" on a piece of copper-clad epoxy (*Photo 1*). I used the copper as a groundplane, and made all the other connections with the wires of the components themselves. This construction technique—often used by radio amateurs—is an excellent method for experimental circuits. I mounted the two Darlington transistors on a single heatsink with proper insulation (*Photo 1*).

POWER SUPPLY

It turned out that the circuit requires a good power supply. For initial listening tests, I hooked it up to my bench PSU, but this was unsuccessful. There was no appreciable noise even at very high listening levels, but I noticed an unpleasant hum. The level of this hum was independent of the volume control, so I concluded that it originated from the supply rail and that I should go for a dedicated, ripple-free PSU.

A good power supply based on the capacitance multiplier principle is presented on two different Internet pages: the Australian site of Rod Elliott[4] and the British site of Geoff Moss, which is honoring Jean Hiraga, John Linsley-Hood, L. Nelson-Jones, and J.E. Sugden[5]. The original schematic (designed by Rod Elliott and reproduced here with permission) uses a BD139 and a TIP3055, which I didn't have in my junkbox.

I replaced the BD139 with a 2SD600K (which is equivalent) and the TIP3055 with a 2SC3281 (*Fig. 2*). The 2SC3281 is a very sturdy device that you can use instead of the TIP3055 as a pass transistor. I mounted the 2SD600K and the 2SC3281 on a single heatsink with proper insulation. Construction details for the capacitance multiplier circuit are given on Rod Elliott's website [4].

TABLE 1
SIMULATION DATA (SIMETRIX) OF HEADPHONE AMPLIFIER CIRCUIT NB BC109 AND 2 × BD135 USED FOR THE SIMULATION

PARAMETER	32Ω LOAD (WALKMAN-TYPE HEADPHONE)	8Ω LOAD (SANSUI SS-2 OR BEST SH50-BV
BIAS 215MA		
Frequency response	3Hz–412kHz (–3dB)	8Hz–409kHz (–3dB)
Gain	17×	15.7×
THD (1mW output)	0.017%	0.011%
THD (10mW output)	0.055%	0.042%
THD (100mW output)	0.172%	0.427%
BIAS 100MA		
Frequency response	3Hz–429kHz (–3dB)	8Hz–411kHz (–3dB)
Gain	17×	15.6×
THD (1mW output)	0.032%	0.238%
THD (10mW output)	0.111%	0.889%
THD (100mW output)	Onset clipping	Clipping at <100mW

You can use the capacitance multiplier to power a 10W Class A amplifier, so it is somewhat overkill for the present application (current draw of 200–400mA, see the following). However, I am happy to report that the headphone amplifier proved free of hum and totally quiet with this dedicated, outboard power supply.

THE TWEAKING VIRUS AGAIN

I started listening at this stage of the project, and was pleased with the sound. But I suffer from a permanent infection with the tweaking virus. PSpice simulations indicated that the performance of the circuit at 100mA quiescent current is strongly load-dependent (*Table 1*). For headphones of 32Ω or more, a bias of 100mA is probably enough. But if the amp drives 8Ω phones, output power is limited to 10mW at 1% harmonic distortion (*Table 1*).

A great improvement (about a 20-fold reduction in THD) appears possible by increasing the quiescent current from 100 to 200–220mA. At 1mW output in 8Ω (normal listening volume for a Sansui SS-2), THD then drops from 0.238% to 0.011%. And at 10mW output in 8Ω, THD falls from 0.889% to 0.042%. At 200–220mA bias current, the amp can deliver >100mW in 8Ω, which it cannot do at all at 100mA. One hundred milliwatts is about the maximum a Sansui SS-2 can handle, and much more than you would ever use under normal conditions.

So I decided to modify the amplifier once again. I reduced the resistor at the base of the 2SD1138 from 1.5M to 560k and changed the emitter resistor of the 2SD844 from 4 × 432Ω/1W carbon resistors (connected in parallel) to an oversized 100Ω 30W ceramic resistor (NOS item for use in valve transmitters). The DC operating point values of the final version of my amplifier (*Photo 2*) are listed in *Table 2*.

MEASURED PERFORMANCE

Simulations do not always tell the truth; therefore, I measured the final version of the amplifier, using equipment from my shack: a function generator, a true RMS voltmeter, and an oscilloscope (*Table 3*). Square-wave response of the amplifier proved to be excellent, without any sign of ringing or instability. Some "tilting" of the horizontal parts of the wave occurred at low frequencies because the circuit is capacitor-coupled.

I measured output impedance by connecting a multiturn potentiometer to the output, setting the volume control to a moderate level (0.5V RMS output at 1kHz in 100Ω) and slowly reducing the setting of the multiturn pot until the voltage had dropped to 0.25V RMS ($Z_{out} = Z_{multiturn}$). At this point I disconnected the multiturn pot and measured its resistance, which turned out to be 1.5Ω. I repeated the measurement a couple of times at various settings of the volume control and consistently found values ranging from 1.5 to 1.8Ω.

Frequency response ranged from less than 10Hz to more than 130kHz (–3dB), and the circuit was capable of delivering 300mW of power into 32Ω and more than 100mW into 8Ω. So I met the design goals. However, the proof of the pudding is the eating, and the proof of an amplifier is the listening.

LISTENING

Initially, I had no high-quality headphones available to evaluate this "product." Besides the Sansui SS-2 and Best SH-50BV (8Ω, see previous) I used a Sony MDR-24 (32Ω, good-sounding pair

TABLE 2
DC OPERATION POINT VALUES OF THE FINAL VERSION OF THE AMPLIFIER

B+ from capacitance multiplier	35.2V
B+ after decoupling RC filter	29.2V
Voltage drop over 100Ω resistor	20V
Bias current (each channel)	200mA
Dissipation	7W/channel

PHOTO 2: The final version of the headphone amplifier in a metal cabinet. I used separate volume controls for each channel rather than a dual Alps pot since I like to make occasional small adjustments of channel balance.

TABLE 3
MEASURED PERFORMANCE OF THE FINAL VERSION OF THE AMPLIFIER

Voltage gain	20.6× (volume control set to maximum)
Frequency response	8Hz–135kHz (–3dB)
Output impedance	1.5Ω (at 1kHz)
Maximum output (onset of clipping)	4.5 Veff in 600Ω (34mW) 3.1 Veff in 32Ω (300mW) 0.925 Veff in 8Ω (107mW)
Signal-to-noise ratio	Not measurable with my equipment, > >70dB (hum and noise inaudible)

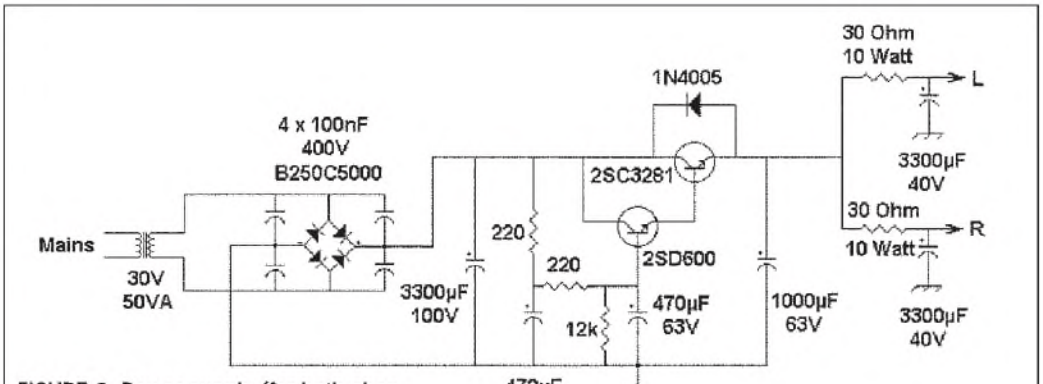

FIGURE 2: Power supply (for both channels combined). Part values shown are of the final version of the amplifier.

of cans which is sold with Discman® portable CD-players) and a Sennheiser HD-465 (a 60Ω item).

As the simulation predicted, the sound improved after I had increased the bias current. This improvement was not noticeable on the Sennheiser (60Ω), but I could hear it on the Sony (32Ω) and Sansui (8Ω). There was better resolution of fine details and a cleaner reproduction of loud transients. So I decided to stick with the high bias current.

With most recordings, the amplifier sounds sweet, clean, and involving. I must confess that I am addicted to vocal music, and I really like its rendering of the human voice. Singers are pinpointed on the stage, not artificially enlarged. There is no unnatural emphasis on sibilants. And in good recordings you can understand every word that is being said.

These are all characteristics of good equipment. Moreover, you can easily listen to several CDs in series, whereas lesser equipment rapidly causes listening fatigue.

By this I don't mean to say that the amp is perfect. I noticed two, relatively minor, flaws:

1) In rare instances, the circuit could sound slightly "hard." It was difficult to decide whether the amp, the headphone, or the recording was the culprit in such cases, but I suspect that the electrolytic output coupling capacitor may cause occasional hardening of the sound. It could be worthwhile to try different brands in this position (e.g., low ESR types meant to be used in switching power supplies) and see whether there are audible differences.
2) With the Sony MDR-24 (an open headphone system), a very high sound quality was achieved (surprisingly high for such a cheap item), but my low-impedance phones did not sound as good. This is certainly due to the fact that they are vintage items (closed systems) with particular colorations (a somewhat "hollow" and "boomy" sound, which is slight in the Sansui and annoying in the Best). However, I also seemed to hear that the bass was less well-controlled with eight rather than 32 or 60Ω phones. This could be due to the relatively poor quality of these vintage items or to a lower damping factor of the amplifier for low-impedance phones.

I seldom heard the effect, though. Tympani, for example, sounded glorious. In the "Halleluyah" chorus and the finale ("Worthy is the Lamb") of Händel's "Messiah," the reproduction of tympani was superb. You even noticed that they are above the level of the orchestra.

At this point Mr. Dell, the illustrious editor of this magazine, encouraged me to go to a hi-fi store and borrow Grado headphones for additional listening tests (does he have a share in Grado Labs?). It turned out that a small and friendly store at a stone's throw distance from the hospital where I work is a Grado dealer, so this was not too difficult a task (there are relatively few Grado dealers in The Netherlands). The sound quality of the Grados was impressive, but I suspected that the demonstration in the store was far from optimal (the headphone output of a mid-fi CD player is definitely not the ultimate as far as headphone driving is concerned).

I took a pair of SR-125 phones home and plugged these into my homebrew Class-A amp. This proved to be a true revelation, regarding the quality of both the Grados and my own product. I must confess that I have never heard such good headphone sound before (I have listened to several mid-priced AKG and Sennheiser models, besides a few electret headphones from Stax).

The Grado bass is perfect (deep and well-controlled), strings sound sweet and liquid, and very small sonic details in an orchestra or a choir are clearly audible. Apparently, the minor flaws I described previously were due to the cheap headphones rather than to the headphone amplifier. You may guess the outcome: I bought a Grado SR-125 and I have been happy listening ever since.

In summary, even novice builders can build this amplifier, which sounds great. Optimal results are obtained with 32Ω headphones, such as the Grados, and it is also capable of driving 8Ω cans. I encourage you to build this unit, and I don't think you will be disappointed. Kudos to Rudi Stor for an excellent amplifier design, to Rod Elliott for a quiet power supply, and to Ed Dell for his suggestion to give the Grados a try.

TEST TRACKS

During the testing phase of the amplifier, I listened to various kinds of music. The following recordings moved me most:

1. **Francis Poulenc: *Choeurs sacrés* ("Exultate Deo," "Litanies à la Vierge Noire," "Quatre Motets pour un temps de pénitence," "Salve Regina," "Laudes de Saint Antoine de Padoue," "Ave Verum Corpus," "Quatre Motets pour le temps de Noël") Marie-Claire Alain, organ. Groupe Vocal de France conducted by John Alldis. EMI France CDM 565165 2. Recording date: 1981, release date: 1987, ADD.**

   A marvelous recording. The purity of singing is amazing—all voices have a clear position on the stage and they sound utterly clean. The music is great, too, especially the Litany for the Black Virgin, which, with its references to human suffering, causes shivers to move along my spine. Congratulations to Alain Lanceron, Eric Macleod, and Serge Rémy (producers and engineer) for this beautiful gem. If you can appreciate 20th century religious music, buy this disc immediately.

2. **Gustav Mahler: Blumine, Fünf Rückertlieder. Johannes Brahms: Choralvorspiele op.posth.122, Vier ernste Gesänge op.121. Jard van Nes, mezzo-soprano. Robert Holl, bass. Gelders Orkest conducted by Yoav Talmi. Ottavo OTR C98402. Recording date: 1984, release date: 1986, DDD.**

   Although the music is totally different, there is a link with the previous recording since this CD also deals with loss and grief. My favorite is the Mahler hymn "Um Mitternacht" (At Midnight), which I cannot hear without getting tears in my eyes. Brahms' hymns (based on the Biblical book of Ecclesiastes) are also deeply moving. The voices of Jard van Nes and Robert Holl sound natural. Although the Gelders Orkest is a Dutch provincial orchestra without the fame and stature of the Concertgebouw Orkest, they do play very well. However, the orchestra could have been recorded with greater clarity and detail (this may be due to the acoustics of the recording venue, "Musis Sacrum" in Arnhem). Nevertheless, a great disc.

3. **Georg Friedrich Händel: "Messiah HWV 56." Arleen Auger, soprano. Anne Sofie von Otter, contralto. Michael Chance, alto. Howard Crook, tenor. John Tomlinson, bass. The English Concert Choir and The English Concert conducted by Trevor Pinnock. Archiv 423 630-2 (2 CDs). Recorded and released in 1988, DDD.**

   Much more joyful than the two previous recordings, and music almost everyone loves and knows. An excellent performance. Voices and instruments sound natural. Recorded in the famous Abbey Road Studios (London, UK) with good acoustics. See text for additional comments.

4. **Domenico Scarlatti: *Harpsichord sonatas.* Gustav Leonhardt, harpsichord (built by Skowroneck Bremen). Sony Classical SBK 60099. Recorded in 1978. Released as LP in 1979, as CD (20-bit remaster) in 1998, ADD.**

AMPLIFIER PARTS LIST

¼W carbon or metal film resistors	
R1, R1A	10k
R2, R2A, R6, R6A	47k
R3, R3A	220k
R4, R4A	1k8
R5, R5A	220Ω
R7, R7A	5k
R8, R8A	560k
1W carbon or metal film resistors	
R9, R10, R11, R12, R9A, R10A, R11A, R12A	432Ω
Note: R9–R12 and R9A–R12A were later replaced by a single 100Ω 30W resistor (ceramic item from the brand Berco)	
Variable resistor	
P1, P1A	100k potentiometer (audio taper, volume control)
Capacitors	
C1, C1A	10µF 100V MKT or MKP
C2, C2A	4.7µF 100V MKT or MKP
C3, C3A	470µF 16V electrolytic
C4, C4A	1000µF 35V electrolytic
C5, C5A	2200µF 40V electrolytic (bypassed with 10µF 100V MKT or MKP)
Semiconductors	
T1, T1A	2SC1327 (or BC109, BC169, BC184, BC239, BC549)
T2, T2A	2SD1138 (or BD239F)
T3, T3A	2SD844
Other hardware	
Two RCA input jacks (gold-plated)	
One 6.3mm (¼") headphone jack (gold-plated)	
Two knobs for volume controls	
Metal cabinet	
Heatsinks for output transistors	
Wire	
Banana jacks and plugs for power supply	
Additional Channel Decoupling	
R16, R16A	30Ω 10W wirewound
C14, C14A	3300µF 35V electrolytic

EXTERNAL POWER SUPPLY PARTS LIST

½ or 1W carbon resistors	
R13	12k
R14, R15	220Ω
Capacitors	
C6, C7, C8, C9	100nF 400V MKP
C10	3300µF 100V electrolytic
C11, C12	470µF 80V electrolytic
C13	1000µF 63V electrolytic
NB voltage ratings are overkill, but this is what I had in my junkbox	
Semiconductors	
T4	2SD600K (= BD139)
T5	2SC3281 (or TIP3055)
D1	1N4001
B1	B250C5000
Transformer	
Tr1	2 × 15V secondary, 50VA

I have included this disc because harpsichord recordings are a severe test for audio equipment. Play this disc on the CD-ROM drive of your computer with a cheap set of speakers and you will hate it. Play it on a single-ended tube amp with a proper set of loudspeakers and it will sound great. On the headphone amp that is the subject of this article, it also sounds great—like hearing Leonhardt live (I have heard him live on many occasions).

The headphone amp reveals several small details, such as bird noises in the background on track 2, a change of ambiance between track 2 and track 3, and another change between track 6 and track 7 (different takes of the recording?). These details can only be heard on high-quality equipment. Leonhardt plays the sonatas very well, but his interpretation may not be to everyone's taste. It is stern and intellectual rather than frivolous and playful.

5. **Johann Sebastian Bach: *Six Suites for Solo Cello*, Two Sonatas for Cello and Piano, BWV 1027 and 1028. Janos Starker, cello. György Sebök, piano. Mercury Living Presence 432 756-2 (2 CDs). Recorded in 1963 and 1965 with only three microphones and tube equipment. Digitally remastered and released on CD in 1991, ADD.**

A marvelous set of discs. The two sonatas are my favorites. Purists may raise their arms in horror at the idea of Bach being played on the piano, but what a great performance! The cello is very well-recorded (I used to have a roommate for several years who played this instrument). The suites are great, too.

Maybe you will not like this music at first hearing (only one cello playing for almost two hours is too much for the uneducated). But after a while, you will begin to admire the tremendous qualities of Bach's score. The six suites are like a microcosmos, spanning the whole range of human feeling and emotion.

6. **Charles Ives: Three Places in New England; The Unanswered Question; A Set of Pieces for Theatre or Chamber Orchestra; Symphony no.3 (The Camp Meeting); Set no.1. Orpheus Chamber Orchestra with Gilbert Kalish, piano. Deutsche Grammophon 457 911-2. Recorded and released in 1994, released again in 1999 (Galleria series), DDD.**

An orchestral recording with tremendous dynamics and many small details. My favorite is the third symphony. Ives' music is not everyone's cup of tea, but I must confess that his weird mixtures of church hymn fragments, brassband tunes, and revivalist songs always keep me hooked. It is amazing that the Orpheus Chamber Orchestra can play such complex scores with marvelous precision without any conductor.

Be careful when you listen to this disc with headphones: there are very soft and very loud passages. If you have adjusted the volume control in a soft passage, your ears will fry at the first crescendo.

7. **Wolfgang Amadeus Mozart: Concerto for two pianos and orchestra in E flat major KV365; Concerto for three pianos and orchestra in F major KV242; Fantasia in F minor KV 608; Andante and variations for four hands in G major KV501. Murray Perahia and Radu Lupu, piano. English Chamber Orchestra. Sony Classical SK44 915. Recorded in 1988 and 1990, released in 1991, DDD.**

A remarkable recording. The playing of Lupu and Perahia radiates joy and wit—my spirits rise after the first bar! Great music, excellently played, and well-recorded by the CBS team. Every detail of the toucher seems to be revealed by the headphone amplifier. However, the recording is rather close-miked, in the CBS tradition, which means that you hear relatively little hall ambiance. But what a great disc! ❖

REFERENCES

1. Rudi Stor, Audio page, http://www.rudistor.com/sound-lab. Rudi's site is physically based in Trieste (Italy).
2. The Simetrix program is a product of Newbury Technology, http://www.newburytech.co.uk. I used release 3.1d, © 2000. You can download fully operative demo version from the Newbury website.
3. The schematic of the headphone amp is reproduced with permission of the designer (R. Stor). Some component values were altered (see text).
4. Elliott Sound Products, http://www.sound.au.com.
5. Class A amplifier site, http://www.gmweb.btinternet.co.uk.

A Tube Preamp for Computer Soundcards

If you dig the sound from tubes, you'll appreciate this versatile preamp for both studio and concert venues. **By Graham Dicker**

PHOTO 1: Complete preamp with plugpack.

