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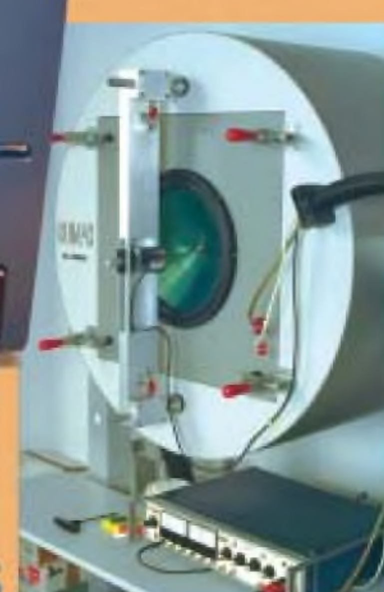

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## DEPARTMENTS



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

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

JOHN STUART MILL



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# A Small Reflex TL

Here's a new way of assembling a transmission line enclosure. You don't need the use of a table saw to build this compact loudspeaker enclosure, which gives excellent bass performance. **By Matt Hills**

I had been using my current PC speakers—a pair of Alpine 6½″ coaxial types with a capacitor in series with the tweeter—for a number of years. They started out in the doors of my old 1980 Toyota Tercel hatchback, but now they're mounted in small sealed boxes above my PC down in my basement workshop. They're pushing 20 years old now and it shows (and sounds!). I thought that the time was ripe to replace them with something nice and simple that could be constructed easily.

I always wanted to build a simple transmission line with Sonotube—perhaps a one-way design. It seems a perfect enclosure material if you stay within its limits. I had been thinking about a transmission-line enclosure based on concentric tubes of increasing diameters folded back on itself. If I may, I will coin the name "concentric reflex transmission line." A drawing of this type of enclosure is in *Fig. 1*. The finished pair of speakers is shown in *Photo 1*.

## DRIVER SELECTION

I looked around for a fairly efficient, broadband (full-range) driver in the 5–6½″ range. The Fostex Model Sigma 168 had a nice, flat response, but the $X_{MAX}$ spec of 1.25mm indicates that these are intended for horn designs. They were also a little pricey for this application.

Then I saw the little full-range Vifas at Parts Express. For less than 20 bucks US each, full range, rubber surround, and a cast (OK, injection-molded plastic) basket, what's the worst that could happen? They don't have an extremely low $f_s$ (self-resonance frequency), so the design could be fairly compact and I'd be able to exploit the low-frequency performance in a fairly lightweight, compact enclosure. With no tweeter or crossover, it would require only a couple of weekend's effort!

Or not—whizzer cones are best left to the Lowthers and Fostexs of the world. Not that whizzers are necessarily a bad idea, but I think that these folks must put a lot of effort into designing the mechanical system and choosing their materials for true high-end performance. The high-frequency response of this driver wasn't terrific. There was a huge (12dB deep) notch at about 8500Hz or so. This kind of high-frequency response was unusable.

The low-end response was another story. A brief audition in the first spin of the enclosure showed great promise in the bass and lower mids, but the highs were indistinct and distant. I removed the whizzer cones with an X-acto #11 blade by cutting as close to the dust cap as possible. Removing the whizzer exposed a gap in the voice coil, which I sealed with a drop of five-minute epoxy.

The published versus measured T/S (Thiele/Small) parameters are in *Table 1*. The Qs weren't too far off, but the $f_s$ was more like 57Hz (91.7Hz published). I measured these parameters prior to removing the whizzer cone.

I'm always leery of any driver $f_s$ spec with one-tenth hertz resolution. I think that it's meaningless to provide that much precision for a parameter that can change with room temperature. It may be the driver design nominal, although I don't know about that either because the web page advertisement stated that an $f_3$ (–3dB system frequency) of 75Hz was obtainable in a 0.25ft$^3$ vented box. It seems unlikely that a woofer could ever go 20Hz *below* its $f_s$

PHOTO 1: The completed pair of speakers.

**ABOUT THE AUTHOR**

Matt Hills graduated from Algonquin College in 1984 with a degree in Electronics Engineering Technology. He currently designs RF hardware for phased-array antennas. A recent father of one, he enjoys all things audio, cooking, camping, and spending time with his family. He can be reached at margmatt@sympatico.ca.

CARDAS
CARDAS
Quad Eutectic Solder
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after putting it *into* a box.

Anyway, it's just as well. All things being equal, I'd rather have a speaker that could do 65Hz than only 90Hz. This value of $f_s$ is just fine for the tube lengths I had in mind. In order to keep the design small (<1ft$^3$), I had hoped to use tubes about one foot long. This would result in an Fp (pipe resonance) of about 75Hz—this means a total equivalent pipe length of about 44–48″, which is the line length of the enclosure shown in *Fig. 1*.

Since this is no longer a single driver design, I was in need of a tweeter—one small enough to fit on the baffle. The high frequencies for this design are handled by a small (10mm) Audax dome tweeter P/N TW010E10 that was on sale at Solen for about $8 each (Canadian). Since the Vifa drivers were reasonably well behaved up to about 6kHz (the recommended crossover frequency for the Audax tweeters), I thought that these would be a good match.

I would have liked to have crossed over at 3–4kHz, but this tweeter wouldn't tolerate the excursions. There was also nearly no tweeter output at those frequencies. Audax specified the recommended 6kHz high-pass response for the tweeters as a first order with a 2μF series capacitor.

## DRIVER RESPONSE

Some of you are probably already saying that the $f_s$ of the driver must be the same as the $\lambda/4$ length (Fp) of the line. I had been reading George Augspurger's[1] paper on transmission lines presented at the 107th AES convention in New York. He has some new ideas about modeling transmission-line behavior as a classical lumped element model of a transmission line normally found in textbooks dealing with microwaves and RF. With this model a SPICE simulator may be used to analyze the performance.

I find the symmetry of this approach quite pleasing. His measurements appear to bear out the validity of the models. The trick is to determine the losses and other line parameters used to compute the values of the Rs, Ls, and Cs in the final model. This is probably an empirical process but does not invalidate the approach.

Mr. Augspurger also supplies a range of $f_s$ to Fp ratios, which he says should produce satisfactory results for certain specific design types. He indicates that $f_3$/Fp ratios outside the range of 0.7–1.4 should be avoided if you don't want to sacrifice efficiency (and I suspect transient response). For this design the ratio would be about

$$57\text{Hz} / 74\text{Hz} = 0.77\,.$$

There is certainly no end of debate in the field of loudspeaker design as to the exact merits, principles of operation, and design of transmission-line loudspeakers. What I can offer about the operation of this design is that the frequencies above about 300Hz are greatly attenuated at the rear of the cone, the

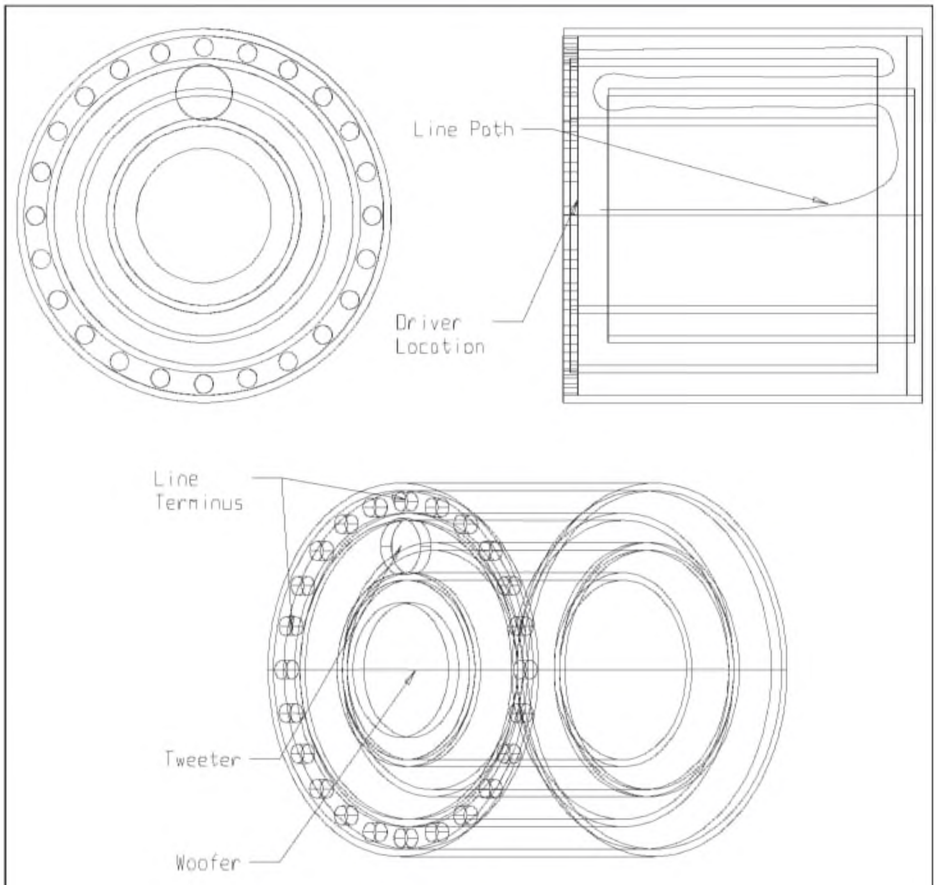


FIGURE 1: Multiple views of the enclosure produced in 3-D CAD.