In this day of the digital domain, many musicians and recording studios use computer-based audio editing and multitrack facilities. The standard setup is the use of 24-bit soundcards with an impressive theoretical dynamic range of 144dB (6dB/bit). The results are truly pristine and clinical, but lack the warmth and character that a tube front end can offer, hence the need for this project.

Many domestic soundcards can also benefit from an outboard universal tube preamp. As in most microphone channels, the op-amps used often produce slew-induced and other distortions, whereas the line-level inputs suffer less from this problem.

DESCRIPTION

This preamp offers several unique design features that make it a very handy piece of equipment to have in the studio. It offers balanced or unbalanced inputs and outputs, easily changed tube selection for different uses and applications, and variable output level up to +8dBm.

Uses include microphone, guitar, or line input preamp for recording, or as an outboard preamp to drive mixers or even power amplifiers directly. You can use the same unit at a desk insertion point to sweeten the overall sound or as an external audio post processor. The preamp features separate input stage and master volume controls. This way you can overdrive the input or run it clean as required, with the output level setting adjusted for the sensitivity required.

While the best solution for my applications has been to use the 12A3/12AT7 tube, you can use the 12AU7 for less overall gain, and the 12AX7 for more gain. Another of my favorite sounds comes from the RCA 5963A, good if you want a traditional Nashville sound for your recordings. Mullard tubes give a warm, rich character to the sound. A good selection on hand provides the option to change the sound qualities at will.

While the preamp that I built is a mono one, you can just as easily construct a stereo version by duplicating the circuit (a point made by John Boath in his sidebar evaluation). Make sure that you also duplicate the power-supply section to improve crosstalk and ensure adequate stage decoupling. A shared H.T. transformer is quite OK, because the current requirements are small.

HOW IT WORKS

Photo 1 shows the completed preamp with external 12V AC plugpack to power the unit. The use of a plugpack solves a number of problems, including reducing one of the main causes of hum from magnetic induction; the ready availability of an approved power transformer in all countries of the world; safety in that only low voltages come out of the plugpack, thus removing the requirement to wire up mains voltages; and assisting in the safety aspect.

POWER SUPPLY

The schematic diagram in *Fig. 1* shows the plugpack output is wired to a SPST on/off switch mounted on the front panel, and the output of the switch is wired to a front panel mounted, power indicator LED current limited by a 1k0 series resistor. One side of the 12V AC goes to power-supply ground, the tube filaments, and the secondary of a reverse-connected power transformer to supply the H.T. for the tube. I tried various methods to feed the filaments, in-

PHOTO 2: Closeup of preamp from front.

SOUNDCARD PREAMP EVALUATION ONE

By Phil Crossing, Crescent Studios, Semaphore, South Australia

Like a great many guitarists who have gone back to tube amps over the past few years, I have noticed that there is nothing like tube distortion as opposed to transistor distortion, which is rude and unmusical. This preamp is so useful, it lends itself to many other applications in the studio apart from recording guitars.

What this tube preamp does for me is to "de-glass" digital vocal recordings, which are particularly noticeable in loud sections of a track. Even though compressors can limit transient peaks to prevent clipping, audible distortion was always present, producing disappointing results. This tube preamp serves to blur the edges on the waveforms in a manner that is very musical and in keeping with what our ears have been listening to and appreciating all our lives.

The negligible noise floor and massive headroom makes this unit very usable in the studio. The ease with which tubes can be interchanged to emulate a genre or style makes a great deal of sense, and I highly recommend a good selection for different applications and sessions.

A great tip that I would like to offer is to try recording midi instruments (i.e. synthesizers, samplers, and sound modules) through it to take that "very Samie GM" sound to your midi files. It works out well as an external overall processor to sweeten up final mixes.

For overall use my preference is to use the RCA 5963A, or a vintage 1953 Mullard 12AU7. It is virtually impossible to overdrive the preamp with microphone level inputs in this manner. This also provides a nice character for recording blues or country tracks.

Best overall sound for guitar use was the ROLA 12A3, by a very large margin over the more common Mullard 12AX7; the 12A3 is far warmer in overdrive mode, and a much quieter and less microphonic tube. The Toshiba 12AT7 gives good performance in a line driver application, or to drive a power amp directly for stage use from a microphone or guitar. I have used this preamp to directly drive a pair of Yamaha PDM2200 power amps (1400W RMS plus) to full power with no problem at all.

The best description of this preamp that I can offer as a sound engineer is it offers a "lights on/lights off" effect when it is in use. I now never record vocal tracks without it.

cluding a regulated DC supply, a 500R Morganite preset pot with the wiper to ground to null the filament AC noise; and the simple grounding of one heater pin, which was the final solution. In practice I found little difference in the noise performance.

The key here is to wire the input stage triode filament to ground. There is a difference in hum and noise by about 1dB if you wire the output triode filament to ground. A simple reversing of the fila-

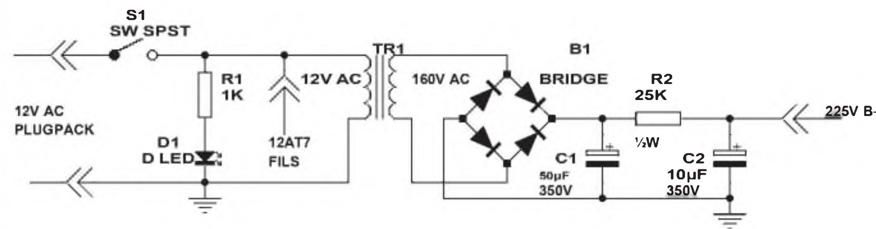

FIGURE 1: Power supply schematic.

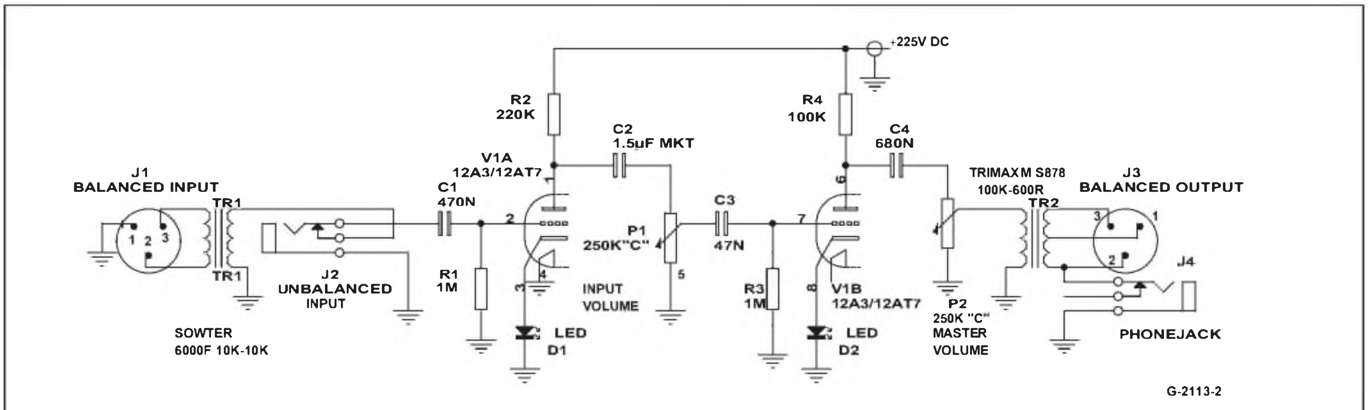

FIGURE 2: Preamp schematic.

ment leads will show the difference.

The reason for this abnormality lies in the construction of twin triodes, which are constructed internally with two separate triodes in the glass envelope. The heaters are wired in series with the center tap brought out to a separate pin (usually pin 9). In this way the tube can be wired up with dual voltage filaments of either 6.3 or 12.6V. With triodes of this nature I have also used the method of taking pin 9 to ground instead of using a nulling pot.

I used a novel method of reverse connecting a 240V to 18V 5VA power transformer to get 160V RMS for the high tension supply. This is rectified by 4 × 1N4004 diodes to end up with +225V DC for the tube anodes. The input filter is an RC pi filter consisting of a 47µF 350VW computer grade electrolytic (often found in discarded computer power supplies), a series 25k resistor made up from 2 × 47k resistors in parallel, and an output capacitor of 16µF 350VW. As the supply current is approximately 2mA, this configuration provides a ripple-free power supply.

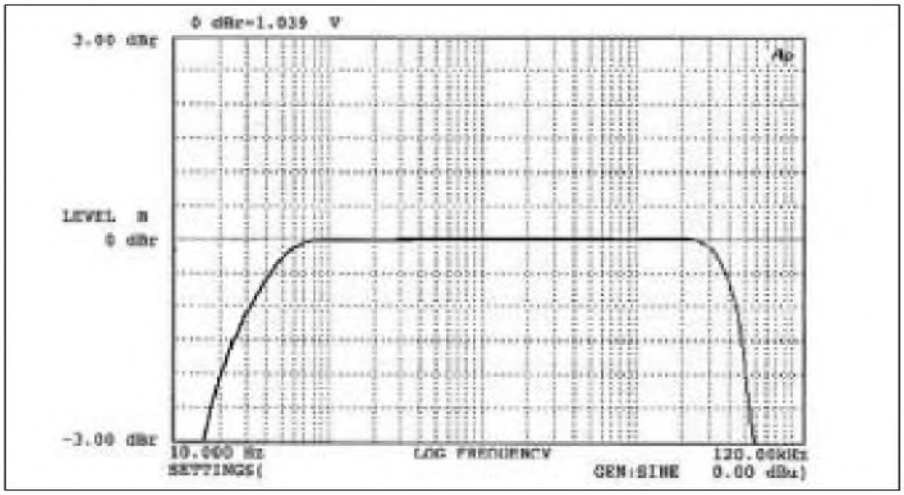

FIGURE 3: Sowter 6000F frequency response.

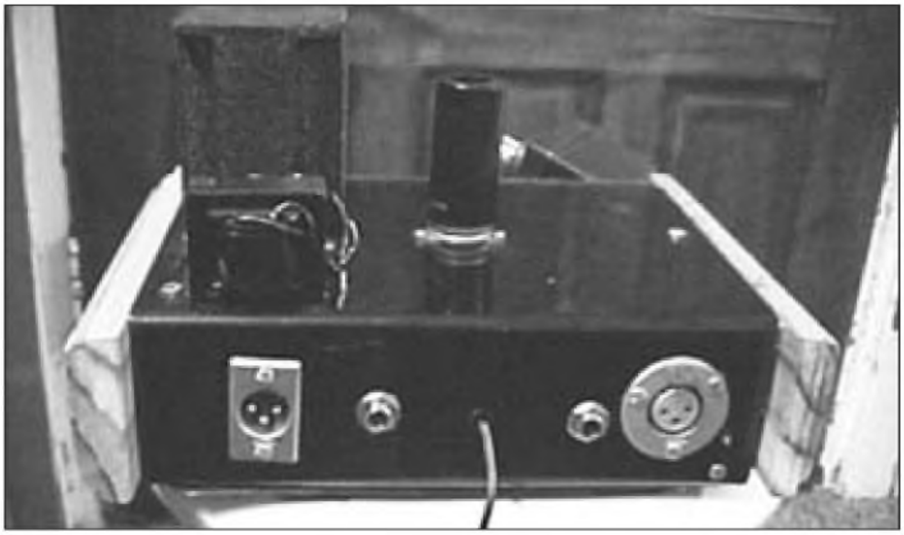
PHOTO 3: Rear view of preamp.

PREAMP

The preamp schematic (*Fig. 2*) shows a dual triode 12A7/12AT7 wired as two voltage amplifiers V1a and V1b. The input stage (V1a) has a plate load of 220k and a quiescent current of around 500µA. This is almost on the verge of anode starvation, and uses the nonlinear characteristic of the LED in the cathode bias circuit to best advantage, to produce up to 10% second harmonic distortion under large signal swings.

If so desired, you can add 1k0 grid stopper resistors to improve stability and SID, which may also provide an audible figure of merit. The grid is returned to ground via a 1m0 grid resistor. The .47µF input coupling capacitor and grid resistor form a high-pass filter with a turnover frequency of 0.34Hz.

I used the same turnover frequency for the second voltage amplifier V1b. The output from V1 is coupled via a 1.5µF output capacitor to the input volume control. This has a turnover frequency of 0.4Hz. The output impedance of this stage is around 13k and provides a voltage gain of 50 times.

The input transformer provided a challenge: Because of the general-purpose usage, any choice would be a compromise. The transformer I chose was the Sowter 6000F, made in the United

Kingdom. This company and transformer come with good credentials, with many manufacturers such as Ampex, Fairchild, Neve, and others using the products. The transformer is a 10k to 10k isolating type, Mumetal shielded and screened. The maximum input level for less than 1% THD is around +12.5dBm. This makes it ideal for line level, but a compromise for microphone level in regards to optimum power transfer and noise performance.

The basic characteristics of the transformer are: Mumetal core for minimal harmonic distortion, internal electrostatic shield for high common-mode noise rejection, and Mumetal can for magnetic shielding. The frequency response curve is shown in *Fig. 3.*

IMPEDANCE RATIO 1:1
MAXIMUM LEVEL 0.5% THD at 50Hz +12.5dBu
DISTORTION at 50Hz (40Ω source) 0.1% at +6.5dBu
INSERTION LOSS 7k5Ω source 10kΩ load 0.75dBu
FREQUENCY RESPONSE ±0.5dB 50Hz to 50kHz
INPUT/OUTPUT BALANCE at 15kHz >50dB.

The transformer primary winding is connected to a Canon XL3-F female input connector. The secondary output is switched out when a 6.5mm phone plug is inserted into the switched input socket. This ensures that no inductive effects load the unbalanced input. If you wish, you can wire a Canon plug in unbalanced mode equally, but you then restrict the input impedance to 10k. By switching out the transformer, the input impedance of V1a is 1M0, suitable for a number of older-style crystal pickups and microphones, as used in phono cartridges, acoustic pickups for guitar, double bass, and harmonica.

The transformer input impedance is nominally 10k and suitable for bridging 600Ω lines, or feeding from CD, cassette players, and tuners. An added bonus is the extended low-frequency response when driven from a low impedance source, i.e. 600Ω. Some microphones have output impedances of 50 or 200Ω, and many sound sources such as CD players often have op amp output stages with output impedances of

less than 1k0. While the frequency response curve may not look that impressive initially, on tests with a low impedance source the response soon looks more favorable.

V1b has a quiescent current of 1mA, a voltage gain of 46, and an output impedance of around 11k. The 680nF plate capacitor and the combination of the output transformer with the master volume control set the output corner frequency to 2Hz, making the dominant poles determined by the input and output transformers. The output circuit that I used is a little different than would be expected; it offers a few advantages over placing the primary in the plate circuit of V1b.

The main advantage is there is no DC component flowing through the primary in this circuit. This results in lower overall distortion and an increase in the manufacturer's maximum level rating, as there is no DC to saturate the core. This is a technique I have also used in single-ended power amplifiers with great success. The downside is lower efficiency.

Even though the source impedance varies with the settings of the master volume control, a positive effect is an improvement in the damping factor at low volume settings. The second benefit is that as the master volume control is reduced, so is the residual noise. The low noise from this preamp is indeed one of its features.

The output transformer is a Trimax MS878 rated at +10dBm with jumperable primaries of 25k or 100k, and secondaries of 150 or 600Ω. Under normal soundcard conditions, the output level used would be around −10dBm; however, most studio desks use +4dBm, and with the transformer used the headroom is around 18dB before severe distortion is caused through the transformer saturating. A great use for driving normal phone lines to extend the frequency response is to use the output wired as a 150Ω source.

ASSEMBLY

In *Photo 2* you can see how I assembled the preamp on a 200mm section of builders "C" channel. This is available as an off-cut from most supply houses for a few dollars. For the final finish, you may simply paint the chassis, powder-coat it, or plate it in chrome or gold. The chassis shown here is finished with black powder coat.

The ends are of pine with a routed edge, finished in two-part clear epoxy. When routing the end pieces, always remember to continue past the end of the

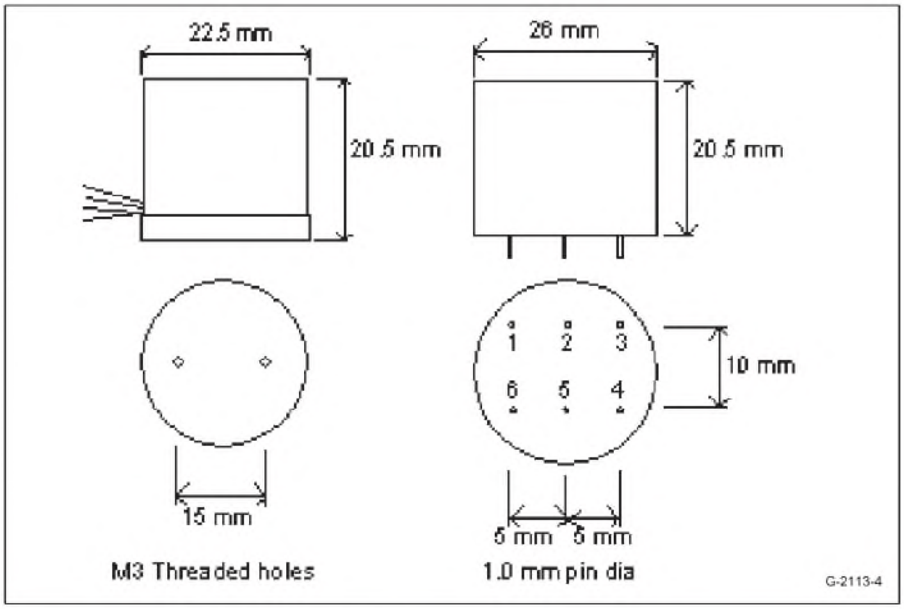

FIGURE 4: Sowter 6000F transformer styles.

TABLE 1
SOUNDCARD PREAMP BILL OF MATERIALS

ITEM	QUANTITY	EACH	COST
Powder coated chassis	1	$25	$25
Timber end plates	2	$12	$24
12V 1A plugpack AC	1	$29.95	$29.95
HV transformer DSE M2155	1	$22.50	$22.50
1N4004 diodes	4	$.35	$1.40
47μF 350VW electrolytic	1	$12.95	$12.95
16μF 350VW electrolytic	1	$4.95	$4.95
RESISTORS			
250k "C" log pots mono	2	$9.95	$19.90
1k0 ½W 1% Beyslag resistor	1	$.06	$.06
47k ½W 1% Beyslag resistor	2	$.06	$.12
100k ½W 1% Beyslag resistor	1	$.06	$.06
220k ½W 1% Beyslag resistor	1	$.06	$.06
1M0 ½W 1% Beyslag resistor	2	$.06	$.12
LEDS	3	$.15	$.45
CAPACITORS			
.47μF 250VW MKT	2	$1.95	$3.90
1.5μF 250VW MKT	1	$3.95	$3.95
.68μF 250VW MKT	1	$2.65	$2.65
Sowter 6000F input transformer	1	$99	$99
Trimax MS878 output transformer	1	$129	$129
9 Pin ceramic noval valve socket	1	$19	$19
Rola 12A3 or Mullard 12AT7 twin triode	1	$89	$89
Canon XL3-P plug chassis mount	1	$6.95	$6.95
Canon XL3-S socket chassis mount	1	$6.95	$6.95
6.5 MM phone socket mono	1	$2.95	$2.95
6.5 MM phone socket mono switched	1	$3.95	$3.95
LED BEZEL	2	$.25	$.50
Front panel brushed aluminum	1	$6	$6
Power switch SPST toggle	1	$4.95	$4.95
SATO large Diam 50MM knob	2	$5.56	$11.12
2 way tagstrip	1	$.94	$.94
TOTAL			**$532.33**

* Bill of materials costs in Australian dollars (1USD = 2AUD—Ed.)

wood with the router to ensure that splits or poorly defined edges do not occur. The ends are fixed to the chassis with a section of aluminum angle 1cm × 1cm × 5cm long. Affix a few self-tapping screws to the bracket and wood side panels, where you use nuts and bolts to fix the assembly to the main chassis. A piece of flat steel covers the preamp bottom for safety; here I used the rubber feet mounting screws to also hold

SOUNDCARD PREAMP EVALUATION TWO

By John Boath, Rock Musician, Happy Valley, South Australia.