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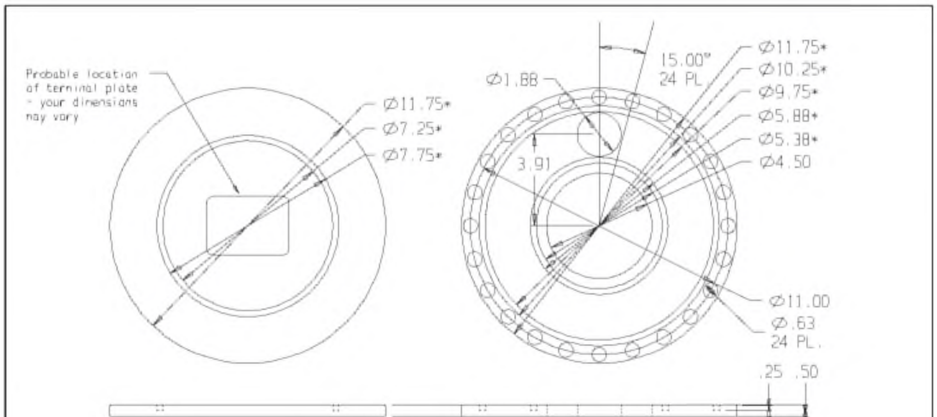


FIGURE 2: Machining details for the MDF front and rear panels.

B-hills-2

design does not have a fourth-order rolloff as vented designs do, and the $f_3$ of the system is equal to the $\lambda/4$ length chosen for the line length. There is also definite reinforcement of the bass frequencies by the enclosure.

## DESIGN AND TEST TOOLS

For this design I used SpeakerWorkshop V1.00 by Audua Inc.[2] This is the first time I've had access to such a tool for loudspeaker measurement. This is an all-inclusive loudspeaker system design package, which is available as a Beta test version on the Internet. It performs a wide variety of loudspeaker measurement and design functions, including near-field, on-off axis, and frequency-response measurements. It is also capable of MLS, sweep and simple sinusoidal measurements, file and design management as well as crossover design and simulation.

It also comes with some decent help files but still requires that you have a clue about what you are doing. Previous experience with CAD or measurement software is desirable. I can honestly say, however, that it is a most remarkable piece of software, and I have come to rely on it quite heavily. Once you get the hang of it, it becomes very easy to use. It is very helpful to construct a measurement jig for use with the package, as I have. More information about the jig is available on the Internet.[3]

My microphone is a calibrated Panasonic capsule from Liberty Instruments[4], and the preamp is a 20dB gain, TL072-based design similar to the Wallin preamp[3], but without the clipping indicators. I also use XOPT and an old SPICE simulator. XOPT is a crossover optimizer that is no longer available.

Note that I performed all frequency-response and impedance measurements with SpeakerWorkshop 1.00, and measured all frequency-response data at one-meter, one-third octave smoothed, and with a Sound Blaster Live Platinum 5.1 Live-Drive card set for 48kHz sample rate.

## CONSTRUCTION

The cutting details of the MDF pieces are given in *Fig. 2*. Particularly important is the fact that the tubes are nearly never the exact size that they are sold

**TABLE 1**
**T/S PARAMETERS FOR THE VIFA PARTS EXPRESS 299-246**

| PARAMETER | PUBLISHED VALUE | MEASURED VALUES |
|---|---|---|
| Frequency Response | 65–12000Hz | approx. 65–12000Hz |
| Resonant Frequency ($f_s$) | 91.7Hz | 57Hz (measured with SpeakerWorkshop and verified with HP 3577A network analyzer) |
| Sensitivity | 92dB/2.83V/1m | - |
| Magnet weight | 14.6 oz | - |
| Nominal Impedance | 4Ω | 4Ω |
| $Q_{MS}$ | 3.1 | 3.6 |
| $Q_{ES}$ | 0.80 | 0.663 |
| $Q_{TS}$ | 0.637 | 0.561 |
| $V_{AS}$ | 0.14ft$^3$ (3.96 ltr) | 0.452 (12.8 ltr) (All Q values and $V_{AS}$ measured by SpeakerWorkshop using added mass method) |
| $X_{MAX}$ | 4.5mm` | - |

as. It's probably the same reason why a 2 by 4 is never 2″ by 4″.

Measure the tube diameters before cutting a circle or groove, especially for the outer 12″ cut. It must be a snug fit inside the tube, but not so much so that it is impossible to assemble. The grooves for mounting the inner tubes are ¼″ deep as shown. That was enough to hold the tubes in place.

Refer again to the multi-angle view of the enclosure as given in *Fig. 1*. It is constructed from four different diameters of tubing, one inside the other. The tubes are assembled together with the MDF fronts and backs by gluing the ends of the tubes into slots routed into the MDF pieces. A Jasper circle jig or other fixture is a must for the routing. The tube sections are 10.25″ long for the 6″, 8″, and 10″ diameter pieces, with the outer 12″ diameter section 12″ long.

I cut the tubes with a hard-toothed cross-cut handsaw. The trick to getting a good square end cut on a tube is to wrap the tube with a piece of tractor-feed computer paper or similar long strip more than, say, 4″ wide. As long as the paper is tight to the tube, it will line up with itself once wrapped and define a cut line around the tube that is square.

The tubes and lengths selected result in a path length of about 44″. This gives a $\lambda/4$ length of:

$$1088 \times 12/(4 \times 44) = 74\text{Hz}$$

where 1088 is the speed of sound at sea level in ft/s. The approximate length in inches of the path is 44″.

Even though I selected 6, 8, 10, and 12″ tubes, it is the space between consecutive tubes that forms the line area, and you end up with a cross-sectional area nearly equivalent to a 6″ diameter tube (±22%). This is probably constant enough for an application like this. I have seen working designs with greater variations—more on the "minus" side—to allow the line to fit in an enclosure of the smallest possible external dimensions. The vent holes at the end add up to about 7.36 in$^2$, which is equivalent to about a 3″ diameter port. This is a constriction when compared to the rest of the line.

I selected the holes for appearance and convenience and simply drilled them with a ⅝″ drill bit. I have not quantified the effect of the constriction, but I am happy enough with the results to ignore it if there is one. They may be opened up to about double the area if desired.

I coated the edge of the front baffle with a thin layer of carpenter's glue, inserted it into the rear of the 12″ tube, and slid it in place to ease assembly. I glued the 6 and 10″ sections to the grooves routed into the front (baffle), with the 8″ section glued to the groove routed into the rear. The glue I used for this was regular yellow carpenter's glue, which adhered to the tube material very well, and the final bond was very sturdy.

## STUFFING

After the glue dries, start the enclosure stuffing, which is more like wrapping. The material in this case is the polyester fiberfill usually found in stuffed animals and pillows. You can purchase it at a fabric store in bags resembling pillows. It unfolds into a sheet and is sold by unit area, not by unit mass.

I stuffed the enclosure as indicated in *Table 2*. In the future I would be inclined to purchase the material in bulk off a large roll (as I later discovered it was available). The folding and bagging puts puckers and irregularities in the material, which will hinder your ability to cut it into regular shapes. The material unrolled to about 80″ × 100″ × ½″ thick when uncompressed.

Each of the fill sections is made from multiple strips of the polyester material simply because I needed the length and it simplified the stuffing. In the case of the inside of the 6″ tube, for example, the breakdown was two pieces 90″ long plus a piece 70″ long. It's the total length of 250″ that mattered. To put things into perspective, this quantity of

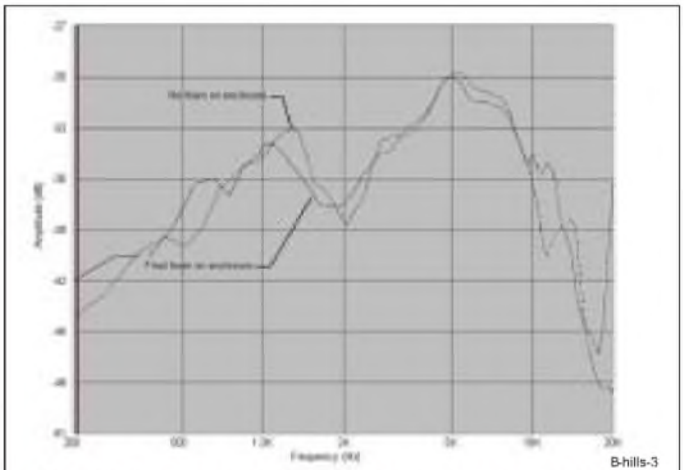


FIGURE 3: Improvement in the frequency response around 2kHz by adding foam to the enclosure edge measured by SpeakerWorkshop.

PHOTO 2: Enclosure halves ready for stuffing (one enclosure).

fiberfill weighed 5 oz. This equates to about 0.31 lb. In a 6″ diameter tube 10″ long, that equates to 1.9 lb/ft$^3$, which is the approximate stuffing density used throughout, except for the last section, which is about 1.25 lb/ft$^3$.

I rolled the 6″ tube stuffing with moderate tension until I obtained a cylinder about 9.25″ long by 8″ in diameter, and then inserted this into the 6″ tube along with the wire for connecting the woofer, which runs along the inside wall of the tube. The wire for connecting the tweeter runs down the exterior of the 6″ tube and is held in place with tape. I used a small diameter wooden stick (like the handle of a wooden spoon) to aid in wrapping the exterior of the 6″ tube with stuffing. I wrapped the polyester strips around the handle of the wooden spoon and inserted the stick into the space between the 6 and 10″ tubes and fixed the polyester to the outside of the 6″ tube. I then spun the stick so that the polyester wrapped around the outside of the 6″ tube in a manner similar to spinning cotton candy onto a paper serving cone.

I simply wrapped the exterior of the 8″ tube by hand and gently stuffed the four pieces for the interior of the 12″ tube evenly into place with a ruler. You can hold external tube wraps on the 6″ and 8″ tubes in place with a rubber band or a section cut from support hose (panty hose). Try not to compress the stuffing too much.