My evaluation of the soundcard tube preamp had two purposes—using it as a guitar line preamp for both live and recording purposes. Being a rock guitarist with many bands over the last three decades had already converted me to being pro valve amplifiers. My own collection includes a Fender twin reverb, a Dynacord 200W, and several smaller low-power 10W valve amps. Guitars include a vintage Stratocaster, a 1950s Les Paul, and others.

I also dabble a little in building home-grown amp and arranging and sequencing classic rock tracks on my home PC, so evaluating this preamp was a most enjoyable experience. The four most unusual aspects of this preamp are:

1. Interchangeable tubes for different applications and sound characteristics.
2. The option of balanced or unbalanced inputs and outputs, allowing use with professional and band gear without patch cord pads and matching transformers.
3. The unusual output circuit that allows level matching with just about anything without a noise compromise. This unit is very quiet.
4. Dual gain controls. It is a truly universal application preamp, for which I think up new uses almost every day.

With the Mullard 12AU7 in the preamp, the sound quality with the Les Paul is unequalled with any preamp at any price. This includes original Neve tube mixing desks and the best amps that Fender, Vox, or Marshall have ever made. It matters not whether you use it to drive a soundcard for recording or a big power amp on stage . . . the sound is so sweet without any hint of clipping or overdrive. If you like country or jazz, this is the way to get a great sound.

(Continued on page 42)

With the ROLA 12A3 or the Svetlana 12AX7 tubes and the Stratocaster, this preamp changes color entirely to produce some of the best overdriven sounds I have ever heard, with the cleanest sustain and no sign of anything nonmusical in its output.

The 12A3s also work a treat with the preamp in vocal mode. It is here that this preamp really shines as a microphone amp. Rarely do vocalists invest in tube add-ons, but this one is a must. I tested it with a Shure SM58, on stage, with our lead singer doing rock numbers, and there never occurred a sign of clipping or overdrive. The headroom is phenomenal, and best of all, no need to have to tweak knobs all the time; this one is a "set it and forget it" deal.

I used it as a recording preamp with externally powered KM84, C451, and C12A condenser microphones. For vocal, acoustic guitar, and drums, the clarity is fabulous. This preamp should be in every musician's kit. My only regret is that it was not built in a two-channel version for recording and post processing, or to daisy-chain the output of one channel into the others' input for those Hendrix-type sounds. I guess you could always use two monoblocks. On a scale of one to ten, this one gets a twelve. Best of all, it's easy to build.

the bottom cover plate in position.

Using a chassis punch (Q-max), first mark and punch all of the large holes for the tube sockets, pots, switches, output transformer, and input/output sockets. Next drill all of the small mounting holes and the holes for the LED indicator bushing. You can use another LED bush as an input grommet as shown in *Photo 3* for the plugpack lead. Ensure that all holes are deburred with an oversize drill on both sides of the chassis.

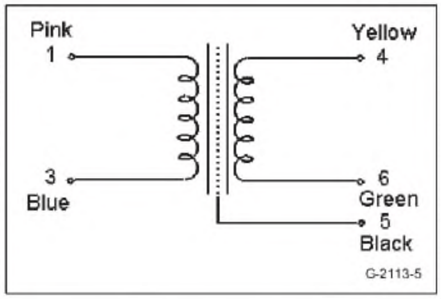

FIGURE 5: Sowter 6000F connections.

Mount the screen-printed front panel to the chassis with the pots, on-off switch, and LED bush. Next bolt in the input, output, and tube sockets, HT and output transformer, and then mount the Sowter input transformer as per *Photo 4.*


PHOTO 4: Closeup of input wiring.

The mechanical details are given in *Fig. 4* for the input transformer. You may also elect to mount the H.T. and output transformer at the last minute to give you a relatively flat surface on which to rest the inverted chassis for wiring up the other components.

The chassis wiring is point to point, with two main earth systems. The H.T. and filament wiring is direct to chassis ground. All of the tube audio paths including the grid resistors and cathode LEDs are wired to the tube spigot. The tube grid is connected to a short length of shielded cable, which is wired to the unbalanced input socket.

The ground for the two volume pots is from a short length of shielded cable between the spigot and a common ground wire between the two volume controls. In this way there is only one audio earth, the one at the input socket. This method ensures no earth and hum loops, and is a method I have used for the last 35 years in building tube amplifiers. You can see the wiring in detail in *Photo 5*.

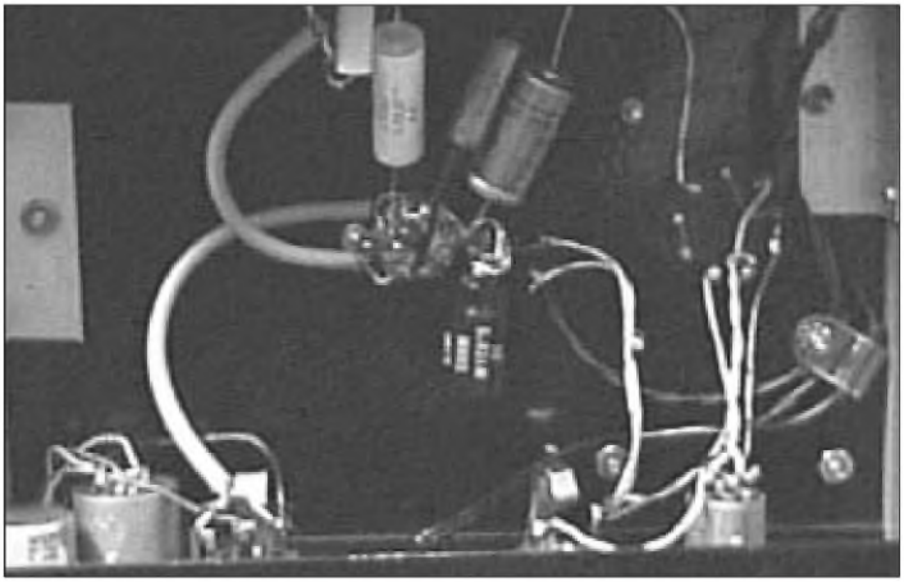

PHOTO 5: Under chassis view.

To wire the input transformer, follow the method I used in *Photo 4*. The transformer connections are shown in *Fig. 5*.

TESTING

When you have finished all of the wiring and after a visual inspection, be sure the tube is out of the socket before you plug in the plugpack switch. The front panel LED should light. With a multimeter check the H.T. for +225V DC, and pins 4 and 5 of the tube for 12V AC. If all is OK, turn off and plug in the tube (watch out, the H.T. electrolytics are still charged). Power on again and check V1a pin 1 for around 60–100V DC. Next V1b pin 6 should be around 100V DC if all is well. Hook up some audio and try it out. ❖

Simple Satellites, Part 2

Follow these steps to build the type of small satellite you need to accompany a subwoofer. **By G. R. Koonce and R. O. Wright, Jr.**

This section contains construction details that are common to all boxes. The individual construction sections contain only the dimensions and details that pertain to individual box types. All panels in all boxes are built from ⅝″ thick particleboard.

COMMON CONSTRUCTION FEATURES

Several panels of each box require cutting holes and routing. This can be extremely difficult to do on the small particleboard panels used here. The approach we used was to lay out the front panel (FP), the damping panel (DP), and the back for one box on a single piece of ⅝″ particleboard with gaps for cutting. We made two identical boards for the pair of boxes, did all the necessary machining on these two boards, and then cut out the various small panels. We highly recommend this approach, which worked well.

The boxes are designed so that the effects of cutting errors are minimized. If you cut all pieces having the same dimension with the same saw setup, these boxes should fit together without a problem. Cutting some panels oversize and later flush-cutting them should not cause a problem.

You should cut the fill strips—to cover the front of the pedestal below the bottom board—to the proper width when you cut other pieces of this dimension. However, you should leave the exact height of this piece for later. Boxes #5 and #6 do not have a fill strip, because the front panel extends to cover the pedestal.

The FP requires the most work. For most boxes you cut a proper size hole for the driver. On the Infinity plate driver boxes, you need to cut an odd-shaped hole. The boxes mounting the 3″ TBspeakers drivers require some rabbeting to flush-mount the driver.

For all boxes, you must relieve the

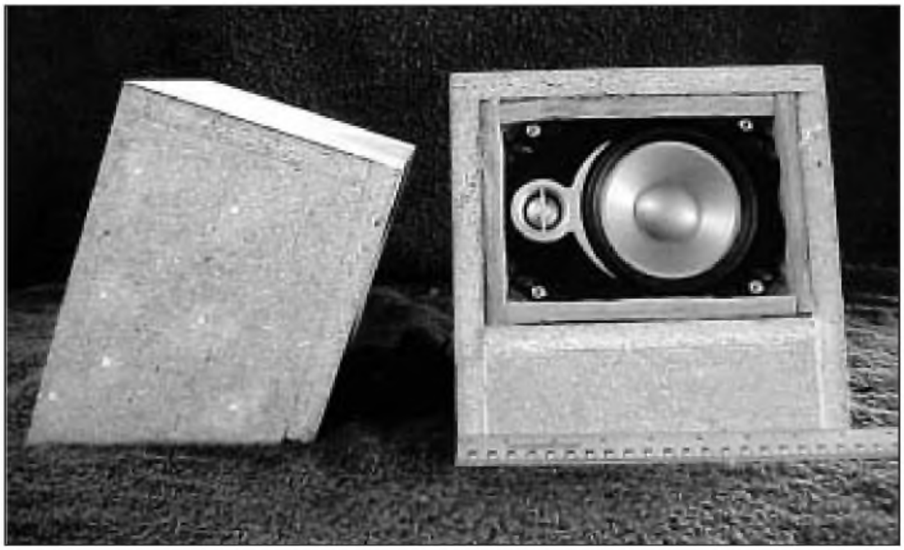

PHOTO 2: Finished boxes #1 and #2.

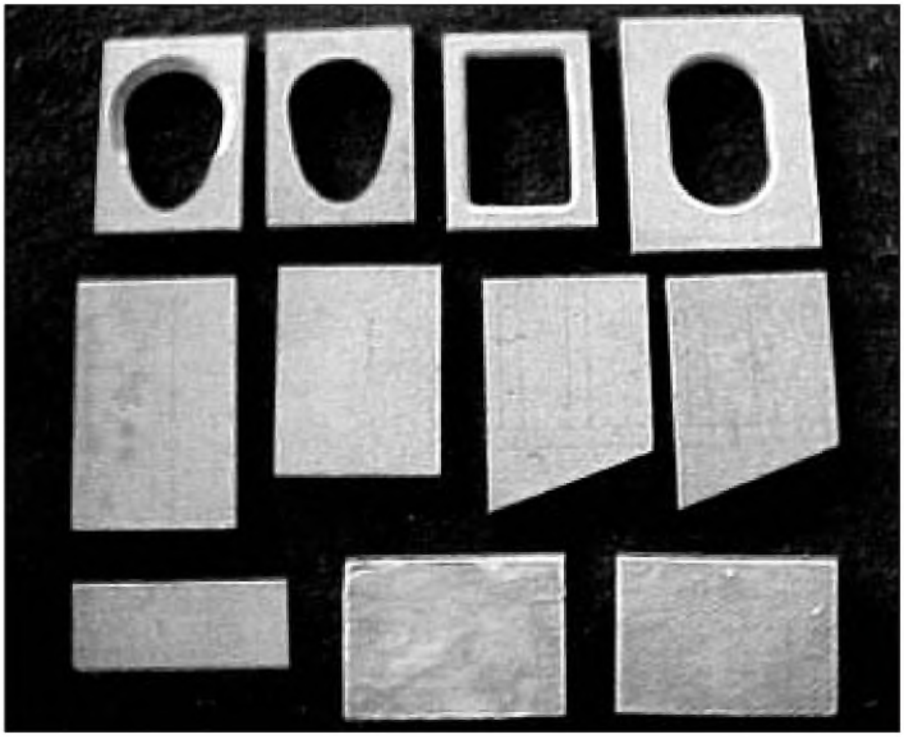

PHOTO 3: Array of pieces to build boxes #1 and #2—see text for layout.

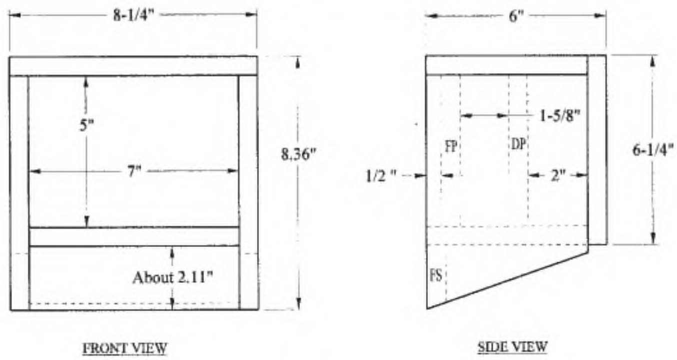

FIGURE 28: Front and side view drawings for boxes #1 and #2.

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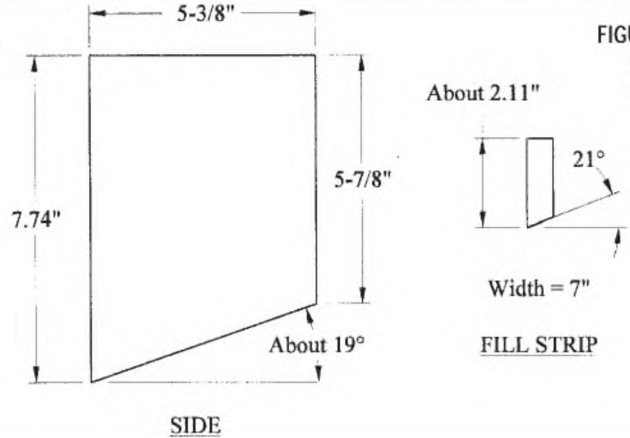

FIGURE 29: Views of side and fill strip for boxes #1 and #2.

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back of the FP to allow air access to the rear of the cone. For most drivers this means filing or routing to remove particleboard between the struts of the driver frame. A routing bit that cuts at a 45° angle works well in cutting these reliefs.

The FP for the boxes using the car coax drivers covers the entire front of the box, making the edges visible. We made it oversize and cut it to size with a flush-cutting router bit after installation. You should drill the holes for the driver mounting screws before assembly and verify that each driver fits properly into the FP.

The DP requires a large rectangular

TABLE 6
PIECES FOR BOXES #1 AND #2

ITEM	SIZE IN INCHES
Back	6¼″ × 8¼″ (cut 6⅞″ × 8⅞″ for flush routing)
FP	Front panel—5″ × 7″
DP	Damping panel—5″ × 7″
Top	8¼″ × 5⅞″ (cut 8⅞″ × 5⅞″ for flush routing)
Bottom	7″ × 5⅞″
Sides	See *Fig. 29*
FS	Fill strip—see *Fig. 29*
OC #705	Damping layers—5″ × 7″

Note: Finished size shown. Raw size actually cut in parentheses.

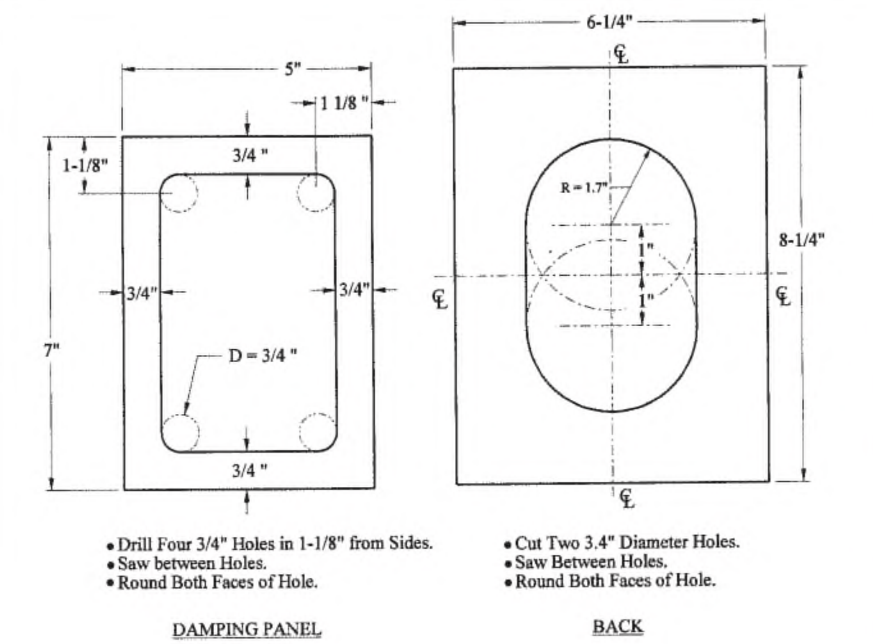


FIGURE 30: Views of damping panel and back for boxes #1–#4. B-2214-30

TABLE 7
PAIR OF UNMODIFIED #1 AND #2 SPEAKERS

QTY.	COMPONENT
	⅝″ particleboard to build enclosures
2	Infinity Kappa 462.5 compound drivers—they are sold as pairs
18	#8 × 1½″ flat-head wood screws to attach the backs
8	#6 × ¾″ pan-head sheet-metal screws to mount the drivers
18	1″ × ¼″ diameter dowels—inserted to hold back screws
	1″ ELCO panel nails or equivalent for enclosure assembly
	1⅛″ ELCO panel nails or equivalent for enclosure assembly
4	5″ × 7″ pieces of 1″ thick Owens-Corning #705 material
	Grille cloth to cover hole in the back and the box front
	Scraps of Owens-Corning #705 material for grille frame edging
	No. 18 zip or equivalent wire for drivers
	Nominal ½″ thick fiberglass to line enclosures
Misc.	Rubber cement, wood glue, and silicon rubber
	Staples and staple gun to attach grille cloth

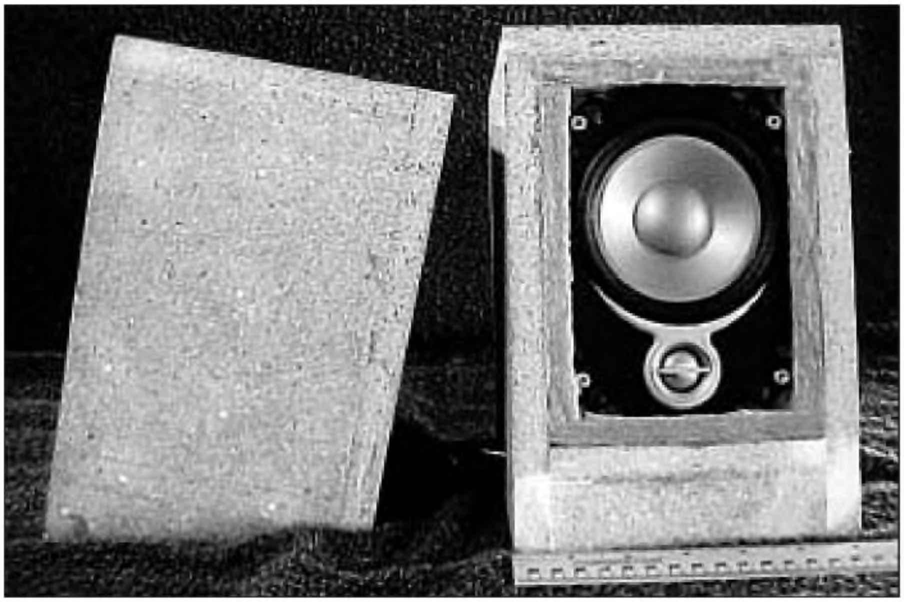


PHOTO 5: Finished boxes #3 and #4.

hole, which you make by drilling holes at the four corners and sawing between them. Unless indicated otherwise for a particular box, rout both sides of this hole with a ¼″ radius rounding-bit or file to smooth the edges and thus prevent the damping material from being pressed up against a sharp edge.

The drawings show the DP inset 2″ from the rear of the box. This is a nominal dimension. You need to cut the two nominal 1″ thick damping layers and measure their total thickness. Install the DP so that attaching the back will compress the damping layer about 1/16″ to ⅛″.

For the five pairs of satellites, the DP inset from the rear was actually in the 2 1/16″ to 2⅛″ range. This changes the depth of the dead-air volume, which is not a problem. On one satellite type (boxes #9 and #10) the DP has two holes with a braced stiffener across the center (covered later for that box type).

The back covers the entire box and thus the edges are visible. The best way to get an accurate fit is to make the panel oversize and after temporary installation to cut it flush with a flush-cutting router bit. For each box the final size for the back is shown and in parentheses the size actually cut for later routing.

With this approach the back is removable and when reinstalled fits correctly. The back board also has a large hole in it. After cutting this hole, you should rout both sides with a ¼″ radius rounding-bit or file. The inside of the hole is covered with grille cloth so that the fiberglass-based damping material

PHOTO 4: Shape of hole and relieving for Infinity plate driver.

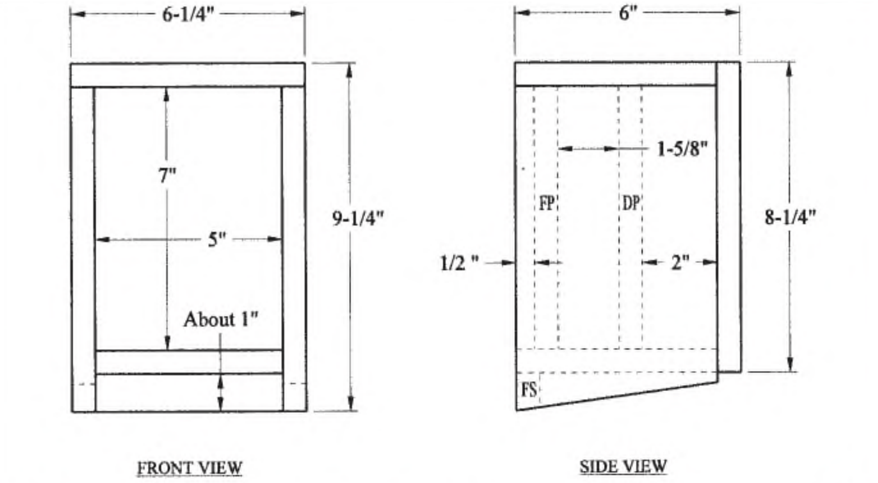


FIGURE 31: Front and side view drawings for boxes #3 and #4.

TABLE 8
PIECES FOR BOXES #3 AND #4

ITEM	SIZE IN INCHES
Back	6¼″ × 8¼″ (cut 6[illegible]/8″ × 8[illegible]/8″ for flush routing)
FP	Front panel—5″ × 7″
DP	Damping panel—5″ × 7″
Top	6¼″ × 5[illegible]/8″ (cut 6[illegible]/8″ × 5[illegible]/8″ for flush routing)
Bottom	5″ × 5[illegible]/8″
Sides	See *Fig. 32*
FS	Fill strip—see *Fig. 32*
OC #705	Damping layers—5″ × 7″

Note: Finished size shown. Raw size actually cut in parentheses.

is contained within the box.

When working with the OC #705 material, remember that it is fiberglass-based and take the necessary precautions. You can easily mark it with a dull #2 pencil and cut it accurately with a serrated bread knife.

To securely hold screws driven into the edge, we installed dowels in the sides, top, and bottom wherever a screw for the back was to be installed. Before assembly, mark the screw locations on each panel. Drill these edge locations about 1″ deep with a ¼″ drill. Next, glue ¼″ diameter dowels into the holes and cut them flush with the board edge after the glue has dried. It is not necessary to center the screws into these dowels. If any part of the screw thread is in the dowel they will not strip when tightened.

ASSEMBLY

Assemble these boxes with glue and panel nails—both 1″ and 1[illegible]/8″ lengths. If the longer nail came through into a hole in the FP or DP, then we used the shorter nails. We "set" below flush all nails in from the top or sides so that

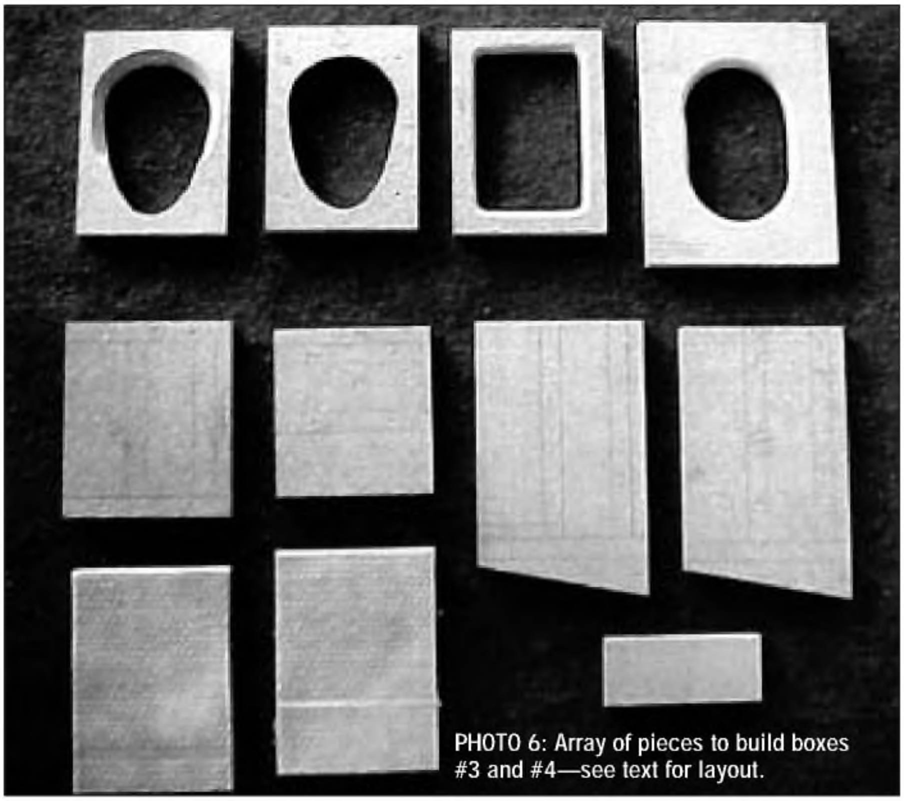


PHOTO 6: Array of pieces to build boxes #3 and #4—see text for layout.

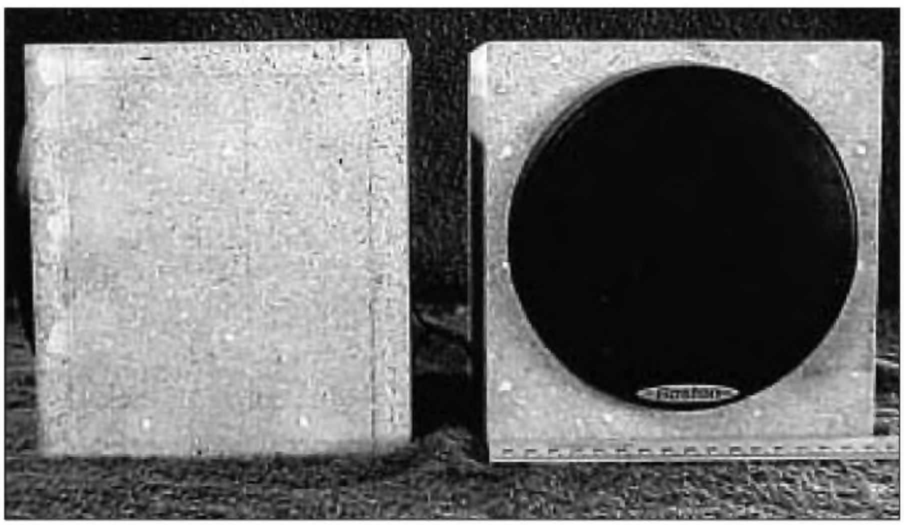


PHOTO 7: Finished boxes #5 and #6.

TABLE 9
A PAIR OF UNMODIFIED #3 AND #4 SPEAKERS

QTY.	COMPONENT
	⅝″ particleboard to build enclosures
2	Infinity Kappa 462.5 compound drivers—they are sold as pairs
20	#8 × 1½″ flat-head wood screws to attach the backs
8	#6 × ¾″ pan-head sheet-metal screws to mount the drivers
20	1″ × ¼″ diameter dowels—inserted to hold back screws
	1″ ELCO panel nails or equivalent for enclosure assembly
	1⅝″ ELCO panel nails or equivalent for enclosure assembly
4	5″ × 7″ pieces of 1″ thick Owens-Corning #705 material
	Grille cloth to cover hole in the back and the box front
	Scraps of Owens-Corning #705 material for grille frame edging
	No. 18 zip or equivalent wire for drivers
	Nominal ½″ thick fiberglass to line enclosures
Misc.	Rubber cement, wood glue, and silicon rubber
	Staples and staple gun to attach grille cloth

they could be filled to finish the boxes. To be sure the nails went in straight, we drilled all the panel faces (but not the board edges) with a #55 bit. If the box uses a fill strip, be sure to drill the necessary nail holes in the sides to install this strip later.

Once you have cut and machined all pieces, clamp the FP and DP to one side in the approximate position, being sure they are flush at the top of the box. Then position the bottom board against the DP and FP and nail it to that side. Use plenty of glue at the interface to be nailed.

Next, glue the FP and DP, properly positioning and nailing them to the bottom and side. Then check the other side to fit properly before gluing and nailing it to the bottom, FP, and DP. Finally, glue and nail the top, cut oversize, into position. After the glue has dried, cut the top to proper size with a flush-cutting router bit.

You must assemble boxes #5 and #6 about the DP only. Once the top is on and sized, install the FP. After the glue has dried, rout the top and side edges of the FP flush with the box.

Clamp the back into position and mark the locations of the retaining screws to match the dowel locations. Then drill small holes (#40 drill) into the back and dowels to take the #8 × 1½″ flathead wood screws. Remove the back and enlarge its holes to pass the screw body (#13 drill) and countersink the back side to permit the screws to go in flush. Then install the back and cut it to final dimensions with the flush-cutting router bit.

If the box requires a fill strip (FS), size and install it at this time. We initially cut the FSs higher than needed with an angle on the bottom edge. This angle made a couple of degrees more than the tip angle of the box so only the front edge touched the floor. After assembling the boxes, slip in the FS and mark it for needed height and then cut the right-angled (top) edge for final fit. This allows correcting for the fact that ⅝″ particleboard is not always exactly 0.625″ thick.

Drill a hole (#2 drill) in the bottom between the FP and DP to pass the speaker wire. We used #18 zip cord sealed with silicon rubber where it goes through the particleboard. On some of the boxes it was necessary to file a notch in the bottom edge of the back to pass this wire coming out of the bottom pedestal. Boxes #9 and #10 require two driver wires (discussed later).

TABLE 10
PIECES FOR BOXES #5 AND #6

ITEM	SIZE IN INCHES
Back	7″ × 7¼″ (cut 7⅞″ × 7⅞″ for flush routing)
FP	Front panel—same as back
DP	Damping panel—5¾″ × 5¾″
Top	7″ × 5¼″ (cut 7⅞″ × 5¼″ for flush routing)
Bottom	5¾″ × 5¼″
Sides	6⅞″ × 5¼″
OC #705	Damping layers—5¾″ × 5¾″

Note: Finished size shown. Raw size actually cut in parentheses.

FINAL FINISHING

You must now decide how to finish your satellites. The nail holes are filled with wood filler and then sanded smooth. We stopped at this point, but we intended to wrap them in stick-on vinyl after final assembly. We planned to staple the grille cloth on the front and miter-cut half-round pieces to cover the staples and finish the grille frame.

For testing purposes we left the boxes uncovered and simply draped grille cloth over the front for listening. This allowed us to observe the drivers for excessive motion. The #5 and #6 boxes, with the car coaxes, use the grilles and frames supplied with the drivers. If you do not plan to wrap your satellites in vinyl, you may choose to rout a radius on the various edges before final finishing.

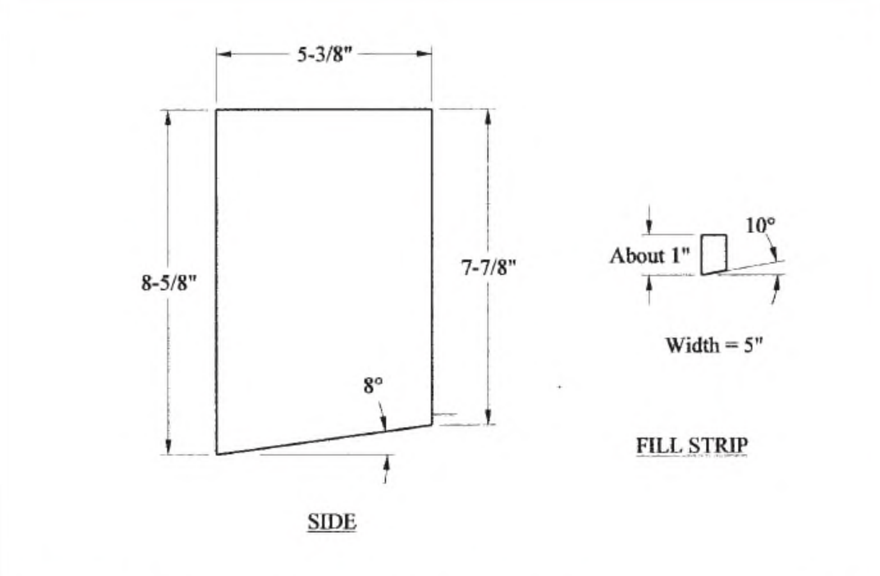


FIGURE 32: Views of side and fill strip for boxes #3 and #4.

We covered the inside walls between the FP and DP with nominal ½″ thick fiberglass insulation material on all boxes. This material was held in place with rubber cement. After this we inserted the damping layers and installed the backs.

Final assembly includes soldering the wires to the drivers and mounting them. Next, cut ½″ × ½″ strips of the OC #705 damping material and insert them around the grille frame, using rubber cement where necessary. If you are not using the OC #705 material, substitute your favorite front panel treatment.

With a temporary grille cloth covering, the satellites are now ready for playing. For the #9 and #10 boxes you may wish to experiment with the coil inductance and, you hope, discover that the selected coil will mount in the pedestal area.

The photos show the layout of the finished panels before assembly of most of the boxes. Note the degree of rounding and relieving used on the various panels. The parts are laid out as follows:
(top row from the left) inside of the front panel, outside of the front panel, the damping panel, and the back;
(second row from the left) the top, bottom, and the two sides; and (bottom row) the fill strip and the two Owens-Corning #705 damping layers, but in no fixed pattern.

BOXES #1 AND #2

Boxes #1 and #2 use the Infinity 462.5CFP compound driver in a side-by-side configuration. With unmodified drivers, we determined these boxes sounded inferior to boxes #3 and #4 using the same driver. We recommend building these boxes only if you plan to make the driver modifications discussed, in which case you may prefer to increase the pedestal height. If you don't choose to modify the drivers, we recommend building boxes #3 and #4. Information to build

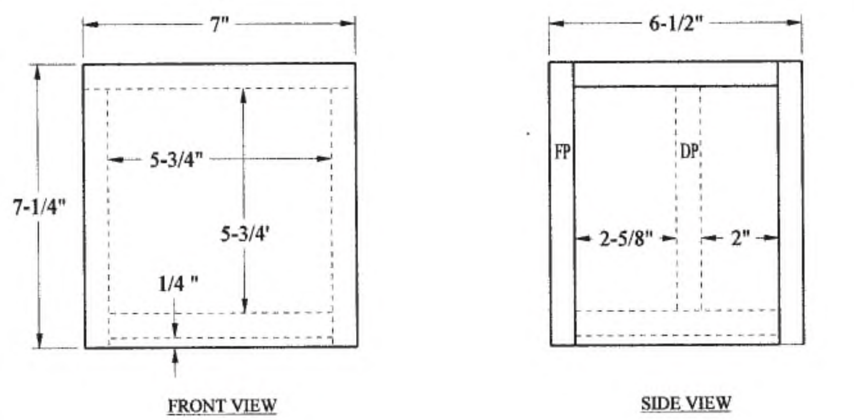


FIGURE 33: Front and side view drawings for boxes #5 and #6.

TABLE 11
PAIR OF #5 AND #6 SPEAKERS

QTY.	COMPONENT
	[illegible]/8″ particleboard to build enclosures
2	Boston Acoustics FX5 coaxial drivers (sold in pairs)
22	#8 × 1½″ flat-head wood screws to attach the backs
8	#8 × ¾″ pan-head sheet-metal screws to mount the drivers
22	1″ × ¼″ diameter dowels—inserted to hold back screws
	1″ ELCO panel nails or equivalent for enclosure assembly
	1-5/8″ ELCO panel nails or equivalent for enclosure assembly
4	5¾″ × 5¾″ pieces of 1″ thick Owens-Corning #705 material
	Grille cloth to cover hole in the back
	No. 18 zip or equivalent wire for drivers
	Nominal ½″ thick fiberglass to line enclosures
Misc.	Rubber cement, wood glue, and silicon rubber
	Staples and staple gun to attach grille cloth

TABLE 12
PIECES FOR BOXES #7 AND #8

ITEM	SIZE IN INCHES
Back	5[illegible]/8″ × 5[illegible]/8″ (cut 5¾″ × 5¾″ for flush routing)
FP	Front panel—4[illegible]/8″ × 4[illegible]/8″
DP	Damping panel—4[illegible]/8″ × 4[illegible]/8″
Top	5[illegible]/8″ × 5¾″ (cut 5¾″ × 5¾″ for flush routing)
Bottom	4[illegible]/8″ × 5¾″
Sides	See *Fig. 36*
FS	Fill strip—see *Fig. 36*
OC #705	Damping layers—4[illegible]/8″ × 4[illegible]/8″

Note: Finished size shown. Raw size actually cut in parentheses.

PHOTO 8: Rear view for boxes #5 and #6 showing front panel relieving.

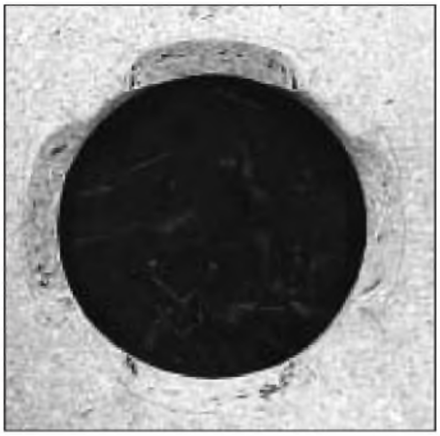


PHOTO 9: Typical front panel relieving used for four-strut driver.