I tried a couple of different stuffing densities—a low density stuffing (<1 lb/ft$^3$) and the final one (nearly double

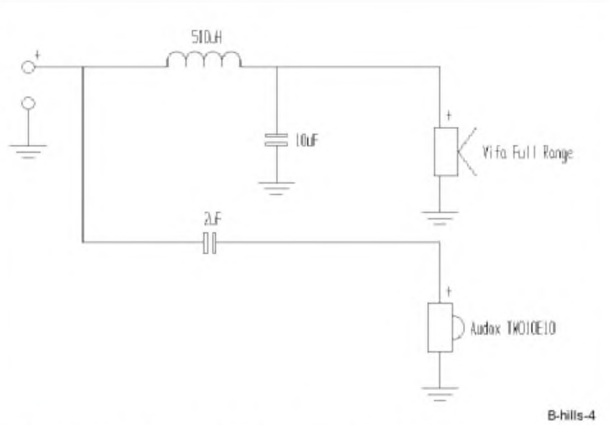


FIGURE 4: A schematic of the crossover network.

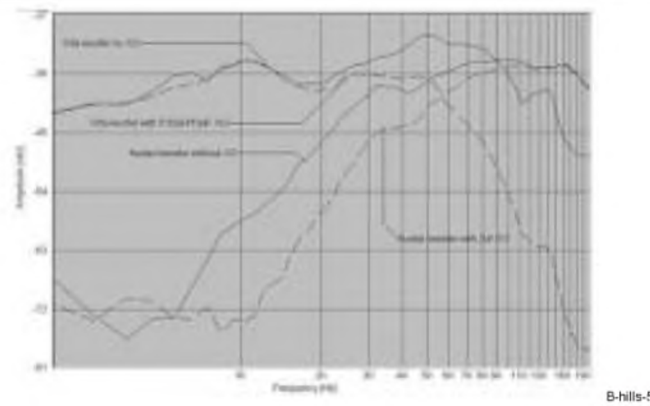


FIGURE 5: Plot of individual driver responses with and without the effects of the crossover network measured by SpeakerWorkshop.

that) given in *Table 2*. I listened to and measured both stuffings, and ultimately selected the higher density stuffing more on the basis of measurement of the low-frequency performance (near field) than listening tests. Both stuffings sounded acceptable. A complete parts list is provided in *Table 3*. The stuffed enclosures are shown in *Photo 2*.

## A MINOR GLITCH

The selection of enclosure shape is critical to the performance of a loudspeaker. As it happens, I've selected one of the worst shapes for an enclosure. I've seen Olsen's plots[5] regarding the effect of enclosure shape on frequency response plenty of times since I first became interested in loudspeaker design. Spheres are the best; cylinders are not the best.

My measurement of the frequency response showed a 6dB suckout at 2kHz. This is purely a result of the selection of a cylinder with a driver in the center of one end, and because the driver is equidistant from all of the enclosure edges. It's roughly a half wavelength from the driver edge to the enclosure edge at 2kHz. Any edge-diffraction effects are maximally reinforced in the far field.

A way to deal with this would be to select a different shape for the MDF baffle (but, of course, still route a round groove for the tubes). To bear out my theory, I cut a 12″ diameter hole in a 2′ × 4′ piece of Styrofoam and inserted the enclosure front into the hole so that the drivers were flush with the surface of the Styrofoam mimicking a 2′ × 4′ baffle. The frequency response flattened out. Perhaps a 14″ × 14″ square or an obround shape would have been a better choice for the baffle to minimize the degree of the effect.

Another way to deal with this is to apply some form of edge treatment to suppress the diffraction and subsequent cancellation phenomenon. This is the solution I chose. You can introduce a lossy material at the edge of the baffle; I selected a waffle or convoluted foam normally used as packing in shipping boxes. The foam was about 2½″ thick and was cut into 3″ wide strips for wrapping around the enclosure edge.

This approach is similar to one often employed to smooth ripples in antenna radiation patterns at antenna test ranges and in certain fixed installations. In those cases a lossy carbon-loaded foam (E-field) or ferrite-loaded paint (H-field) is employed. For acoustic applications foam works just fine.

*Figure 3* shows the results of a one-





**TABLE 2**
**DIMENSIONS OF FILL MATERIAL FOR THE ENCLOSURE**

| SECTION | POLYESTER FILL SIZE |
|---|---|
| Inside of 6″ tube | 250″ × 9.25″ |
| Outside of 6″ tube | 220′ × 9.25″ |
| Outside of 8″ tube | 250 × 9.25″ |
| Inside of 12″ tube | Four strips 7″ × 78″ |

**TABLE 3**
**PARTS LIST**

| DESCRIPTION | QUANTITY |
|---|---|
| 6″ forming tube | 2′ |
| 8″ forming tube | 2′ |
| 10″ forming tube | 2′ |
| 12″ forming tube | a bit more than 2′ |
| 2 MDF | approximately 4ft² |
| polyester stuffing | approximately 3 lb |
| speaker wire | approximately 5′ |
| 10µF bi-polar cap | 2 |
| 2µF polyester or polypropylene cap | 2 |
| 510H/0.3Ω coil | 2 |
| Vifa full range Parts Express P/N 299-246 with whizzer cut off | 2 |
| Audax TW010E104 tweeter | 2 |
| Terminal cup | 2 |
| screws, putty, glue, foam, carpet, and so on | as required |

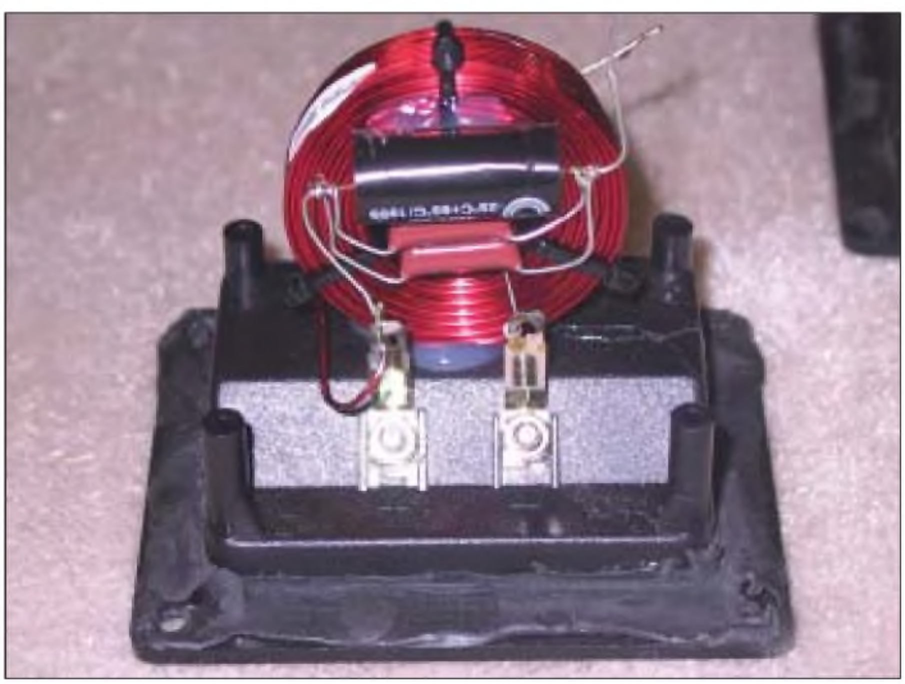

PHOTO 3: The assembled crossover network.

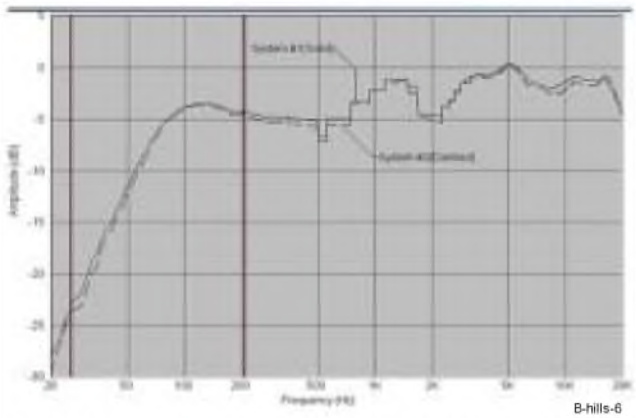


FIGURE 6: The total response of both systems measured by SpeakerWorkshop.

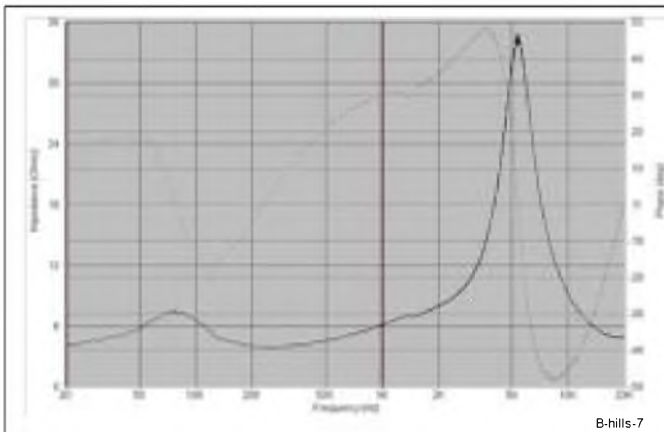


FIGURE 7: Plot of the system impedance measured by Speaker-Workshop.

foot diameter enclosure with and without a small foam ring. The suckout doesn't disappear, but it is reduced.

You assemble the enclosure by resting the front baffle assembly on its front and inserting the rear section with a twisting motion in the direction of the stuffing wrap to keep it from coming undone. When you're about to insert the rear panel, brush carpenter's glue on to the edge of the MDF panel and gently tap it into place.

## THE CROSSOVER

The crossover schematic with drivers is shown in *Fig. 4*. A first-order high pass is used on the tweeter (per Audax) and a second-order low pass for the woofer. The assembled crossover is shown in *Photo 3*. All components are hot-melt-glued to the terminal cup. All connections are made by soldering, and Teflon sleeving is used for the insulation of any bare leads.