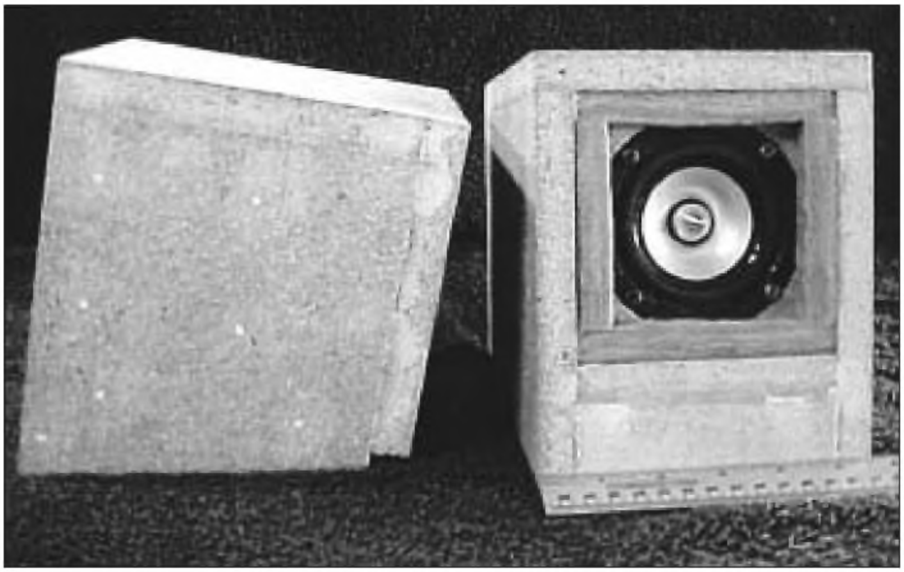


PHOTO 10: Finished boxes #7 and #8.

boxes #1 and #2 is contained in the following:

Photo of finished boxes: *Photo 2*
Front and side view drawings: *Fig. 28*
Views of side board and fill strip: *Fig. 29*
Damping panel and back drawings: *Fig. 30*
Sizes of pieces used in construction: *Table 6*
Layout of pieces before assembly: *Photo 3*
List of materials to build a pair of satellites: *Table 7*
Shape of driver mounting hole and relieving: *Photo 4*

No front panel layout is shown, because the Infinity plate driver does not mount in a simple round hole. You should cut a hole in a cardboard template until it fits down on the back of the plate while clearing the drivers and crossover parts. Use this template to draw the required hole on the front panel, positioning the hole so that the plate is centered on the FP leaving a distance of about ½″ to all edges. The back side of this hole must be heavily relieved in the woofer area because this plate driver was designed to mount in a thin car panel (*Photo 4*).

We mounted the plates dry with #6 × ¾″ pan-head sheet-metal screws into holes drilled (#46 bit) into the FP. If a rattle or leak problem developed, our plan was to seal the plate in with silicon rubber, but this was not necessary. Other construction is as covered in the general section. For unmodified drivers the wider input terminal is the plus input. The section on modifying these drivers identifies the individual woofer and tweeter terminals.

BOXES #3 AND #4

Boxes #3 and #4 also use the Infinity 462.5CFP compound, and you can

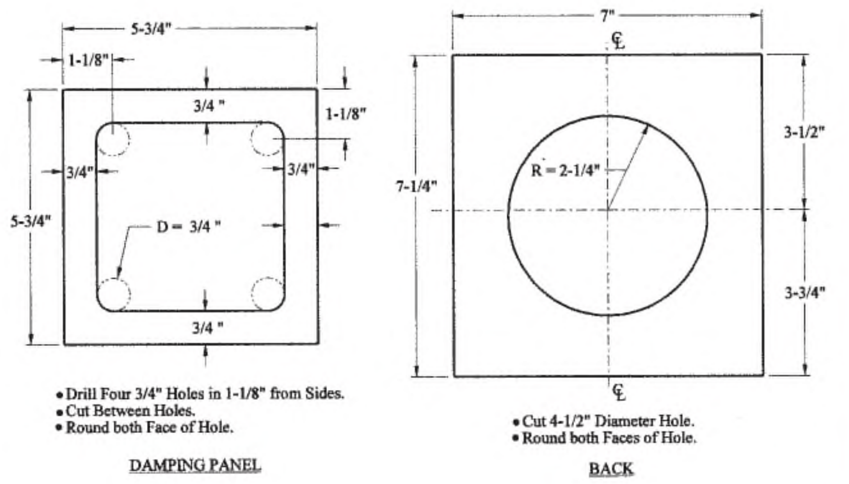


FIGURE 34: Views of damping panel and back for boxes #5 and #6. B-2214-34

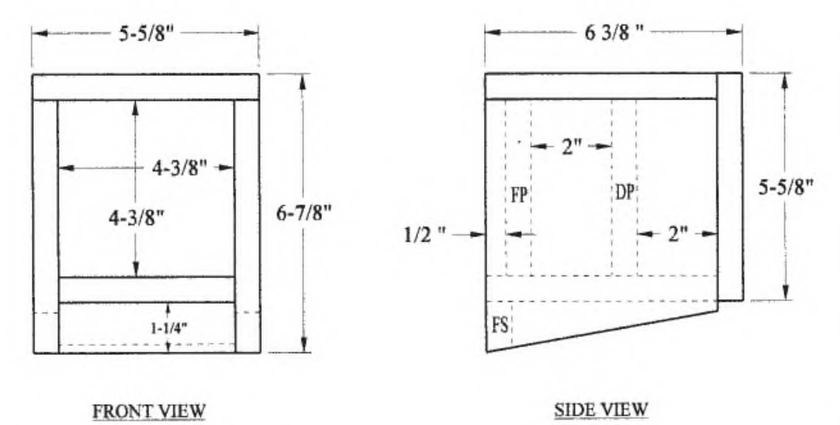


FIGURE 35: Front and side view drawings for boxes #7 and #8. B-2214-35

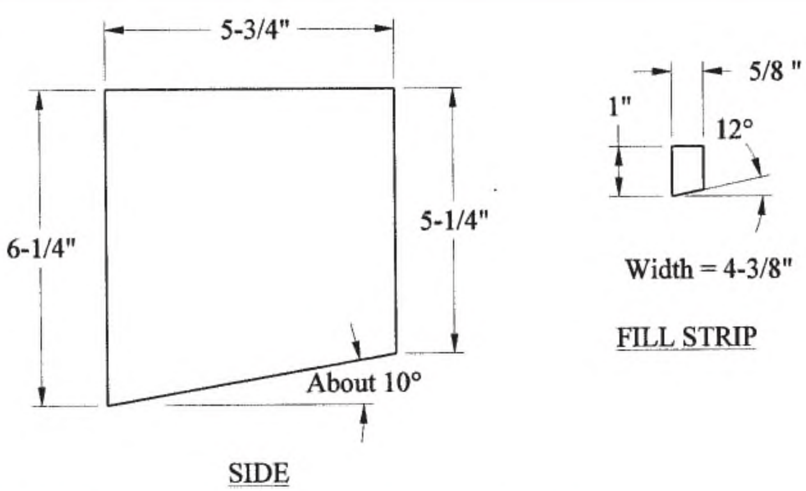


FIGURE 36: Views of side and fill strip for boxes #7 and #8. B-2214-36

build them with the drivers in stock form or modified as described earlier. If you are going to modify the drivers, you may choose to increase the pedestal height to allow more room for crossover components. See the following for detailed information:

Photo of finished boxes: *Photo 5*
Front and side view drawings: *Fig. 31*
Views of side board and fill strip: *Fig. 32*
Damping panel and back drawings: *Fig. 30* (same as for boxes #1 and #2)
Sizes of pieces used in construction: *Table 8*
Layout of pieces before assembly: *Photo 6*
List of materials to build a pair of satellites: *Table 9*
Shape of driver mounting hole and relieving: *Photo 4*

The damping panel and back are identical to those for boxes #1 and #2, just rotated 90°. Sharp-eyed readers may notice that these figures indicate a designed front panel tip angle of 8°, while Table 4 showed the finished satellites had a tip angle of 10°. This was caused by a measurement error while laying out the side boards, a clear demonstration that you can vary the tip angle with no change in the basic box structure. The front panel construction is the same as for boxes #1 and #2. Other construction is the same as shown in the general section and for boxes #1 and #2.

BOXES #5 AND #6

Boxes #5 and #6 use the Boston Acoustics FX5 car coax. You can find the required building information in the following:

Photo of finished boxes: *Photo 7*
Front and side view drawings: *Fig. 33*
Damping panel and back drawings: *Fig. 34*
Sizes of pieces used in construction: *Table 10*
List of materials to build a pair of satellites: *Table 11*

The front panel requires only a round hole for the driver, its center down 3½″ from the top of the finished FP on the vertical centerline. We forgot to record the hole size used, but it is about 4½″. Do measure to be sure.

Also, relieve the back of the FP between the driver struts. *Photo 8* shows a rear view of the box, bottom to the right, showing the damping panel and front panel relief. The inserted dowels to retain the back mounting screws are evident in this photo. See also the typical relief used for a four-strut driver on the back of a test baffle (*Photo 9*) for clarity.

Unlike all the other box types, these have the FP mounted on the front of the box rather than inset. The FP also extends down to the floor so no fill strip is needed. As discussed earlier, the top is oversize and routed to be flush. We installed the oversize FP and routed it flush, and temporarily installed the oversize back and routed it flush.

This simple box takes more work than we initially thought it would, but does not require you to construct a grille frame. If you plan to paint the finished satellites you may choose to rout a radius on the external edges for a more contemporary look.

Remaining construction is as discussed in the general section. We mounted the drivers with a thin coat of silicon rubber on the front of the FP only. If you need to remove the driver, the silicon rubber is accessible to be cut with a thin blade. Note that the drivers are mounted in the boxes upside down with the crossover capacitor and word "Boston" at the top to get the desired listening angle.

The grille frame mounts with the same screws as the driver and is installed right side up. We used #8 × ¾″ pan head sheet metal screws in holes (#42 bit) drilled into the FP. You then simply press the metal grille structure into the frame. For wiring, the wider input terminal is the plus terminal.

BOXES #7 AND #8

These boxes use a single TBspeakers W3-871s driver from NUERA Acoustic Technology. Sources of information to build the boxes are as follows:

Photo of finished boxes: *Photo 10*
Front and side view drawings: *Fig. 35*
Views of side board and fill strip: *Fig. 36*
Details of driver mounting hole: *Fig. 37*

TABLE 13
PAIR OF #7 AND #8 SPEAKERS

QTY.	COMPONENT
	[illegible]/8″ particleboard to build enclosures
2	TBspeakers W3-871s nominal 3″ shielded drivers
16	#8 × 1½″ flat-head wood screws to attach the backs
8	#6 × ½″ pan-head sheet-metal screws to mount the drivers
16	1″ × ¼″ diameter dowels—inserted to hold back screws
	1″ ELCO panel nails or equivalent for enclosure assembly
	1[illegible]/8″ ELCO panel nails or equivalent for enclosure assembly
4	4[illegible]/8″ × 4[illegible]/8″ pieces of 1″ thick Owens-Corning #705 material
	Grille cloth to cover hole in the back and the box front
	Scraps of Owens-Corning #705 material for grille frame edging
	No. 18 zip or equivalent wire for drivers
	Nominal ½″ thick fiberglass to line enclosures
Misc.	Rubber cement, wood glue, and silicon rubber
	Staples and staple gun to attach grille cloth
	Rope putty or equivalent to seal/damp driver rims

TABLE 14
PIECES FOR BOXES #9 AND #10

ITEM	SIZE IN INCHES
Back	5[illegible]/8″ × 9¾″ (cut 5¾″ × 9[illegible]/8″ for later flush routing)
FP	Front panel—4[illegible]/8″ × 8½″
DP	Damping panel—4[illegible]/8″ × 8½″
Top	5[illegible]/8″ × 5¾″ (cut 5¾″ × 5¾″ for later flush routing)
Bottom	4[illegible]/8″ × 5¾″
Sides	See *Fig. 40*
FS	Fill strip—see *Fig. 40*
OC #705	Damping layers—4[illegible]/8″ × 8½″

Note: Finished size shown. Raw size actually cut in parentheses.

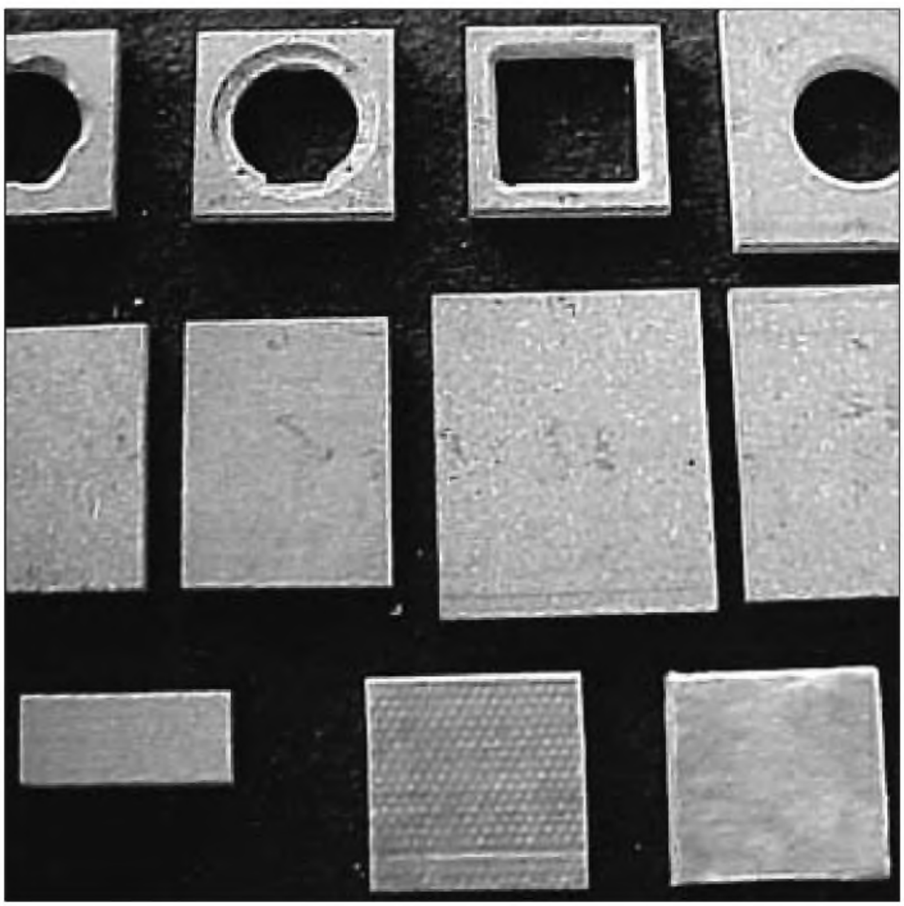


PHOTO 11: Array of pieces to build boxes #7 and #8—see text for layout.

Damping panel and back drawings: *Fig. 38*
Sizes of pieces used in construction: *Table 12*
Layout of pieces before assembly: *Photo 11*
List of materials to build a pair of satellites: *Table 13*
Shape of driver mounting hole: *Photo 12*
Relief on rear of front panel: *Photo 13*

These drivers must be flush-mounted (*Fig. 37* shows details of the rabbeting we used). Not all rabbeting bits cut the same amount, so you need to size your through hole such that your [illegible]/16″ rabbeting bit will produce the outside diameter shown. You want this rabbeted hole diameter to be accurate because testing indicates having a gap around the outside of the frame rim can cause as much diffraction as surface-mounting the driver. The driver mounting hole also requires a filed notch to clear the terminal strip mounted on the driver.

This driver is a bit strange, having four mounting holes but six struts from rim to magnet. For a nominal 3″ driver it is heavy because of the magnet shield. You need to relieve the back of the FP in six places between the struts while not making the particleboard too thin at the mounting holes—a bit of a compromise. For clarity see the front of the driver hole (*Photo 12*) and the relieving on the back (*Photo 13*) for the test baffle for this driver. Two sets of driver mounting screw holes appear, because this baffle mounts multiple TBspeakers nominal 3″ driver types.

The remaining construction is as shown in the general section. We installed the drivers with #6 × ½″ pan-head sheet-metal screws in holes (#50 bit) drilled in the FP. We have had earlier problems with these pressed metal frame rims "ringing," so we fitted the back of them with rope putty that presses slightly against the FP to form a seal. The wiring polarity for these drivers is marked on the terminal strip.

BOXES #9 AND #10

These boxes each use two of the TBspeakers W3-871s drivers. Refer to

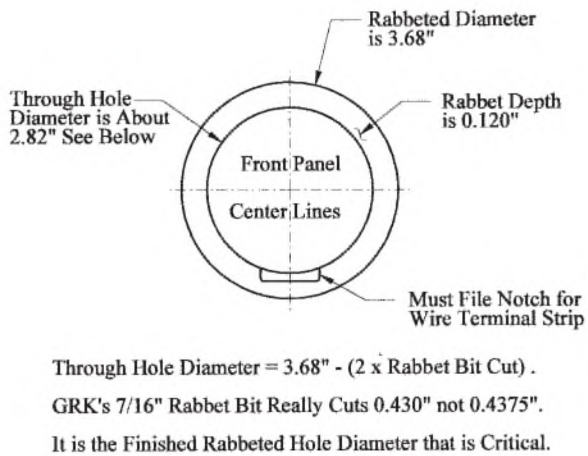


FIGURE 37: Details of mounting hole for W3-871s driver. B-2214-37

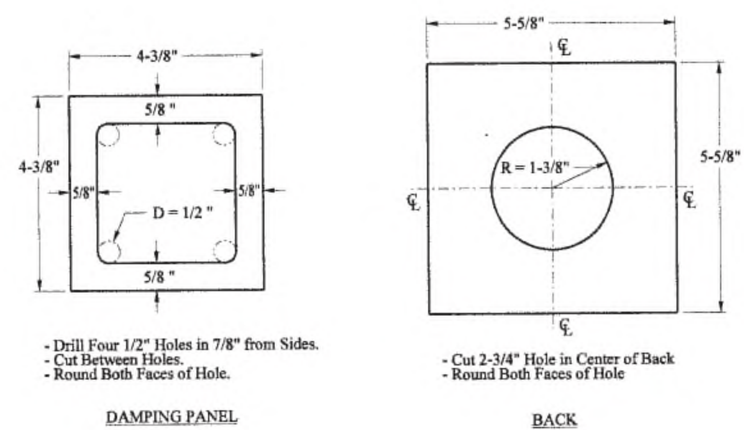


FIGURE 38: Views of damping panel and back for boxes #7 and #8. B-2214-38

the following for the information you need to build them:

Photo of finished boxes: *Photo 14*
Front and side view drawings: *Fig. 39*
Views of side board and fill strip: *Fig. 40*
Details of driver mounting hole: *Fig. 37* (same as boxes #7 and #8)

PHOTO 12: Front of hole to mount W3-871s driver.

PHOTO 13: Relieving of front panel rear for W3-871s driver.

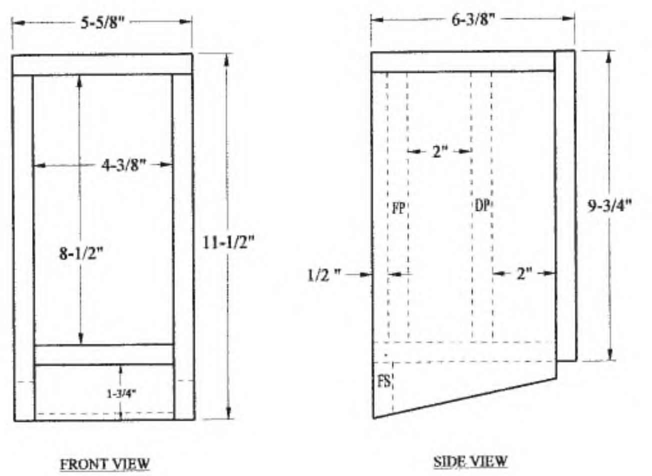


FIGURE 39: Front and side view drawings for boxes #9 and #10.