I tried to get by with a first order on the woofer, but the peak in the response at 5kHz needed to be tamed. The crossover shown provided the flattest frequency response. A plot of the driver responses with and without the crossovers is given in *Fig. 5*, with the total response in *Fig. 6*. The staircase effect in *Fig. 6* is probably due to the resampling after splicing the near-field low-frequency data to the on-axis response. It can be removed with smoothing; however, I left it in as the data had already been smoothed once.

I measured the data below 500Hz in the near field about 3″ from the baffle in an attempt to sum the port and driver responses. I then spliced it into the on-axis 1m response using SpeakerWorkshop. The system impedance is shown in *Fig. 7*.

## MORE CONSTRUCTION

You can mount the drivers on the front baffle with screws and seal the drivers against the baffle with plumber's putty or other gasket material. The woofers and terminal cups use #8 × ¾″ pan-head screws; the tweeters use #6 × ½″ pan-head screws. The drivers are electrically connected in phase. The other ends of the wires are, of course, connected to the crossover per the schematic.

Mount the terminal cup with sealing putty and glue the industrial carpet to the front baffle with contact cement and tube—cut the holes for the drivers and terminus details afterward (trust me!). When the driver holes are cut, insert the drivers for fit. Then use the driver flanges as templates for cutting the carpet so you can mount the drivers on the wood. The foam for the edge diffraction is held in place with hot-melt glue.

## SOUND

I've been listening to these units in my living room system (much to my wife's chagrin), and they sound pretty good. There's more real bass than I would have ever expected from a 5¼″ driver, and the cone excursion doesn't look too high even for loud, low bass, although I haven't tried the TELARC *1812 Overture* yet! The highs are unremarkable, but for an $8 tweeter they're acceptable. All in all, they are much better than the little Alpines I've been using. They are easy to drive and mount (hanging by light chain and hooks from the floor joists above my workbench).

## FINAL THOUGHTS

This type of enclosure is limited to smaller "mid-bass" drivers. I'd hate to see such a design attempted with a 12″ woofer. Assuming that the depth can be tolerated, I could see extending the enclosure to 18″ long, giving a low end of about 45Hz assuming four sections. Beyond that, tube diameters up to 24″ are available, and if you had the patience you could build a reflex of ten tubes. An 8″ deep enclosure could render an 80″ path, but it would be two feet in diameter. Not a pretty sight.

If I was going to do this again, I might not use a circular baffle. An alternative is to offset the tubes axes so that they are tangent to one another, so that the woofer is not in the center of the baffle. This would reduce the ripples in the response. As they are, however, they sound better than the low cost would indicate (especially the bass), and I'm happy with their appearance—foam and all!


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I would like to thank my wife Margo for her support and my friend Pat Hill for suggesting that I write this article in the first place. I would also like to thank Mark Zachmann at Audua for his assistance with SpeakerWorkshop. ❖

### ACKNOWLEDGMENTS

1. This design is certainly not meant to compete with the recent "THOR" design published in the May '02 issue of *aX*. I believe that while it may share many of the attributes of a transmission-line design, it probably isn't the high-resolution instrument that "THOR" is. It does, however, represent a compact design that doesn't compromise too much on line length in order to save space. The design was also meant to introduce what I believe is a new idea using concentric tubes to produce a compact transmission-line design.

2. Recently *aX* published "THOR," a scientifically designed transmission-line loudspeaker based on the work of G.L. Augspurger[1]. "THOR" incorporated a number of features said to give good low-frequency performance, which could also be modeled on a computer (a tapered line, offset drivers from the end of the line, and an air volume behind the driver). My concentric reflex design doesn't borrow as much from Augspurger, nor was it modeled on a computer. I simply tried to stay within the range of $f_s$/fp (woofer resonance frequency to pipe resonance frequency) recommended in the paper. The notion of a compact transmission line has always interested me, given the smaller living rooms I've had to live with. Unfortunately, there is no room for a $\lambda/4$ line at 20–40Hz in my house. This was a method I conceived to address the issue of speaker size.

# A Digital Pooge for HP's 339A Test Set

Introduce your HP distortion meter to the digital age with this handy modification. **By Charles Hansen**

After much consternation over modifying a $1700 precision instrument, I finally decided to bring my HP 339A distortion test set into the digital audio age. I replaced the (to me) useless 30kHz low-pass radio broadcast test filter with a 22kHz low-pass (LP) filter suitable for CD player/digital-analog converter (DAC) total harmonic distortion (THD) measurements.

Up until now, I used the test set 80kHz LP with the 30kHz LP to get about a 27kHz LP, but it always let too much 44.1kHz digital sampling noise through to get accurate THD measurements above 10kHz. The new filter cuts the 44.1kHz to –48dB, while preserving the 20kHz test signal. I did this without any modifications to the existing PC boards by piggybacking a small breadboard onto the back of the Function selector switch A5 PC board shield and tapping into the wiring harness. Thus, it's easily reversible.

The HP 339A has been selling on eBay in the $250–$600 range of late. Similar prices are available for the Sound Technology 1700A, but I don't know whether the circuitry is the same.

## CAVEATS

Before proceeding, please note the following caveats:

1. You are dealing with a precision piece of test equipment, and working in tight confines. If you are not careful, you can damage the test set or knock it out of calibration. The HP 339A uses a number of integrated circuits (identified by HP part numbers) that may not be readily available. However, if you work carefully, especially when soldering, you should have no problems. Having the HP 339A Operating and Service manual handy is a must. You can obtain one from various surplus test equipment manual sources. Check the Internet for availability.
2. After I made the modification, I needed to spend an hour troubleshooting a continuous full-scale meter reading because I accidentally pulled a connector wire from the relative adjust pot connection pin on the PC board. This opened the feedback loop to the meter amplifier op amp, causing it to go to maximum. I couldn't see the wire after I re-installed the A5 board.

   Fortunately, the HP 339A manual is excellent, and my misstep gave me a chance to admire the craftsmanship and ease of maintenance of the unit.

HP 339A front panel.


PHOTO 1: HP 339A interior view, bottom.


PHOTO 2: A2/A5 board area.


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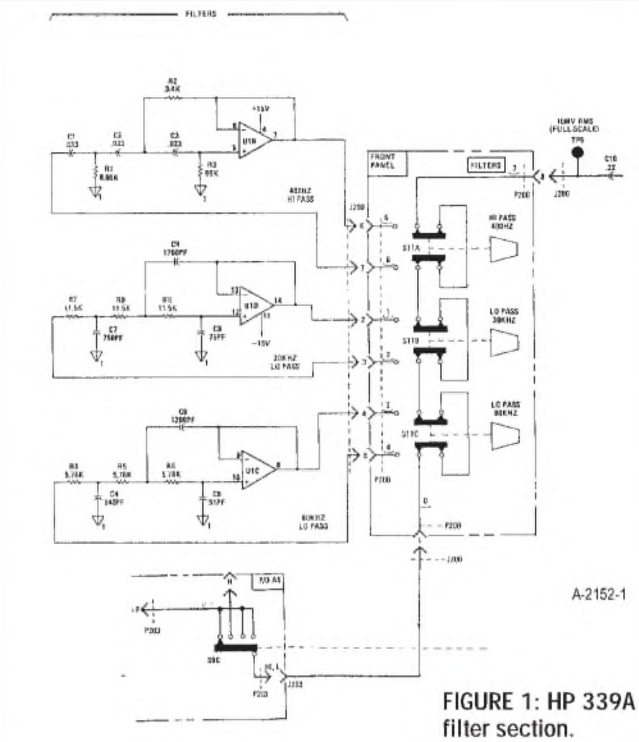


FIGURE 1: HP 339A filter section.

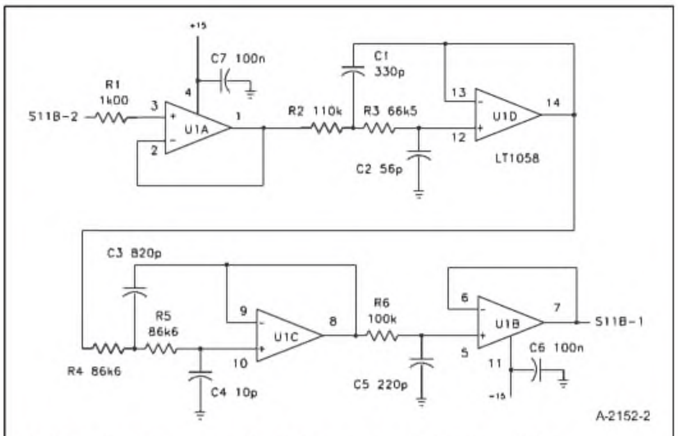



FIGURE 2: Fifth-order 20kHz low-pass Chebyshev filter.

All the PC traces are gold-plated, and the components are absolute top quality. The HP engineers even placed designations on the reverse side of their boards to aid in troubleshooting connections.

This modification does not work with the earlier HP 330 through HP 334 distortion meters, which have a single 1kHz high-pass (HP) filter that is a passive L-C circuit. Since it is in series with the meter range attenuator network, it probably has a characteristic impedance that needs to be matched.

## HOW THE FILTERS WORK

The HP 339A has three unity-gain single op amp approximations of third-order filters that are selected by front-panel push-button switches. The schematic diagram for the filter section is shown in *Fig. 1*.

The input to the filter section comes from Function switch S9C in the bottom of the figure. S9 selects inputs from the internal sine oscillator, the distortion analyzer amplifier, the analyzer input jack, or enables the variable-gain relative level amplifier. The oscillator level and analyzer input signals first pass through the input voltmeter attenuator. The three push-button switches either select or bypass each of the filters (all three can be placed in series, if you so desire).

Note that in Distortion mode the filters never act on the audio signal itself. All the THD processing is done before the filters. The distortion amplifier presents the distortion product signal, after the fundamental notch filter, to the filter selector switches. The output of the filter section goes to a 40dB amplifier stage to convert the filter 10mV full-scale (F.S.) signal to 1V F.S., and then to the RMS-DC converter for the front-panel meter circuit.

The 400Hz HP filter, U1B, is at the top of *Fig. 1*. Its function is to reduce the effect of power-line hum for measurements above 1kHz. The 80kHz LP filter is implemented in U1C (bottom circuit). This is used to reject out-of-band noise for analog 20–20kHz audio measurements.

The middle 30kHz LP filter U1D is used to provide the band limiting required by the FCC for proof-of-performance radio broadcast testing. Since I have never needed this in my years of audio testing, I decided to appropriate 30kHz switch S11B to engage a steep 22kHz LP filter that is compatible with digital audio distortion testing. Modern audio analyzers, such as the Audio Precision products, use a 6-pole 22kHz digital filter meeting CCIR 468-3 limits for this purpose.

High-order filter design is not a trivial exercise[1]. I used FilterWiz from Schematica Software to design a number of filters.

After providing design data and selecting a filter topology, the software calculates component values, then generates schematics and response graphs. I decided to use a unity-gain Chebyshev filter topology, with three second-order LP Sallen-Key stages in series. I also input the schematics into SPICE and verified their responses before proceeding to the breadboard phase.

## FILTER TESTING

My attempt to build an actual 6-pole filter did not go well. One of the capacitors was only a couple pF, and the design was very sensitive to its actual value. My breadboard wire-wrap wire probably added a couple pF here and there. It also had two non-standard capacitor values, which necessitated using parallel caps to achieve the design value. Steep analog filters also tend to have several response peaks near the end of the passband.

Although the full-scale signal level through the HP-339A filters is only 10mV, I did my testing at 1V RMS, to ensure stability at all times. The sixth-order circuit had a +2.9dB peak near 15kHz, along with the expected bump at about 18kHz. With a decreasing frequency sweep, it developed a +8.5dB jump at 10.6kHz that did not occur when I swept back up in frequency. The waveshape after the last filter section became triangular between 8kHz and 10kHz. The circuit also did not deliver the amount of attenuation I expected at 44.1kHz.

Next I tried a 22kHz fifth-order design, which proved stable, with expected response peaks. The SPICE simulation showed –43dB at 44.1kHz, and the actual circuit resulted in 42dB. In order to get a bit more HF attenuation, I tried a 20kHz fifth-order design as well. SPICE predicted –48dB attenuation at 44.1kHz, and this is what I measured. It was only –0.7dB down at 20kHz, with –7.5dB at 22kHz.

Since the highest test CD frequency is 20kHz, I decided to use the 20kHz fifth-order Chebyshev filter. The lowest critical capacitor value was 10pF, which made it a bit less sensitive to wiring than the 8p2 required in the 22kHz filter. It had a bit more dip (–0.26dB vs. –0.08dB) at 16kHz, but the HP 339A is not all that sensitive to amplitude changes during THD measurements. As long as it stays within 6dB or so, you don't even need to change the input voltage range.

I wanted to use a high-performance op amp for this steep filter. I chose the LT1058 quad FET op amp, which I have used in the past for audio test equipment. You also need accurate 1% metal film resistors and high-quality capacitors. I used 5% NP0 ceramic for their small size, but polystyrenes are also viable candidates.

My final filter schematic is shown in *Fig. 2*. The parts list is in *Table 1*. Parts cost less than $15. I used the fourth op-amp section for an input buffer. Although not necessary, 1k resistor R1 will protect the HP 339A circuitry from a fault on my perfboard, and is cheap insurance.

## CIRCUIT BOARD LAYOUT

The parts layout for my wire-wrapped perfboard is shown in *Fig. 3*. The layout is also suitable for a PC board as shown. The board is 2¼″ × 2″, with three ⅛″ mounting holes. I brought both the wiring from S11B and the input and output at J200 from the 30kHz LP filter to my perfboard. This makes it easy to restore the 30kHz filter without major disassembly, other than removing the bottom cover of the HP 339A.

**TABLE 1**
**PARTS LIST**

| SYMBOL | VALUE | DESCRIPTION | DIGIKEY PART # |
|---|---|---|---|
| C1 | 330pF 50V 5% | Ceramic NP0 | P4931-ND |
| C2 | 56pF 100V 5% | Ceramic NP0 | P4846-ND |
| C3 | 820pF 50V 5% | Ceramic NP0 | P4936-ND |
| C4 | 10pF 50V 5% | Ceramic NP0 | P4837-ND |
| C5 | 220pF 50V 5% | Ceramic NP0 | P4929-ND |
| C6, C7 | 100nF 50V | Ceramic X7R | P4923-ND |
| R1 | 1k00 1% | Metal film | 1.00KXBK-ND |
| R2 | 110k 1% | Metal film | 110KXBK-ND |
| R3 | 66k5 1% | Metal film | 66.5KXBK-ND |
| R4, R5 | 86k6 1% | Metal film | 86.6KXBK-ND |
| R6 | 100k 1% | Metal film | 100KXBK-ND |
| U1 | LT1058CN | Op amp | LF1058CN-ND |

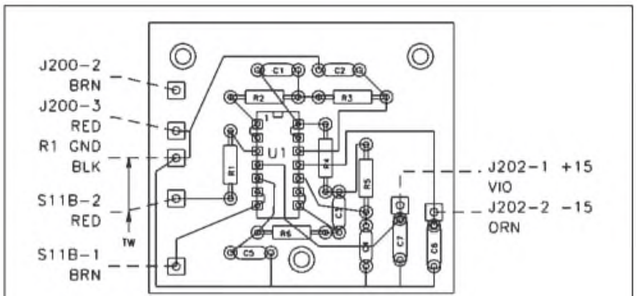


FIGURE 3: 20kHz filter board parts layout.

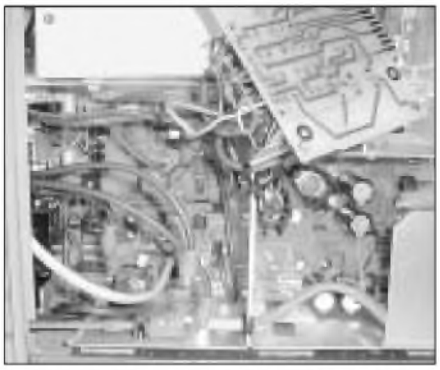


PHOTO 3: A5 board removed and secured.

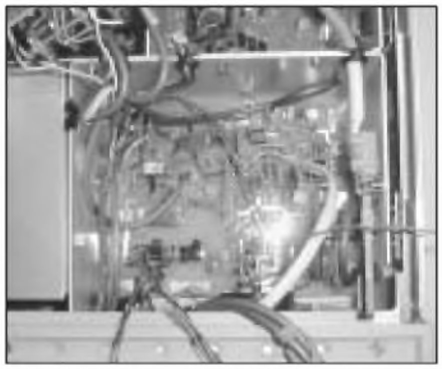


PHOTO 4: New wiring installed.

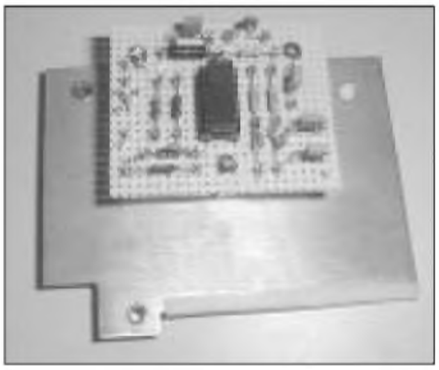


PHOTO 5: Perfboard installed on A5 shield.

## DISASSEMBLY

Before you take the HP 339A apart, review the assembly/disassembly section of the manual. Next, take a representative set of "before" data to spot any problems after modification. I took the following data:

- Maximum oscillator output voltage, all ranges, at 20Hz, 200Hz, 2kHz, and 20kHz.
- THD+N measurements of the internal oscillator at 1V RMS, and the above frequencies.
- Monitor output voltage at 3V RMS oscillator input, 1kHz, 3V range.

First remove the top and bottom covers to access the filter circuit wiring. One captive Phillips screw is used on the rear of each cover. *Photo 1* shows the interior view from the bottom. The three filter circuits are located on the A2 board of the HP 339A. The S9 Function rotary switch is located on the A5 board, which plugs into the edge-card connector J203 on the A2 board. These sections are in the lower left of *Photo 1*.

I decided to mount my 20kHz filter perfboard on the back of the aluminum shield for the A5 board, the only good location available. This requires removing the A5 board. *Photo 2* shows a close-up of the area.

Place S9 in the OSC LEVEL position. Use a 1/16″ Allen wrench to loosen from the shaft of S9 the set screw closest to the switch body. Now you can pull the front-panel knob and shaft forward to clear the A5 board. Only one screw beneath the A2 board holds A5 to A2, but five RCA connectors carry audio signals from the various other boards in the HP 339A to A5. I had to label and disconnect three of them to remove the A5 board. Labeling is important, because all the coax wires are gray, with no other marking.