B-2214-39

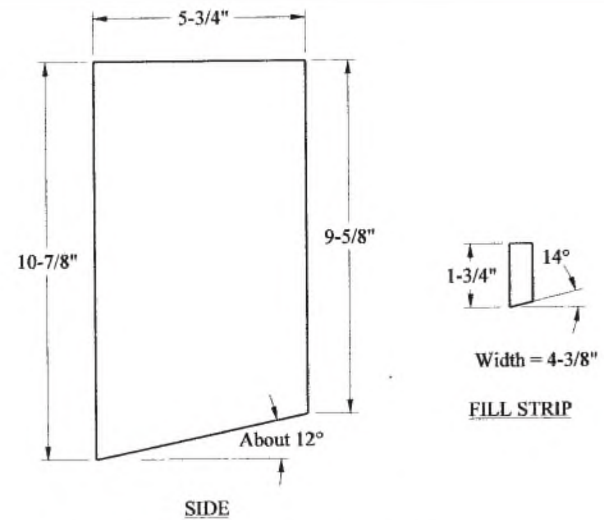


FIGURE 40: Views of side and fill strip for boxes #9 and #10.

B-2214-40

Damping panel and back drawings: *Fig. 41*
Sizes of pieces used in construction: *Table 14*
Layout of pieces before assembly: *Photo 15*
List of materials to build a pair of satellites: *Table 15*
Shape of driver mounting hole: *Photo 12*
Relieving on rear of front panel: *Photo 13*

These boxes are simply a taller version of boxes #7 and #8. If you know the size of the coil you are going to use, check to see whether it will fit in the present pedestal structure. If not, you can extend the sides and fill strip for a larger pedestal. We selected the 2.5mH coil—an iron bar-core unit (#266-556 from Parts Express)—which just fits under the pedestal shown by cutting corners off the plastic coil former.

The basic locations of the two holes in the front panel are indicated in *Fig. 41*. Refer to *Fig. 37* and *Photos 12* and *13* for details of the holes needed to mount each driver. We faced the terminal strip toward the FP center for each driver.

Construction is the same as for boxes #7 and #8 except for those items pertaining to the dual drivers. The FP and back have two holes. Since the damping layers were becoming rather long, the DP also has two holes.

When you make the DP, do not radius the front side of the center rib. Cut a piece of ¾″ diameter dowel lengthwise to form stiffeners to glue on the front side of the DP rib (*Photo 15*). We initially made the dowel too long and after gluing it to the DP, we cut it flush with the edges of the DP.

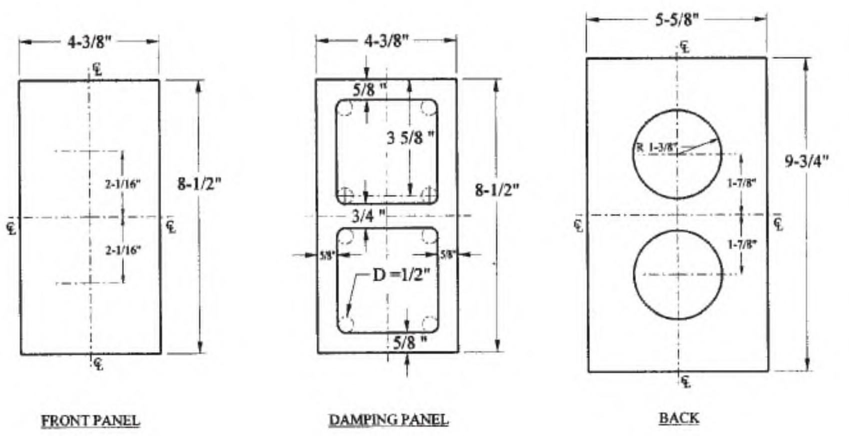
Remaining construction is as covered in the general section and for boxes #7 and #8. Note that two driver wire holes are required in the bottom, so you must position them where they do not interfere with mounting the coil.

SUMMARY

This article covers the development and construction of five small, simple satellite types. They are simple because they do not require you to construct a complex crossover and they have a low driver count. We developed all prototypes to be floor-standing, but stand mounting is possible with some designs. Two of the best-sounding satellite types resulted from major crossover redesign on the Infinity Kappa 462.5CFP drivers. These satellites are thus not simple or low cost, but are good-sounding small boxes.

Box types #1 and #2 use the Infinity Kappa 462.5CFP compound driver mounted with the drivers side-by-side. In unmodified form they are acoustically inferior to box types #3 and #4. With the added complication of a new crossover design, these boxes are fine-sounding satellites that you could use on stands. These drivers are not shielded. You should not use these satellites, which are nominal 4Ω systems, below 100Hz.

Box types #3 and #4 use the Infinity Kappa 462.5CFP compound driver mounted with the woofer above the tweeter. This design is a loud-playing, high-presence satellite. The basic response has some diffraction spreading loss compensation for floor-standing



Front panel - Make Two Driver Holes in Location Shown - See Text.

Damping Panel - Drill Four 1/2 Diameter Holes in 7/8" Edges.
- Drill Four more 1/2" Holes in 7/8" From Sides & 3-5/8" From Top/Bottom.
- Saw Between Holes.
- Round All Edges of Holes Except Front of 3/4" Center Rib.

Back - Cut Two 2-3/4" Diameter Holes at Locations shown.
- Round Both Faces of Holes.

FIGURE 41: Views of damping panel, back and front panel for boxes #9 and #10.

B-2214-41

application, but this is not a candidate for stand mounting. With the added complication of a new crossover, these are delightful-sounding satellites that you could use on stands. These drivers are not shielded. You should not use these satellites, which are nominal 4Ω systems, below 100Hz.

Box types #5 and #6 use the Boston Acoustics FX5 car coax. This simple box has the driver mounted on the front using the frame and grille supplied with the driver. These units will play very loudly, but the sound is high presence and very forward. Aiming the boxes slightly away from the listener corrects this to some degree. These boxes need to be on the floor, probably near the back wall, and are candidates for a floor-standing bookcase, but not for stand mounting. These drivers are not shielded. You could probably use these satellites, which are nominal 4Ω systems, to somewhat below 100Hz.

Box types #7 and #8 use the TBspeakers nominal 3″ full-range driver. These tiny boxes suffer badly from diffraction spreading loss. They are only acceptable if you use them near the floor/back-wall interface, never on stands out in the room. Even with a limited power rating and low sensitivity, they will play at reasonable volume levels with the proper crossover. They are fully shielded drivers that allow you to build a system with no electrical crossover above about 100Hz. Do not try to use these satellites, which are nominal 8Ω systems, below 100Hz.

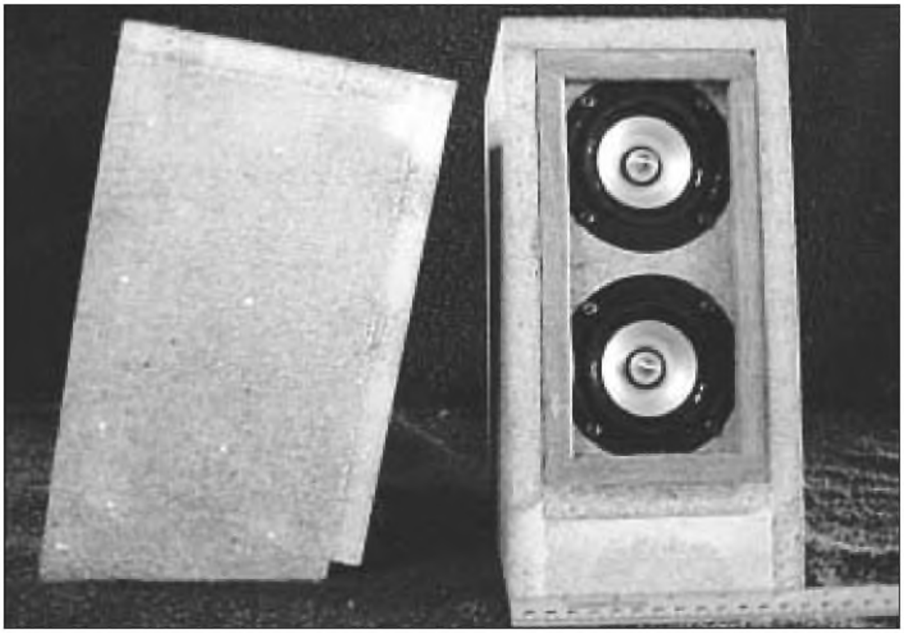


PHOTO 14: Finished boxes #9 and #10.

Box types #9 and #10 each use two of the TBspeakers nominal 3″ full-range drivers. The top driver is used full-range; the bottom driver only for the purpose of diffraction spreading loss compensation. This requires the use of a single crossover component, a coil.

Other than the satellites using modified Infinity drivers, these are the best satellites we developed, sounding almost bass heavy when on the floor and good candidates for stand mounting, but probably not for use in a bookcase. They have a very pleasant, musical sound and will play at acceptable levels in a two-channel stereo application with the proper crossover. The drivers are fully shielded.

You should not use these boxes below 100Hz as satellites with a subwoofer. However, for low-playing-level applications, such as a near-field computer speaker, they sound very good

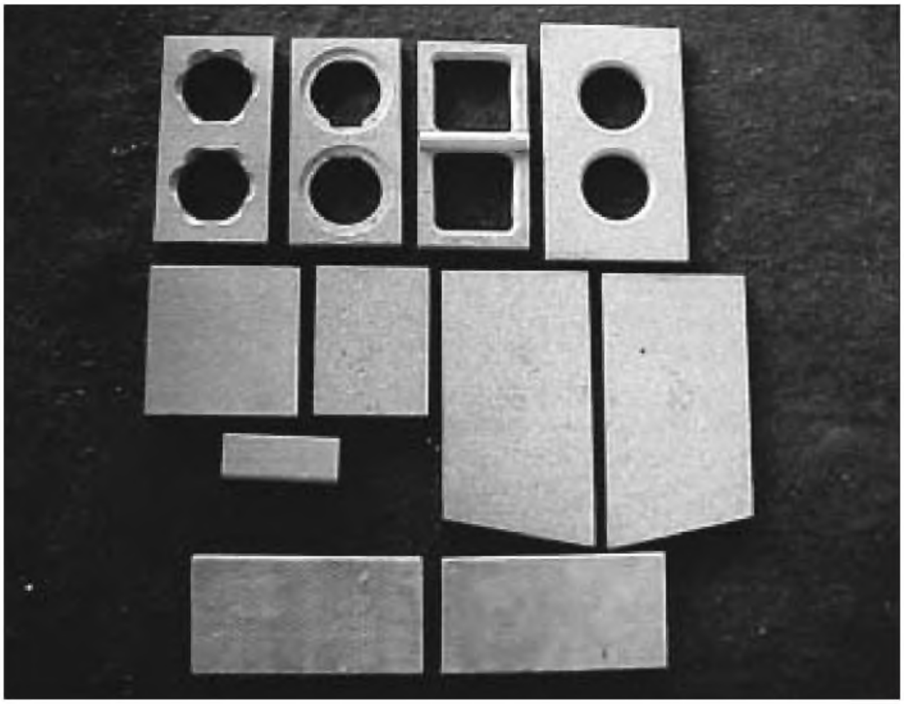


PHOTO 15: Array of pieces to build boxes #9 and #10—see text for layout.

SOURCES

Boston Acoustics, Inc.
300 Jubilee Drive
Peabody, MA 01960
978-538-5000
www.bostonacoustics.com

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15 rue J.C. Verpilleux
B.P. 201
42013 Saint-Etienne cedex 2, France
(33) 04 77 43 16 16
www.focal.tm.fr

Infinity Systems, Inc.
250 Crossways Park Drive
Woodbury, NY 11797
800-553-3332
www.infinitysystems.com

NUERA Acoustic Technology
6538 Albery Place
Burnaby BC
Canada, V5E-4G2
604-517-6683
Mr. Billy Lau
www.nuera-acoustic.ca

Owens Corning
For information on obtaining the #705 damping material, call the Customer Service Center: 800-328-7617

TABLE 15
PAIR OF #9 AND #10 SPEAKERS

QTY.	COMPONENT
	5/8″ particleboard to build enclosures
4	TBspeakers W3-871s nominal 3″ shielded drivers
20	#8 × 1½″ flat-head wood screws to attach the backs
16	#6 × ½″ pan-head sheet-metal screws to mount the drivers
20	1″ × ¼″ diameter dowels—inserted to hold back screws
	1″ ELCO panel nails or equivalent for enclosure assembly
	1⅛″ ELCO panel nails or equivalent for enclosure assembly
4	4⅛″ × 8½″ pieces of 1″ thick Owens-Corning #705 material
	Grille cloth to cover hole in the back and the box front
	Scraps of Owens-Corning #705 material for grille frame edging
	No. 18 zip or equivalent wire for drivers
	Nominal ½″ thick fiberglass to line enclosures
Misc.	Rubber cement, wood glue, and silicon rubber
	Staples and staple gun to attach grille cloth
	Rope putty or equivalent to seal/damp driver rims
2	2.5mH coils—we used Parts Express #266-556
6″	¾″ diameter dowel—split to brace damping panel

used full range without a subwoofer. Due to the dual parallel drivers at low frequency, these satellites are nominal 4Ω systems.

How loudly a small satellite can play is very much a function of how you perform the crossover from the subwoofer to the satellite. Using the passive summer/crossover/equalizer developed for our subwoofer[1], the satellites see the full signal bandwidth, which greatly reduces their maximum playing level. We deemed this approach acceptable only for background music listening.

We found the best approach was a true fourth-order crossover on the satellites and the subwoofer. With this approach, each satellite needs only a few watts, while drums hammer away and the satellite cones show small displacements. The crossover frequency was 110Hz for all satellite types.

Although we didn't use the units with a multi-channel system, these satellites could be used in a 5.1-, 6.1-, or 7.1-channel system. Remember, only boxes #7–#10 use shielded drivers, which allow their use as a center speaker. Using these requires that your multi-channel receiver/amplifier provide the proper crossover functions to limit the low-frequency content fed to all the satellites. Much of the available equipment will allow you to divert all low-frequency material to the subwoofer (LFE channel), as these small satellites would require. Playing at a reduced level, examine the cones for all satellites to verify they are not dancing around due to receiving low-frequency content.

All the satellites used the infinite box technique. While not evaluated, these same drivers should perform just fine in small closed boxes. Unless you are using a high-slope crossover, vented boxes would be dangerous with these drivers.

We learned that small drivers that are truly full range, down to about 100Hz, are now appearing on the market. This means that simple satellites not requiring a crossover are now possible. With proper design, the future for small satellites looks promising.

We found diffraction spreading loss (for more on this topic and some tips on how to handle the DSL problem, see our follow-up letter in next month's issue of *aX*) to be an important design factor in producing small satellites that yielded a good-sounding system when used with a subwoofer. Ignore it at your own peril. This result has brought to mind new ideas, and we are already at work on testing the design of new, improved satellite types.

All acoustic testing for this article was performed using Audiosuite by Liberty Instruments, Inc. Their current testing system for the Windows operating system is called Praxis. ❖

A MOSFET Update of the Forte 1a Amplifier, Part 2

A simplified version of the Pass/Thagard A75 serves as an update for the Forte 1a amplifier. **By Joe Berry**

Installing the Forte 1b update involves removing and stripping each heatsink assembly, building the new PC board assembly, performing initial tests and adjustments, and reassembling each channel with final adjustments. It is also possible to modify one channel at a time and evaluate the difference between the 1a and 1b using a monophonic source.

You need only ordinary tools, but you must work carefully to avoid cosmetic and mechanical damage to the amplifier's aluminum metalwork. Be sure to use appropriate precautions against static discharge when handling MOSFETs out-of-circuit. Store them in conductive containers or conductive foam until ready for use, and do not remove them until you have grounded yourself.

1) Before opening the amplifier, turn it off and unplug the power cord from the back of the unit.

2) Remove the top cover and locate the power switch and LED wiring leading to the inside front panel. (Double-check voltages on the power supply caps.—Ed.) Detach the wiring and remove the front panel.

3) On each channel to be removed, unfasten the + (red) and – (blue) DC power leads, ground lead (black), and the loudspeaker output lead (orange) from their connection points inside the chassis, and unsolder the input signal cable from its RCA jack on the amplifier rear panel. Leave all cables attached to the PC board.

4) Remove the screws holding the channel's heatsink assembly to the bottom and rear of the amplifier chassis, then remove the assembly from the chassis (*Photo 2*).

STRIPPING THE HEATSINK

1) Remove the screws holding the ten plastic power transistors, two TO-220 driver transistors, and thermal cutout to the heatsink. Carefully lift the PC board away from the heatsink (*Photo 3*).

2) If desired, invert the PC board and use a well-heated medium-duty soldering iron to remove the input, power, output, and ground leads and the thermal cutout from the old PC board for use with the new board. Alternatively, you may leave the original leads attached to the old board and prepare new cabling. Store the original PC board safely, and set all mounting hardware aside for reuse.

3) Clean the heatsink mounting surface thoroughly to remove all insulating washers, thermal compound residue, and accumulated grit and grime. Naptha (lighter fluid) is an effective solvent for this purpose, but is highly flammable and must be used with caution.

INSTALLING NEW BOARDS

1) If you have not yet done so, partially assemble the new PC boards by first installing all passive components (resistors, capacitors, variable resistors, diodes, cables, and thermal cutout). Double-check that all diodes and electrolytic capacitors are installed with the correct orientation. Next, install MOSFETs Q1–Q6 and Q9. Remember that Q3–Q4 and Q5–Q6 should be matched pairs. Although Q1, Q2, and Q9 need not be matched, they should be tested for proper function before they are installed.

2) If you have not yet done so, test MOSFETs Q7 and Q8 for proper function. Then test and match N-channel output MOSFETs Q10, Q12, Q14, Q16, and Q18 as well as P-channel outputs Q11, Q13, Q15, Q17, and Q19. Bend the leads of Q7 and Q8 at right angles at the point where the leads widen so that they point *toward* the device mounting surface. Bend the leads of output MOSFETs Q10–Q19 at right angles at the point where the leads widen so that they point *away* from the device mounting surface.

3) Temporarily fasten Q10–Q19 in place on the heatsink without thermal pads. Slip the new PC board over the

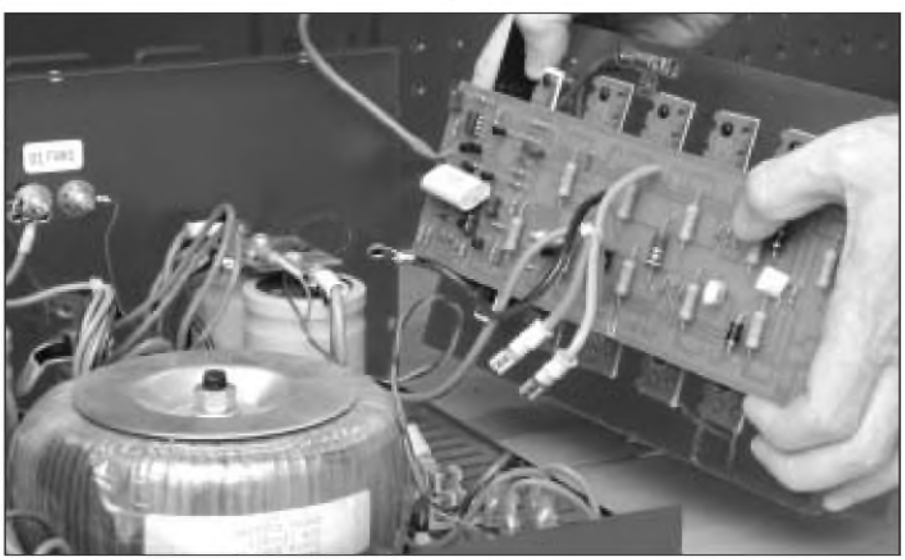

PHOTO 2: Removing a heatsink assembly.

pins of the output MOSFETs, using the board as a guide to align the MOSFETs.

4) Temporarily mount Q7 and Q8 in place on the heatsink standoffs without thermal pads. When installing Q7 and Q8, carefully guide their pins into their holes in the PC board. Note that the pins of Q7 and Q8 enter from the top side of the board, whereas the pins of Q10–Q19 enter from the bottom side of the board.