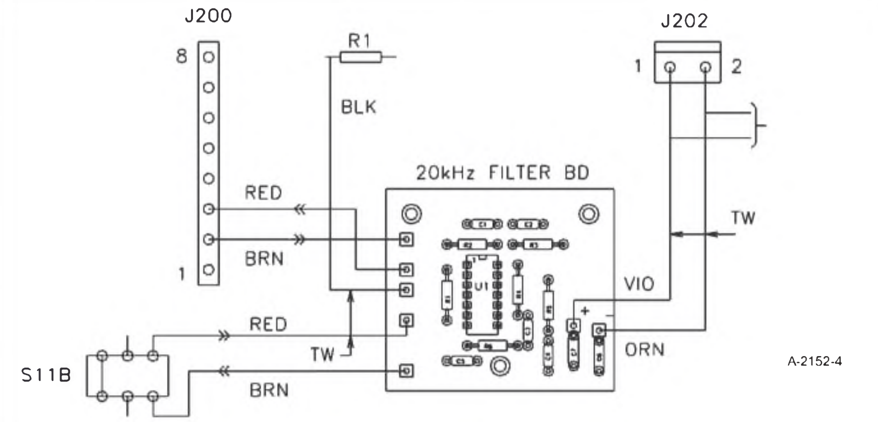




FIGURE 4: Wiring modifications.

A5 has a plastic card guide attached to the chassis shield at the back of the A2 board, and it is a tight fit getting the board past the meter movement. Once you have removed A5, the aluminum shield comes off with two screws. I used a tie wrap to temporarily hold the A5 board out of the way so I could work inside the area above A2 (*Photo 3*). Some of the wiring is directly connected between A2 and A5 and remains attached, so don't pull on the wires and break the connections. The wires to FPA and FPB (the one I opened accidentally) use push-pin connections.

The three filters connect to their front-panel switches by means of 8-pin Molex connector J200. Switch S11B uses pin 2 (output, brown wire) and pin 3 (input, red wire). I cut the two wires and soldered four extension wires to the cut halves—two red and two brown—to reach my perfboard. In addition, you need ±15V DC for the op amp and a circuit ground. I tapped the supply voltages from the wiring harness to J202. The +15V DC supply is the violet wire and −15V DC is the orange wire.

I took circuit ground from resistor R1 near J200. This is *not* the R1 that is on my perfboard; it's on the A2 board in the HP 339A. I used a black wire for ground, and violet and orange for the supplies to avoid confusion later when wiring (*Photo 4*). Use shrink sleeving on all connections.

Be very careful when soldering to the HP 339A wiring harness. The wire insulation melts easily and you don't want to drop molten solder in the unit. I put a piece of paper over the A2 board while I was working above it.

This is also an opportunity to give all the connector pins and rotary switches a coat of Caig PreservIt contact cleaner. The HP 339A is very well sealed, so there was nary a particle of dust inside. I also checked all the PC board fuses for continuity.

I drilled three 1/8″ holes in the A5 board shield and installed three 3/4″ 4-40 aluminum spacers to hold the new 20kHz perfboard. *Photo 5* shows the perfboard mounted on the shield.

## ASSEMBLY

With the perfboard installed on the

PHOTO 6: A5 re-installed and wire routing.

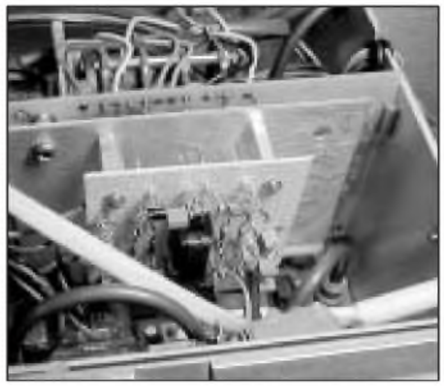


PHOTO 7: A5 and perfboard close-up.

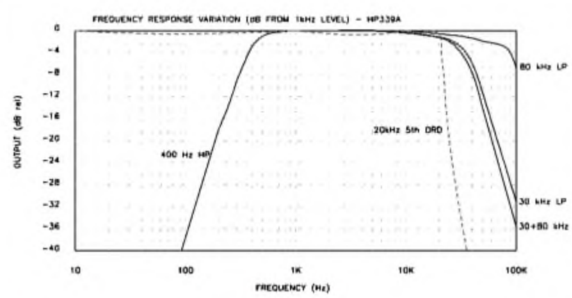
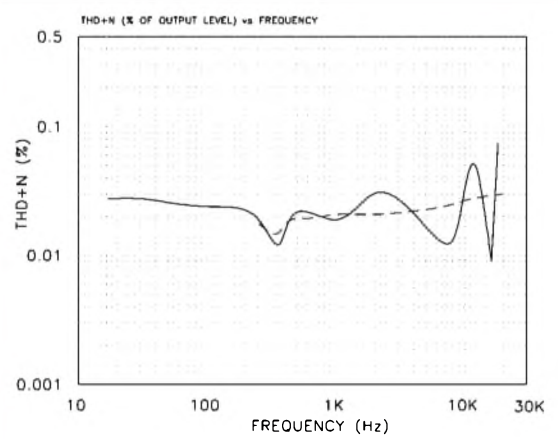



FIGURE 5: Filter response curves.

A-2152-5




A-2152-6

FIGURE 6: THD measurements using 20kHz LP filter.

shield with ¼″ 4-40 screws, replace the shield onto the A5 board with the two long screws, and place it loosely in its card guide. Now you can make all the connections from the wiring you added to the new 20kHz filter board. Route the wiring forward and over the bottom of the meter movement (now facing you) so it won't be pinched when you replace the bottom cover. See *Fig. 4* for a summary of the wiring connections.

I grounded the input to the 30kHz filter circuit from J200-3 (red wire) on the perfboard so it won't pick up noise. There is no electrical connection to the brown wire from J200-2. I soldered it to a bare wire-wrap pin I placed on the perfboard. The black ground wire is twisted with the red 20kHz input wire from S11B for shielding. I also twisted the two supply wires together to cancel any electric fields.

With the soldering to the perfboard complete, plug the three RCA connections back into the A5 board, and check the other wiring, to be sure something hasn't come adrift. Finally, slide the A5 board into place and install the short retaining screw through the bottom of the A2 board near J203 (*Photos 6* and *7*). Reposition the Function switch shaft onto S9, align the knob with the OSC LEVEL selection, and tighten the set screw.

## MEASUREMENTS

Before replacing the covers, check the operation of the unit against your "before" data. Check all the meter functions. Next, with the internal oscillator output connected to the distortion analyzer input, and set for 3V RMS, run a voltage vs. frequency curve for your new 20kHz filter. It should look similar to *Fig. 5* (which also has the curves for the 400Hz HP, the 30kHz HP, the 80kHz HP, and the combined 30kHz and 80kHz HP filters in series). Check the 400Hz and 80kHz curves as well, to be sure these filters are still operating.

I repeated the THD measurements I made earlier on a modified Rotel CD player (*Fig. 6*). The solid line represents the THD using the HP 339A's 30kHz LP filter in series with the 80kHz LP filter. The new 20kHz LP filter (dashed line) smooths the large excursions in the THD measurements. These excursions were most likely caused by interaction with the 44.1kHz sampling frequency that passed through the lower-order filters to the meter RMS-DC converter. ❖

### SOURCE

Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
1-800-344-4539
www.digikey.com

### REFERENCE


1. "*Active-Filter Cookbook*," Don Lancaster, 1975, Howard Sams, ISBN 0-672-21168-8.

# Odyssey of a Line Stage, Pt. 2

This author's journey of discovery in the land of transformer-output line stages continues. **By David Davenport**


PHOTO 1: Line stage module. This is a single channel of the parafeed line stage prototype.

I had reached the point where I needed to decide whether to freeze the design and build what I had, or continue with my experiments. I was learning a lot and having fun, so I decided to continue experimenting.

Now the question was, which way to go? The obvious candidates were balanced push-pull or single-ended parafeed. I chose single-ended parafeed because I could incrementally move into a parafeed design, while balanced push-pull was a total re-design requiring a new transformer.

## PARAFEED PATH

The idea behind a parafeed, or parallel feed, circuit design is to separate the AC audio signal from the DC power current path. The transformer is in parallel rather than in series with the driver so no direct current flows through the transformer. I don't pretend to know the first thing about transformer design, but from what I can gather, not having to cope with direct current makes the design job easier. There are fewer trade-offs to make, and it is easier to optimize the AC signal characteristics.

But, what about using a "regular" transformer in a parafeed configuration? I built the circuit shown in *Fig. 6* with the LL1660/18mA to find out. For some reason—I'm not sure why—I decided to lower the current in the circuit to 10mA and use a 7V bias point. Since there is no current flowing through the transformer, I don't think it makes much difference.

The 3μF "parafeed capacitor" in series with the transformer primary ensures that no DC current flows in the primary. Because of the constant 10mA of current flowing through the tube and cathode bias resistor, there is no need for a bypass capacitor.

## CAPACITOR VALUE

I had great expectations for this circuit and was sorely disappointed with the sound it produced. The character of the sound was okay, but the bass was anemic—puny, as we say in Dixie. Upon examining the frequency response in the bass region, I found a large peak at 9Hz with a corresponding suck-out in the midbass. The value of the parafeed capacitor directly affects the bass response of the circuit.

There are several ways to calculate the value of the parafeed capacitor. Some set the value so that the impedance of the capacitor is equal to the impedance of the transformer primary at some arbitrarily low frequency, say 10Hz. Another method sets the value so that the electrical Q of the resonant circuit is well behaved.

Unfortunately, the different methods give different results, in this case ranging an order of magnitude from 2.5μF to 25μF. I had hoped to get lucky and went with the low value, but it didn't work out. I built the circuit shown in *Fig. 7* not only to determine what value of parafeed capacitor to use, but also to test the result of a cathode-coupled connection and provide the capability for comparing different transformers.