5) Use a ruler or spacers to position the PC board level at ¼″ (6.5mm) from the heatsink mounting surface. Solder the pins of Q18 and Q19 to their pads on top of the board to fix the height at one end. Next, solder the pins Q7 and Q8 to their pads on top of the board to fix the height at the opposite end. Finally, solder the pins of Q10–Q17 to their pads on top of the board.

6) Unfasten Q7, Q8, and Q10–Q19 from the heatsink and remove the PC board. Invert the board and solder Q7, Q8, and Q10–Q19 to their pads on the bottom of the board. Clean the PC board and check all connections and solder joints. Confirm that there are no unwanted solder bridges between pads. Recheck the heatsink mounting surface for any grit or residue that could prevent good thermal contact.

7) Reattach the PC board to the heatsink (*Photo 4*), this time using thermally conductive insulating pads under Q7, Q8, and Q10–Q19. Also use nylon shoulder washers to insulate the metal tabs of Q7 and Q8 from their mounting

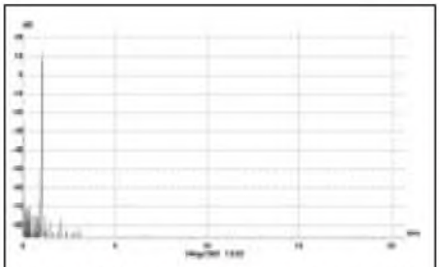

FIGURE 6A: F1a distortion spectrum, 1W @ 1kHz. A-2249-6A

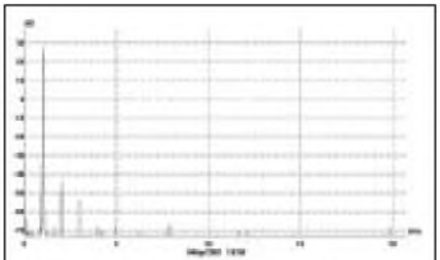

FIGURE 6B: F1a distortion spectrum, 50W @ 1kHz. A-2249-6B

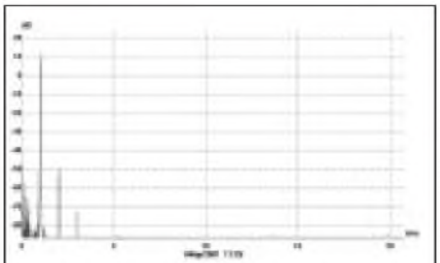

FIGURE 6C: F1b channel A distortion, 1W @ 1kHz. A-2249-6C

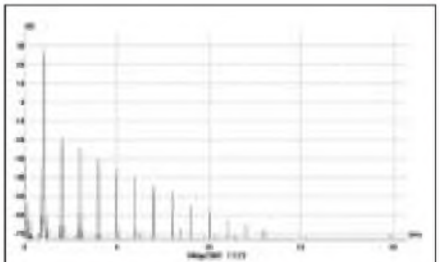

FIGURE 6D: F1b channel A distortion, 50W @ 1kHz. A-2249-6D

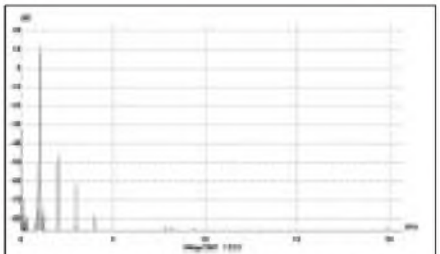

FIGURE 6E: F1b channel B distortion, 1W @ 1kHz. A-2249-6E

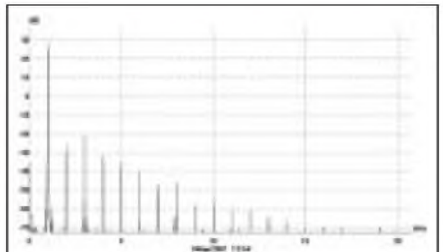

FIGURE 6F: F1b channel B distortion, 50W @ 1kHz. A-2249-6F

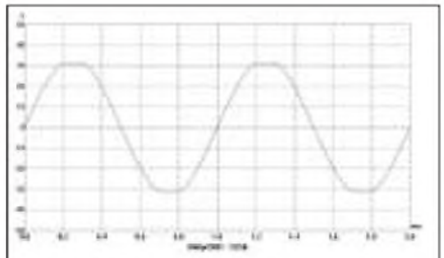

FIGURE 7: Forte 1b clipping waveform, 1kHz. A-2249-7

screws. If available, use conical washers between the TO-247 devices and the heads of their mounting screws. Tighten all screws securely, but do not overtighten because you will risk stripping the threads tapped into the soft aluminum heatsink. Use just enough torque to fully compress each washer.

INITIAL TESTING AND ADJUSTMENT

Recommended practice is to test the finished channels individually by temporarily removing the fuses from the other channel's DC power supply so that it will not draw power. Temporarily replace both DC power-supply fuses in the channel under test with 3A fast-blow fuses. Use an isolated variable AC power source if available, and remember that the AC line is lethal—keep your hands away from any exposed AC wiring!

1) Temporarily reattach the channel not being tested to the amplifier chassis. Reattach the amplifier's front panel and reconnect its AC power switch and LED wiring.

2) Leave the channel under test detached from the amplifier chassis, and use jumper wires to attach its power and ground leads to their connection points inside the amplifier (*Photo 5*). Double-check the DC power-supply connections to ensure correct polarity. Be sure to jumper the input stage ground connection to the ground plate across the channel's main power-supply capacitors; the channel will not work without this connection.

3) Set VR1, VR2, and VR3 initially to their minimum (full counterclockwise) positions. Apply AC power to the amplifier and immediately check for correct DC power-supply voltages (±36V) at the new PC board. If either voltage is absent or if you see smoke, power down immediately, check DC fuses, and recheck the orientation of all polarized components. A blown fuse signals the need for troubleshooting. Be sure that VR3 was not inadvertently turned fully clockwise.

4) If the board has DC voltage and does not smoke, check for about .2V across R19 and R30 to confirm that the input stage and its current sources are working and that the second stage is shut down. If the reading is higher, recheck VR1 and VR2, which may be set fully clockwise by mistake. If the reading is lower, work backward toward Q1/Q2 to find the problem. For example, ZD1 or ZD2 may be installed backwards, or thermal cutout T1 may be defective (open).

5) If the voltage across R19 and R30 checks out, gradually rotate VR1 and VR2 clockwise to power up the second stage. As the voltages across R19 and R30 start to increase, try to keep them roughly equal as you iterate toward a reading of 1V across both resistors. Once you obtain these values, fine-tune them to keep the DC voltage across R25 below 50mV. This step is best done gradually over the course of an hour to allow for the effects of warmup.

6) After successfully adjusting VR1–VR2, slowly rotate VR3 clockwise while looking for about 100mV DC

BUILDING FROM SCRATCH

If this project interests you but don't have a spare Forte 1a, you might consider building the complete amplifier by providing your own enclosure, power supply, and associated hardware.

Two European companies, Fischer and ELCAL, manufacture aluminum enclosures with integral side-mounted heatsinks that would make good platforms for such a project. These enclosures are offered by Erno Borbely (www.borbelyaudio.com) for use with his DC-50 Home Theater Amplifier kits. The smallest unit, the ELCAL "Kraftwerk" 400, appears to meet the Forte 1b's minimum thermal dissipation requirement of 0.25°C/W per channel. Mr. Borbely advises that the mounting points on the heatsinks for all these enclosures may need to be reinforced with extra screws or aluminum bars when using heavy transformers.

You will need to drill the enclosure to match your choice of AC inlet, power switch, signal input jacks, output binding posts, and power-supply components. The drill pattern for the heatsink is determined by the PC board layout; *Fig. A* shows the recommended pattern. This is similar to the drill pattern used in the Forte 1a, except that the 4-40 threads for the Q7 and Q8 mounting screws are tapped into the mounting surface, rather than into standoffs. Q7 and Q8 now mount directly to the heatsink, so you would bend their pins in the same direction as those for output MOSFETs Q10–Q19 and insert them from the back of the PC board.

Two mounting holes are provided for the thermal cutout; use whichever hole places the switch closest to the top of the enclosure. The original thermal cutout was sleeved in plastic tubing and held in place by a metal clip. I have not found an exact replacement, but the thermal cutouts made by Cantherm and sold by Digi-Key seem to be a good alternative, and have a hole that allows them to mount directly to the heatsink without a clip or insulating sleeve. Another option is the AirPax line of TO-220 style thermal cutouts, sold by Newark Electronics.

The Forte 1a's power supply was based on a custom toroidal power transformer having separate center-tapped secondaries for each channel. Such transformers are hard to find off-the-shelf, but an excellent work-around is to adapt the A75's dual power-supply design, which uses the much more common transformer type with a single pair of secondaries. To adapt the A75 power supply for this project, simply replace its 2 × 30V @ 550VA transformer with a 2 × 25V @ 500VA unit and leave out the front end voltage doubler/regulator circuits.

If you take this approach, also be sure to use the A75 grounding scheme, which brings the input, output, and power-supply grounds of both channels together at a single "star ground" point. This star ground is then tied to chassis ground through a 5.1Ω 1W resistor that is bypassed above ±.6V by two 1N5401 diodes. If you prefer, you can replace the resistor and diodes with a single Keystone CL-60 inrush current limiter, as used by Nelson Pass (who attributes the idea to Frank DeLuca[6]) in some of his later designs.

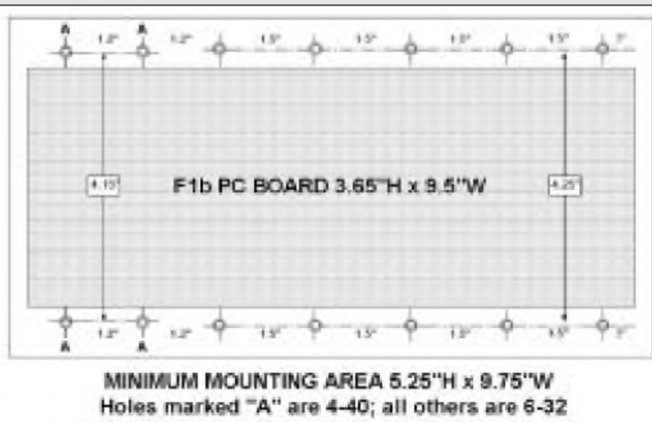

FIGURE A: Forte 1b recommended heatsink drill pattern.

A-2249-A

across each .47Ω output stage source resistor. Confirm that the voltages across these resistors are within 20% of each other, indicating good matching and current sharing among the output devices. Make note of which resistor reads closest to the group average, and use that resistor for subsequent adjustments. This step is also best performed gradually over the course of an hour, during which the heatsink should become hot but not quite untouchable.

7) Check the DC voltage across the output leads and tweak VR1 and VR2 as needed to bring this voltage to near zero while maintaining 1V across R19 and R30. Again, this adjustment is best made over the course of an hour or so to allow for warmup. There may be a small interaction with the bias adjustment, so re-check the output stage bias periodically and readjust VR3 as needed. At the end of the initial adjustment period, the DC voltage across R19 and R30 should be at 1V, the average DC voltage across each output stage source resistor should be 100mV, and the DC offset voltage at the amplifier output should be less than 50mV.

8) Disconnect the channel just tested and repeat steps 1–7 to test the remaining channel.


PHOTO 3: Removing the original PC board.

FINAL ASSEMBLY AND ADJUSTMENT

1) Remove power from the amplifier and disconnect all jumper connections. Replace all 3A DC test fuses with the normal 6A fast-blow fuses.

2) Reassemble the amplifier as needed by following the steps for removal in reverse sequence, but do not reinstall the amplifier top cover yet.

3) Connect a test jumper to the source-side of the output stage resistor in each channel that was previously identified as giving the best group-average reading. Drape the free end of the jumper over the rear panel so it hangs near the updated channel's speaker terminals.

4) Reapply power to the amplifier, then adjust VR3 in each channel for a reading of 125mV DC between each jumper wire and the channel's "hot"

loudspeaker terminal. Also readjust VR1 or VR2 as needed (whichever you can reach) for minimum DC offset voltage across the "hot" and ground loudspeaker terminals.

5) Set the amplifier top cover in place loosely so it will trap warm air in the chassis. Repeat the bias and DC offset adjustments described above periodically over the next hour to bring the final DC bias reading to 130mV (1.4A) and the final DC offset as close to zero as possible.

6) Remove both test jumpers and secure the top cover in place.

OPTIONAL MODIFICATIONS

Later listening experience gave me the impression that the Forte 1b was a bit more sensitive to changes in power-supply quality than the Forte 1a. Accordingly, I made some relatively simple (and reversible) changes in an effort to improve the quality of power fed to the amplifier.

To improve AC current delivery, I replaced the amplifier's stock 16-gauge power cord with a 14-gauge equivalent. This improved bass response, but also added some hardness to the midrange, which I attributed to increased "reversed recovery" noise emission from the bridge rectifiers in each channel's power supply. Placing a ferrite clamp around the cable pair connecting each rectifier to its filter capacitors seemed to tame this problem.

I also borrowed a technique from the Bride of Son of Zen prcject[4] for preventing ground loops. In the stock Forte 1a, each channel's ground is returned to the chassis safety ground at a three-lug terminal strip bolted to the floor of the chassis near the AC power inlet. The ground for the left channel is connected directly to safety ground at this point, while the right channel's ground is connected to safety ground via a 10Ω resistor.

This technique breaks any ground loop between channels, but leaves the left channel vulnerable to ground loops between the amplifier and associated preamplifier. Adding another 10Ω resistor in series with the left channel's ground connection would solve this problem, but would also eliminate a direct path to ground that could prevent electrocution if you ever accidentally connect the amplifier to a "hot" preamp chassis. A safer alternative is to lift both sides from safety ground using inrush current limiters, which have a cold resistance of several ohms, but drop to a much lower resistance due to heating if high voltage forces significant current to flow through them.

Photo 6 shows how you can mount two Keystone CL-60 inrush limiters to a replacement three-lug terminal strip. Installation of this mod is tricky–you must either be a contortionist or temporarily remove one or more filter capacitors to gain clear access to the terminal strip. If you elect to install this option, be sure that you fully restore the safety ground, and when finished, verify continuity between the chassis and the ground pin on the power cord or IEC socket.

MEASURED PERFORMANCE

To assess the objective performance of this design, I ran tests on two differently configured channels. I set up channel A as a partial folded cascode with global negative feedback, while channel B used input stage source degeneration and local feedback. For comparison, I also tested one channel of a second Forte 1a, which I had on hand for use as a control. To document certain test results, I used a high-resolution PC-based oscilloscope[5].

Both channels of the Forte 1b were stable from a DC perspective, holding their bias and offset values within safe levels over periods of several hours. Both channels also had flat sine-wave frequency response from DC to beyond 200kHz (–3dB), and reproduced square waves into 8Ω resistive loads with no overshoot or ringing. In addition, both channels remained stable driving 2μF in parallel with 8Ω, but showed significant overshoot followed by several cycles of decaying ringing under these conditions. The overshoot was more severe than with the Forte 1a test channel, but the ringing was less so.

Figure 6 compares the 1kHz distortion spectrum at 1W and 50W for all three test channels when driving an 8Ω resistive load. The accuracy of the result in *Fig. 6a* is questionable, because it could almost serve as a picture of the

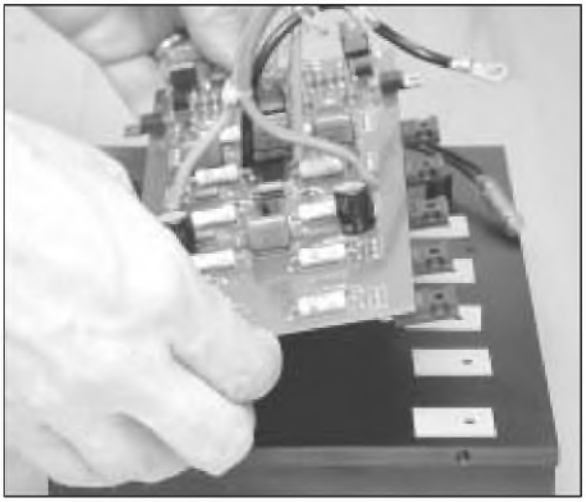

PHOTO 4: Installing the new PC board.

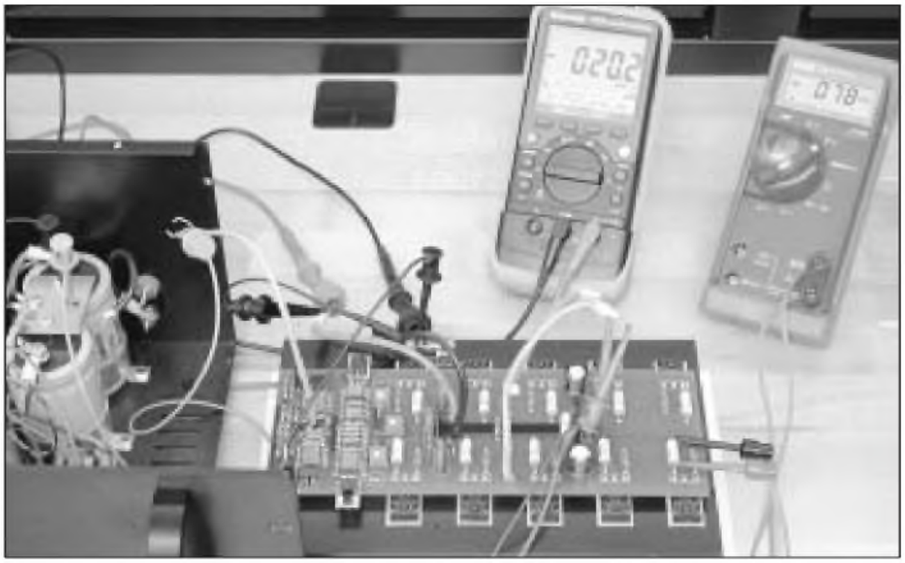

PHOTO 5: Initial testing and adjustment.

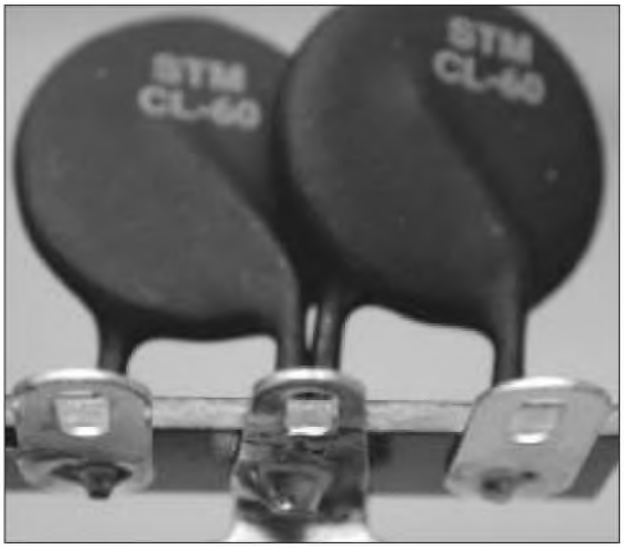

PHOTO 6: Optional signal ground modification.

appear virtually identical in this regard, so only a single waveform is shown.

LISTENING IMPRESSIONS

At last, I come to the question of whether the A75 amplifier, reduced to 50W and stripped of its regulated front-end power supply and second-stage cascodes, can sound good either on its own or in comparison to a well-executed FET/bipolar design using the same supporting hardware.

To my ears and in my system, the Forte 1b somehow manages to outperform the 1a in several subjective areas, despite its design compromises and higher measured distortion. It offers a clearer overall picture of sonic events, with improved retrieval of instrument tonality, low-level detail, and ambience information. It also handles complex symphonic material with greater ease, sounding less strained on orchestral climaxes.