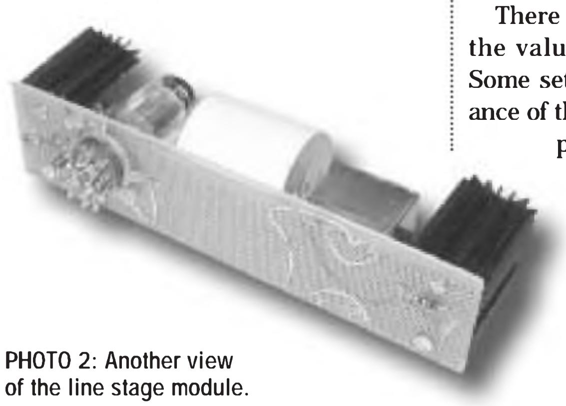


PHOTO 2: Another view of the line stage module.

But let's take one step at a time. For determining the optimal value of the parafeed capacitor, I used electrical measurements as well as listening evaluation and found that there was agreement between the two. The optimal value provides a broad, low-amplitude peak in the infrasonic region that exactly compensates for the low-frequency rolloff, resulting in a composite flat response down to 3 or 4Hz. Furthermore, there is no sonic advantage to using a larger capacitor than is needed; all that happens is a rolloff occurs in the subaudible region with little if any effect on the midbass.

Interestingly, in determining optimal values for the parafeed capacitor to be used with several different transformers, I found the range to be 5μF to 13μF. Because this parafeed capacitor is in the signal path, it must be of the highest quality. I recommend your favorite film and foil. The particular brand of capacitor will affect the character of the sound, but this is a matter of your taste and preference. You'll need to be con-

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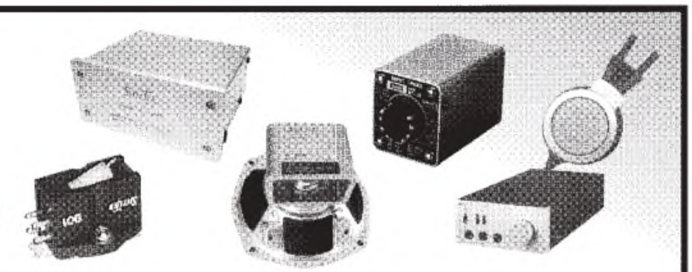

## ■Cartridge

| Model | Price (US$) |
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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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| Area III $27 | North America Oceania Europe |
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## ■Japanease Audio Book

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| Title | | Price(US $) |
|---|---|---|
| Attractive Tube Amps Vol. 1&2 | (Isamu Asano) | 30 each |
| The Joy of Vintage Tube Amps 1&2 | (Tadaatsu Atarashi) | 30 each NEW |
| Direct & Indirect Tube Amps | (Kiyokazu Matsunami) | 40 NEW |
| SE Amps by Transmitting Tubes | (Kouichi Shishido) | 50 |
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| Classic Valve | (Hisashi Ohtsuka) | 40 |
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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 |

**Air Economy

## ■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 |

*Price is for a pair **Air Economy

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

| Model | Specifications | | | | Price (US $) | Postage** (US $) | | | |
|---|---|---|---|---|---|---|---|---|---|
| | W | Pri.Imp(kΩ) | Freq Response | Application | | I | II | III | IV |
| XE-20S (SE OPT) | 20 | 2.5, 3.5, 5 | 20Hz~90kHz | 300B,50,2A3 | 396 | 47 | 56 | 84 | 113 |
| 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 |

Price is for a Pair

**Air Economy

## ■TAMURA TRANS (All models are available)

| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| F-7002 (Permalloy) | 10 | 3.5 | 15Hz~50kHz | 300B,50 | 836 | 60 | 70 | 110 | 145 |
| 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 |

Price is for a Pair

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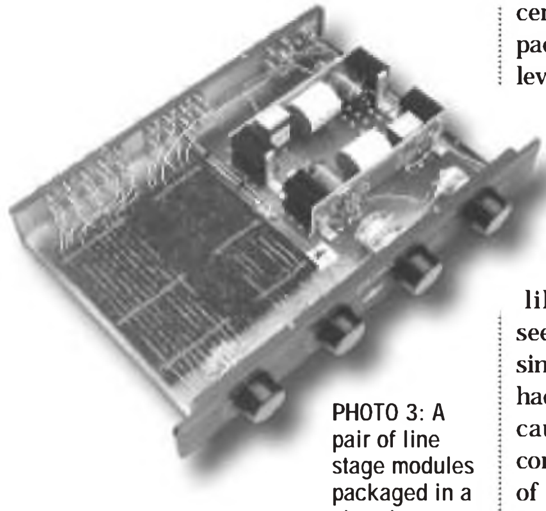

PHOTO 3: A pair of line stage modules packaged in a chassis.

cerned with the voltage rating of the capacitor because it will see the whole B+ level.

## TRANSFORMER COMPARISONS

With the optimal value of parafeed capacitor in the circuit, the bass was now adequate—not as tight and deep as I would have liked, but not puny like before. It seemed that the bass was deeper in the single-ended configuration. I no longer had the luxury of direct comparison because I needed to radically alter the configuration, and it had been a couple of days since I had heard the single-ended configuration. I wondered if this transformer having been designed for single-ended operation actually needed current running through it for optimal performance? Perhaps this would be resolved when I compared different transformers.

In spite of the bass being less than I would have desired, I knew that I was on the right track. The mid and high frequency was far superior to that of the single-ended configuration. In fact, it could be that my perception of a lack in the bass presentation was colored by comparison to the improvements in the mid and treble.

Switching the parafeed capacitor between ground and the cathode provided a noticeable improvement in the sound when connected to the cathode, or cathode-coupled. Because the top of the transformer is no longer connected to the power supply, there is no noise to feed to the cathode, and a bypass capacitor is not needed.

The constant-current source could be a very large resistor, a choke, or an active electronic circuit. I didn't bother with a resistor but compared several chokes and electronic circuits.

The clear winner is a very simple electronic circuit based on the Supertex DN2540N5 N-channel depletion-mode vertical DMOSFET. This device makes it almost as easy to build a high-voltage current regulator as it is to build a low-voltage current regulator with the ubiquitous LM317. In fact, the topology is the same—a single resistor in addition to the device. While there is a simple formula used to calculate the value of the resistor for the LM317, unfortunately there is no such formula for the DN2540; variations in the manufacturing process make this impossible.

However, it is simple enough to determine the resistor value experimentally. *Figure 8* shows you how. It is not necessary to get a particular current to two decimal places. There is not a hill of beans difference in this circuit between using 10mA and 11mA, so pick a value of current that corresponds to a convenient resistance—one that can be provided with a single resistor or parallel pair. In my circuit, about 150Ω provided about 10mA and I found that the value determined using a 9V battery held closely at 250V.

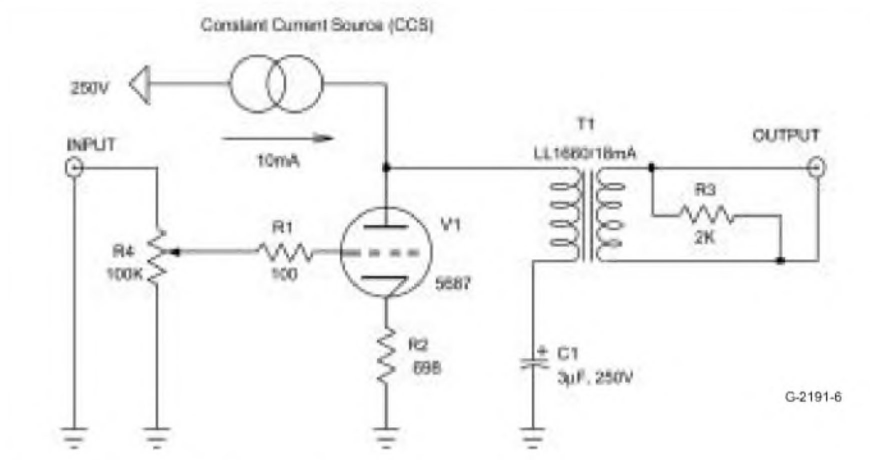


FIGURE 6: First parafeed prototype.

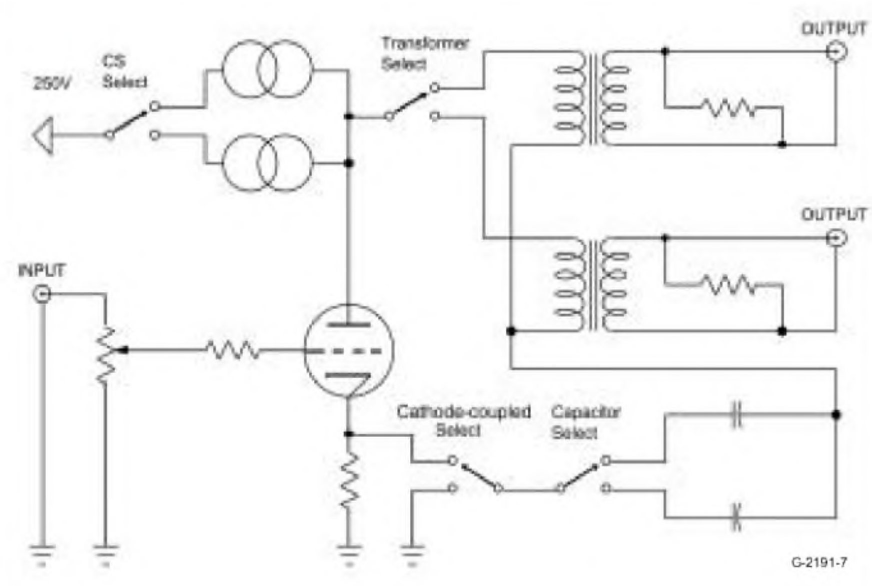


FIGURE 7: Parafeed test bed. Allows direct comparison of current sources, parafeed capacitors, and transformers as well as cathode-coupled connection. Several different devices of each type were inserted into the test bed and compared one-on-one with each other.