The sum of these strengths was a sound that I found more interesting and lifelike than the Forte 1a, although it was not always as tidy, as euphonic, or as kind to imperfect recordings. The Forte 1a remains a highly competent design with strengths of its own, and reasonable minds may differ on which design holds the upper hand with a given system and recording.

I hope that any of you who are in a position to do so will try this project, experience the differences firsthand, and report your own impressions. ❖

test oscillator's distortion residual of about –90dB. From *Fig. 6b*, it is interesting to note that the 12-year-old Forte 1a's harmonic components are still well below the published specification of 0.1% (–60dB), at least for a 1kHz signal.

Figures 6c–6f show the higher harmonic distortion generated by both Forte 1b channels, which is consistent with the general claim of "respectable enough, but nothing to write home about" for the A75 amplifier made by its authors. At 1W, both channels show harmonic components down 60dB (0.1%) or more with respect to the fundamental, with second harmonic dominating and declining amplitude with increasing harmonic order. There is actually a bit more harmonic distortion in channel B than in channel A, with a fourth harmonic evident which does not appear in the channel A spectrum. But judging from the "hash" to the left of the fundamental in each graph, channel B's power-supply rejection appears to be somewhat better.

At 50W, the slight difference in character between Forte 1b channels A and B becomes more apparent. In channel A, second harmonic once again dominates at about –46dB (0.5%), with all higher harmonics decreasing in amplitude with increasing order. Channel B shows the same general trend, but with less consistency (e.g., third harmonic dominates over second) and a more extended harmonic spectrum.

Figure 7 illustrates the relatively benign behavior of the Forte 1b when driven well into clipping. Channels A and B

AUTHOR'S NOTE

Special thanks to Nelson Pass and Norman Thagard, authors of the original A75 articles, for providing the inspiration for this project.

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5. Hansen, Charles, "A DSO Trio, Part 3" (ADC-216), *AE* 5/00, p. 34.

6. Pass, Nelson, "Bride of Son of Zen: The Next Generation" *AE* 5/97, p. 7.

Xpress Mail

CORRECTION

An error was undetected in A.J. van Doorn's schematic on p. 72 of the March 2003 *audioXpress.* R10 should be connected to R4 and R5 as well as the CW end of P1 (*Figure 1*).

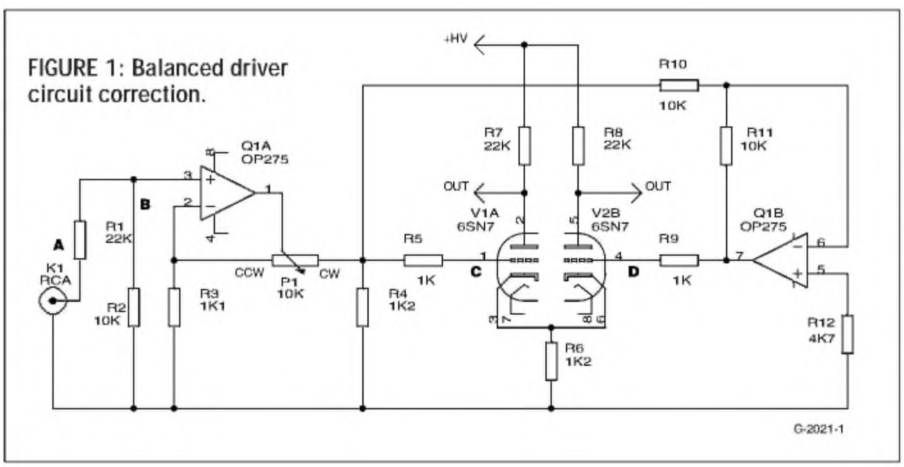


FIGURE 1: Balanced driver circuit correction.

VIDEO BOOST

Video gear contains a significant number of electrolytic capacitors—sometimes in circuits where they can't possibly support quality video, sometimes where they perform "adequately." Here are simple tips to extract vastly improved performance from your electrolytics.

First, I suggest you replace them outright with tantalum, if the values are 100µF or less, followed by the helper caps (described later). If you can't replace them with tantalum, at least parallel them with tantalum, with values from 10% to 100% of the original values—the more the better up to 100%.

My second suggestion is to bypass your tantalum capacitors with polyester or polypropylene (I arbitrarily use 0.1µF), and bypass these with a ceramic cap (again, arbitrarily 82pF).

A very important place to find electrolytics is at the analog and digital supply voltage pins of ICs. The chips can't give their best with unsteady voltages. Any decoupling cap to ground will need the "treatment," which will yield exceptional performance.

The other location where electrolytic caps are critical is in series with the signal (!), where I have found three such instances in the TV set and four in the DVD player. Again, total replacement is best; bypassing is second best, but entirely worthwhile.

Darcy Staggs
Orange, Calif.

ALL ABOUT AUTOFORMERS

The ZERO Autoformer article by Paul Speltz (Jan. '03 *aX*, p.38) was very interesting, and the autoformer design nicely implemented. Like all things technical, there are limitations in applying an autoformer (autotransformer). Let's first discuss the audio transformer as a component.

Figure 2 shows the equivalent circuit of a conventional two-winding step-down audio transformer that would act to increase the impedance of a speaker presented to a high impedance tube amplifier. The turns ratio, a, is equal to N1/N2. In this step-down example, E2 = E1/a and I2 = I1*a, and Z2 = Z1/a^2.

Ip represents the primary current from the amplifier. Ie is the iron core exciting current, which is composed of magnetizing current Im and core loss current Ic. Ic is further composed of hysteresis loss current Ih and eddy current loss Ied. Note that the magnetizing current is inductive and lags the resistive core loss by 90°. The current supplied to the "ideal" primary winding is always decreased by the exciting current, and varies with frequency.

Amplifier voltage Ep that ends up applied to the primary winding is decreased by the voltage drops across the primary winding inductance Lp and resistance Rp. An added frequency-dependent interaction occurs between the signal source impedance Rg and the winding capacitance Cp. Similar voltage drops occur in the secondary winding. This is why a transformer can never be 100% efficient.

At high frequencies the magnetizing current becomes negligible (magnetizing reactance due to Lm becomes very high), but the core losses increase. The effective winding resistance (Rp and Rs) increase due to skin effect and the eddy currents induced in the wire. The winding capacitance and leakage inductance (Lp, Cp, Ls, and Cs) greatly influence the frequency response and phase shift. These parameters can also cause stability problems in feedback amplifiers, and they are somewhat sensitive to load impedance.

In feedback amplifiers, the leakage inductance can be the deciding factor in whether or not the amplifier is stable. The transformer's resonant frequency, in proportion to its inductance and capacitance, can cause high-frequency peaking. (These latter parameters are much less a factor in a low impedance design such as one used to perform the 16Ω:4Ω impedance conversion.)

The permeability (µ, the incremental ratio of *B* to *H*) of laminated iron cores decreases at high frequency. Thinner core laminations can help, but the minimum available lamination thickness is 0.5 mils. This limits the maximum practical iron-core transformer frequency response to about 50kHz.

At low frequencies, the primary leakage reactance Lp is small, and the winding capacitances are negligible. Rh and Red are high. For good low-frequency response, the core area must be large. The primary winding resistance needs to be low so the ratio of Lm to Rp is high. Low-frequency output power decreases if the secondary load impedance increases.

At high power levels, distortion increases due to the increase in third harmonics as the core approaches saturation (the horizontal parts of the B-H curve). The typical push-pull tube amplifier ungapped output transformer can begin to saturate with only a few mA of current imbalance between the two push-pull windings. Any imbalance will reduce the low-level mutual inductance and cause it to vary with signal level.

Single-ended tube amplifier output transformers use a gapped core to handle the output stage DC current without saturation, but this greatly complicates the transformer design and increases the amount of iron required to handle a given power level. There is no inherent reason why a SE amplifier cannot have the same HF response as a push-pull design. The elements of the equivalent circuit are not fully dependent, but real-world cost and fabrication limits usually intervene.

The low-frequency portion of the audio spectrum is where the output transformer demonstrates its greatest limitations. The high-impedance output tube is trying to supply enough current to both magnetize the transformer core and supply load current to the speaker, with a finite primary magnetizing inductance. This inductance is directly dependent on the size and weight of the output transformer. The end result is that, at low frequencies, all of the available tube current fully magnetizes the core, driving it into saturation, which causes an abrupt interruption in the current delivered to the speaker.

If feedback is used in a push-pull amplifier, the closed-loop output impedance may be reduced to such a low value that the magnetizing current can approach the dynamic load current and further limit the output power into an inductive load.

As you can see, the design of a conventional audio output transformer is a complex process with many compromises. With this backdrop in place, I'd like to expand on the advantages and limitations Mr. Speltz listed in his article for an autoformer.

While the autoformer can be loosely described as having one winding, *Figure 3* shows the equivalent circuit for a step-down autoformer, the type used in Mr. Speltz's article. N1 is the upper section of the winding, N2 is the lower section, and the turns ratio is N1/N2. However, due to the fact that the autoformer uses both transformer action and conduction, the voltage ratio (for step-down) becomes E2 = E1/(a = 1). If N1 = N2, the voltage is divided in half. This is the case for Mr. Speltz's 16Ω:4Ω autoformer.

N1 is the portion of the autoformer winding that receives the "primary" current I1. All the frequency-dependent exciting current mechanisms are still in place due to the iron core. The "secondary" current I3 is increased by transformer action by the apparent turns ratio, 1 + a, and is equal to the sum of I1 + I2.

Thus, you can see that the music does not fully "come out on the same winding that it goes in on." Half the speaker current is due to conduction and half is due to iron core transformer action. Any nonlinearities imparted by

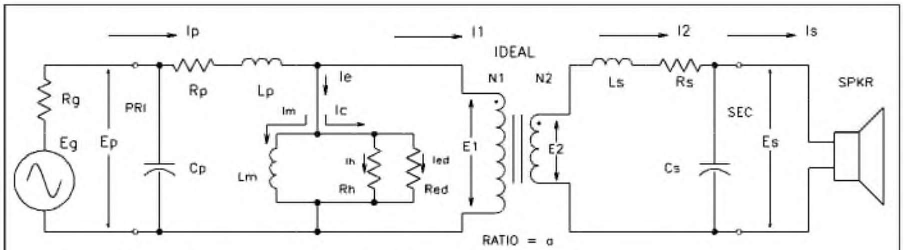
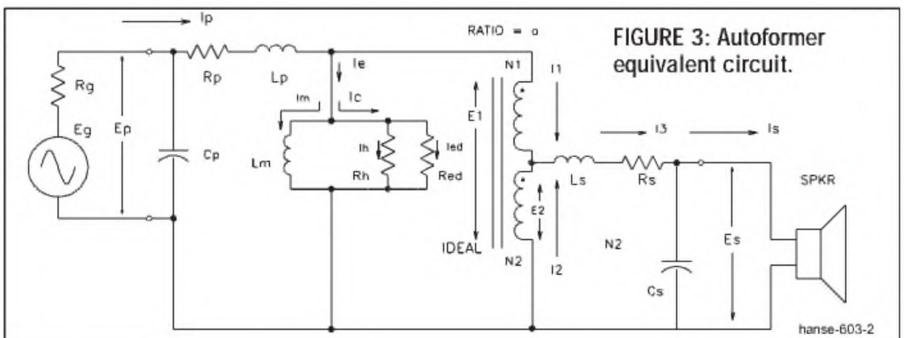


FIGURE 2: Audio transformer equivalent circuit.

FIGURE 3: Autoformer equivalent circuit.



the transformer design show up at the output. The smaller the impedance ratio, the fewer the contributions of the transformer action.

One big advantage of an autotransformer is that it can deliver the same power by transformer action as a two-winding transformer of the same core size. In addition, it delivers the additional power by conduction. In the case of the 4:1 impedance conversion, it can deliver twice the total power for the same core size. The general formula is

Pautoformer = (a + 1)*Ptwo-winding/a

Since the increase in power is that delivered by conduction, as the turns ratio increases, the power increases over a two-winding transformer become smaller. For small values of turns ratio, the autotransformer can be significantly smaller than a two-winding transformer. However, the losses in the core are the same at rated load for either type of transformer.

Charles Hansen
Ocean, N.J.

Paul Speltz responds:

Thanks to Charles Hansen for the informative follow-up to my article. Charles took autoformer theory well beyond the scope of my article, and showed us that, as with all things in this world, everything is always more complicated than first observed. His equivalent circuit models are well described and very helpful in understanding the many variables at hand.

Charles is correct in saying some of the speaker current is due to "transformer action." This is why I think an autoformer has the potential to sound better than a transformer, where all the speaker current is due to transformer action. The ability to provide some of the speaker current by conduction is part of the autoformer advantage.

He mentioned that for good low-frequency response the core area must be large. Since I specified the autoformer down to 2Hz (and it will do it), I wonder how much more power, above its 150W rating, it can handle if the low frequency is limited to the typical 20Hz . . . especially after reading that an autoformer can be made smaller than an equivalent power-rated transformer, and the ZEROs are fairly beefy at 8.5 lbs.

Charles describes the maximum frequency response practical limit to be about 50kHz due to the minimum available iron core lamination thickness of 0.5 mils. I do not have an explanation for it, but I specified the autoformer up to 200kHz, and it will do that loaded up on the bench. It is actually down only 6dB at 2MHz! I know it sounds unbelievable, but that is how it performs sourced and loaded with non-inductive resistors.

Thanks again for Mr. Hansen's efforts. The ZERO Autoformers have been a lot of fun . . . as any hobby should be.

AVOID FIBERGLASS PARTICLES

Regarding "Easy Layouts and Other CAD Tricks" (Jan '03 *aX*, p. 36), thanks to Mr. Brunner for some useful suggestions for applying printed CAD documents directly to metalwork, artwork, and circuit boards. It may be worth a note of caution that when milling or grinding fiberglass it's a good idea to use a vacuum and facemask to control the resulting dust. Apparently fine fiberglass particles can be injurious to the lungs. Most fiberglass workers wear full respirators to control their exposure.

Jeff Chan
Los Altos, Calif.

SAFE SOLDERING

On reading Paul Whiteman's tool modification for "Quick Desoldering" (Jan '03, p. 72), two thoughts popped into my mind.

1. Good idea.
2. Safety hazard. I don't know what else Paul uses his workshop vacuum for, but I often find mine full of sawdust. Sawdust is very flammable, and sanding dust is worse. I know that I have been cautioned not to send hot metal dust (as from grinding metal when sharpening tools) into a shop vacuum. I think the same caution would apply to sending hot solder into sawdust.

I suggest that soldering a couple of pieces of metal tubing onto opposite sides of a closed tin can, and putting the tin can in the plastic tubing between the desoldering tool and the vacuum, would eliminate the possibility of hot solder ending up in sawdust. The air would pass on to the vacuum, but the solder would end up in the can.

I like to see work get done fast, but I don't care to hear any sad stories about flaming vacuums. Recovering from a fire takes a lot longer than removing many, many ICs.

Ted Miller
Elkhart, Ind.

Paul Whiteman responds:

I would like to thank Mr. Miller for his interest in my de-soldering iron modification.

The only woodworking I do is to build cabinets to house my guitar amplifier designs and the speakers for them. These are not built in my shop but in my garage, where there is more room and cleanup is easier.

I do not honestly believe that you can start a fire using molten solder. The temperature of metal grindings, on the other hand, is far above that of molten solder and can indeed be a fire hazard. Like most of us who solder as part of a hobby or profession, at one time or another I have dripped solder onto just about everything in my shop, including myself, and I have never been concerned about it causing a fire.

Looking at my modified iron, I can see solder splatters in the tubing for about six inches from the handle. From this I think it is safe to conclude that by the time the sol-

der gets to the vacuum, it has cooled and solidified. Most of the solder remains in the barrel of the iron, as was originally intended, and can be cleaned out from there. If you are still concerned, empty your vacuum first, thus removing any possibility of a problem.

LETTER REACTION

There is little merit in the letter from Dana J. Olson that appeared in the March 2003 issue (p. 67), except perhaps to remind us that the barbarians are still at the gates. Never has this magazine published another letter quite like it.

I find his letter distasteful on two grounds. First of all, Mr. Olson is neither an expert nor a subscriber, so he has no standing with this magazine. He was, in fact, a guest, and he behaved abominably. He never intended to subscribe, so he obviously never expected to see a reply. The only purpose of this letter was to offend.

Second, the bulk of his letter is filled with so much pseudoscientific bilge that it defies rebuttal. I restrict myself to the following point: at the present time, measurement can only determine roughly how a component is working; it cannot yet consistently predict the way the device will sound. Perhaps some day, a definitive set of measurements will exist, but we certainly do not have them now. It is therefore tragic that Mr. Olson does not trust the evidence of his own ears.

He also makes innumerable errors in logic, which I can expand upon to any skeptic interested enough to e-mail me (musitronix@juno.com). There is no earthly justification for taking up the matter here, as I believe I have more than made my point.

Bob McIntyre
Toledo, Ohio

HELP WANTED

I've been trying to get the amp ("A Hybrid Tube/MOSFET SE Amp," May '01 *aX*, p. 18) to work by contacting the author over the past year or more. Although he's answered most of my questions, I still would like to get in touch with one or more of the two dozen people in the US who the author claims have successfully built his design from a kit that he sells. My requests to Generoso Cozza for an e-mail address for these folks has not been granted. Maybe one of them will "come out of the closet" and share their secrets with us.

I'm becoming frustrated spending so much time and money replacing burnt components and not getting the amp to work. I'm not alone in this situation, as others have contacted me for info that would help them as well.

Joe Wdowiak
flyfishonly1@hotmail.com

Could anyone at *audioXpress* perhaps assist in advising where I can find circuit details and spares for the Philips CD Transport CDM-9/44 used in the UK-built AVI-CD players? Neither AVI nor Philips seems to have spares available, and Philips is not responding to my requests for circuit details, which AVI has lost. I'm really at my wit's end, having two faulty units that the customers are screaming for.

Jan Eigenhuis
Lowveld Audio
South Africa
goodhifi@soft.co.za

I'm looking to build a SE valve amp using two KT66s in parallel. I cannot find any information on this type of amp design. I contacted World Audio in the UK and they advised I e-mail you. Has your magazine ever published any articles on this valve utilizing KT66s in SE mode? I can get info on all other types: 2A3, 300B, 6550, KT88, and EL34, and so on, but not KT66!

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

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WANTED

Information on any books, articles, or sources of information on the construction of audio/step-up/output transformers. Contact: Angel Rivera, alrivera@tampabay.rr.com.

New Chips on the Block

AK4358

By Charles Hansen

Asahi Kasei Microsystems Semiconductor (AKM) has introduced the AK4358, an 8-channel D/A converter compatible with both the SACD DSD format and DVD-Audio 24-bit, 192kHz sampling format. The AK4358 offers high performance in DVD, SACD, AV receiver, and car audio applications. The 8-channel, multi-format-compatible AK4358 is based on the AK4357 6-channel DAC and offers a host of key features. These include reduced out-of-band noise achieved through the use of an advanced multi-bit architecture and excellent anti-jitter performance from the switched capacitor DAC.

In order to minimize power consumption, the AK4358 uses new low-power internal circuitry for a 12% improvement in power consumption efficiency, without adversely affecting audio performance. Other key functions of the AK4358 are dedicated input ports for direct stream digital (DSD), which enables switching between PCM and DSD data. A built-in low-pass filter, specifically designed for DSD/SACD, enables "Scarlet Book" filter compliance to be easily achieved by adding a simple external analog filter.

The same output level is set for PCM data and DSD data so that digital volume and zero-detect functions work in the same way for both PCM and DSD data. This unifies playback levels without the need to be aware of what data format is being used. The AK4358 is available in a 48-pin LQFP package. The part is priced at $5.21 U.S. in 1K quantities. Evaluation boards and samples are available now directly from AKM.

To obtain more information, contact AKM Semiconductors, 2001 Gateway Place, Suite 650 West, San Jose, CA 95110, (888) 256-7364, website http://www.akm.com. ❖