The DN2540N5 is a TO220 package, so you should mount it on a heatsink. The tab connected to the drain needs to be electrically isolated from the heatsink. Also, the DN2540 is prone to oscillation and needs the equivalent of a grid-stopper resistor. A non-inductive, 100Ω resistor connected as close as possible to the gate will suffice.

## EUREKA

As I mentioned earlier, I tried different

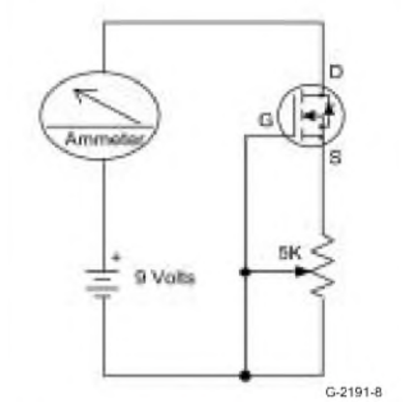


FIGURE 8: Determining the current-source resistor.

## TABLE 2
## PARTS LIST FOR *FIG. 9*

| PART | DESCRIPTION | SOURCE |
|---|---|---|
| **RESISTORS** | | |
| R1 | 100k DACT | Aloha Audio |
| R2, R5 | 100Ω ¼W non-inductive | Mouser 30BJ250-100 |
| R3 | (Note 2) | |
| R4 | 5k trimpot (Note 1) | Mouser 72-T93YB-5K |
| R6 | 698Ω Mills MRB-5 | Michael Percy |
| R7 | 3Ω 3W | Mouser 283-3 |
| R8 | 39Ω ¼W | Mouser 271-39 |
| R9 | (Note 3) | |
| **CAPACITORS** | | |
| C1 | 10µF 250V film and foil | Parts Connexion |
| C2 | 100nF 50V | Digi-Key P4923-ND |
| **INDUCTORS** | | |
| L1, L2 | EMI bead | Digi-Key P9817BK-ND |
| **MISCELLANEOUS** | | |
| T1 | Transformer, Lundahl LL1674 | K&K Audio |
| V1 | Tung-Sol 5687 | ATSI Electronics |
| Q1 | MOSFET DN2540N5 | K&K Audio |
| | LM317T | Digi-Key LM317T-ND |

**NOTES**
1. R4 set for 7V across R6.
2. R3 closest standard value to setting of R4.
3. R9 value chosen for preference.

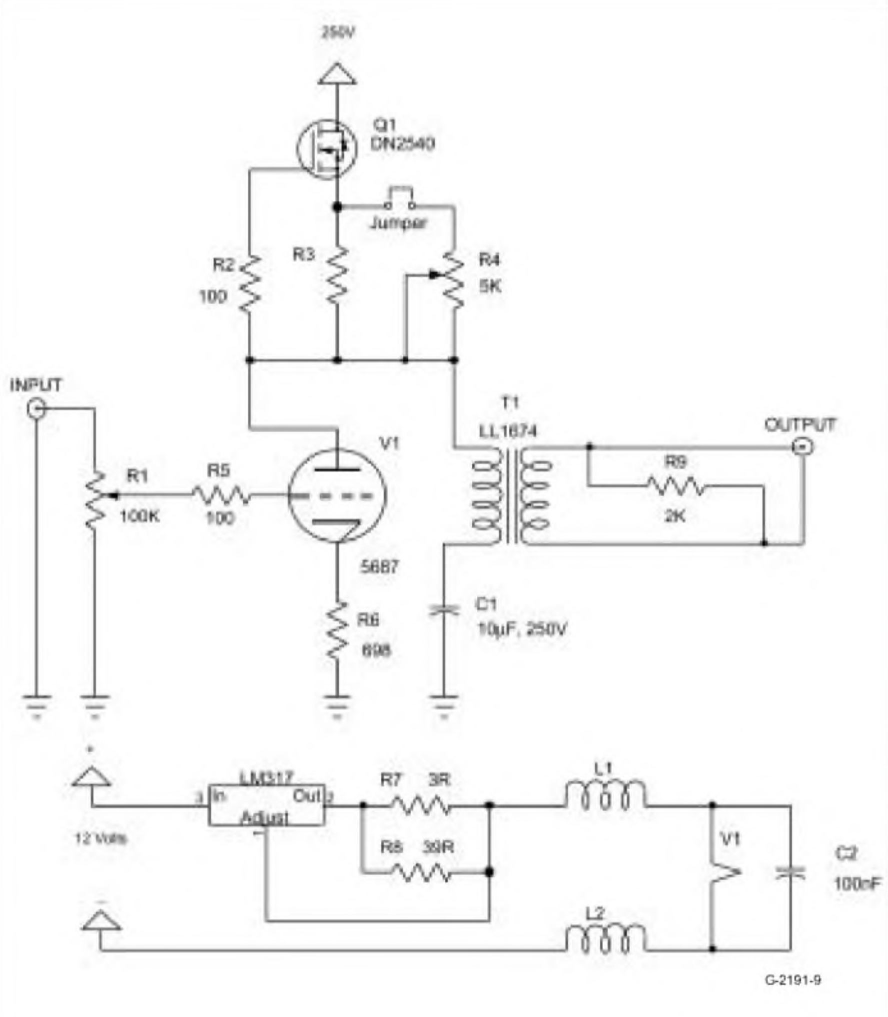


FIGURE 9: Closing in on the design. This is the first step in finalizing the design.

transformers—particularly transformers that were designed not to have direct current through them. Generally these transformers are smaller and have different core materials. From what I can gather, core material makes a big difference, such as Lundahl's amorphous core, with which the company has had great success in some of their smaller high-level line-input transformers.

Unfortunately, Lundahl does not yet have an amorphous core line-output transformer that they specifically recommend for parafeed connection. But then in my mind a transformer is a transformer, and if it can be used as a step-up input transformer, it can be used as a step-down output transformer. Of course, you can't put any current through a line-input transformer, but then that is exactly what I wanted. Kevin (from K&K Audio) had a pair of LL1674 line-input transformers in stock that he agreed to let me try in the circuit.

I put them in and was blown away by the sound. The music had taken on a new life. The bass was deep and solid, a characteristic shared with some other small transformers, but that the LL1660 had lacked; it was now cleaner and tighter than with any other transformer. Treble was more extended and the mid-range was really musical.

I bought the LL1674s on the spot, knowing that I had reached the end of my experimentation and was ready to build a representative prototype of the final package. The schematic of this design is shown in *Fig. 9*. The Lundahl data sheet mentions that no compensation network or termination is required for the LL1674, which is borne out by the very clean scope presentation and good sound. However, I did find that the sound improved subtly with a little load. You would want to do your own experiments on this.

As with many of Lundahl's transformers, you can choose different ratios between the input and output. I did not need or want much gain, so I chose the 8:1 ratio.

Bass response on the transformers is no deep mystery. It all boils down to primary inductance. The single-ended transformer is designed to allow substantial direct current, the push-pull transformer is designed to carry only enough direct current to account for circuit imbalances, and no direct current at all is allowed through a parafeed transformer.

So what has this to do with primary inductance? Well, the problem is that direct current causes a transformer to saturate, and a gap must be introduced in the core to compensate for this. A single-ended transformer has a large gap, a push-pull transformer has a small gap, and a parafeed transformer has no gap at all.

The gap in the core reduces the inductance—the larger the gap, the larger the reduction. So, all other things being equal, the parafeed transformer with no gap at all has the largest primary inductance, which allows it to exhibit the best distortion-free bass.

Many modern components designed for PC mounting make a project simpler to build compared to chassis mounting and point-to-point wiring. However, I have found that it is a good idea to build a prototype of a circuit on a perforated board before etching the printed circuit board. You can easily mount PC components and, instead of land patterns, you have the flexibility of point-to-point wiring, which is easy to change if needed. I think of them as small chassis.

## CONSTRUCTION

I have built many prototype digital circuits over the years, so I am comfortable using 30-gauge wire to interconnect the components, and some folks assert that the smaller the gauge wire, the better from a quality of sound viewpoint. I guess that I could have put everything on a single board, but for flexibility of packaging I chose instead to make a separate board for each channel. They are identical except for which half of the tube is used on each. This provides the ability to get a new tube when the tubes are swapped between boards. Both the LM317 and the DN2540 are TO220 packages that need to be mounted on a heatsink.

I am conservative when it comes to heatsinks, and the large ones that I used become only slightly warm to the touch. These heatsinks and the transformer are both about 1.7″ long, which determines the width of the board. The length of the board depends somewhat on the capacitor you pick for the parafeed. Some high-quality film and foil capacitors are 3 or 4″ long, although most are no wider than 1.7″. The length of the board needs to be about 5″ in addition to what is needed for the capacitor.

For the parafeed capacitor I used some old Wonder InfiniCaps that were the right size. My board was 1.7″ wide by 6.7″ long. *Photos 1* and *2* show the assembly. I am pleased with the resultant package, a compact module that can be easily mounted in a variety of mechanical enclosures.

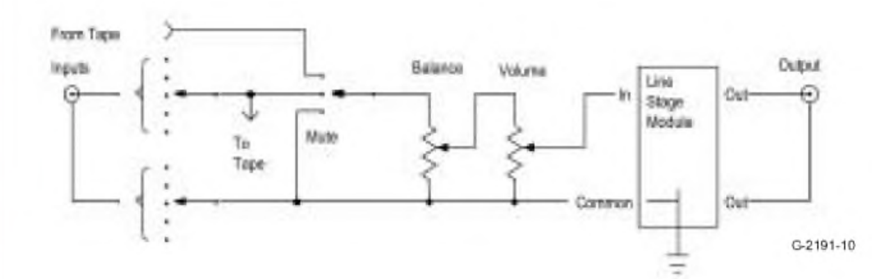


FIGURE 10: Schematic of the one channel of the completed line stage. The power supply is in a separate chassis that is connected to the line stage with an umbilical cable.

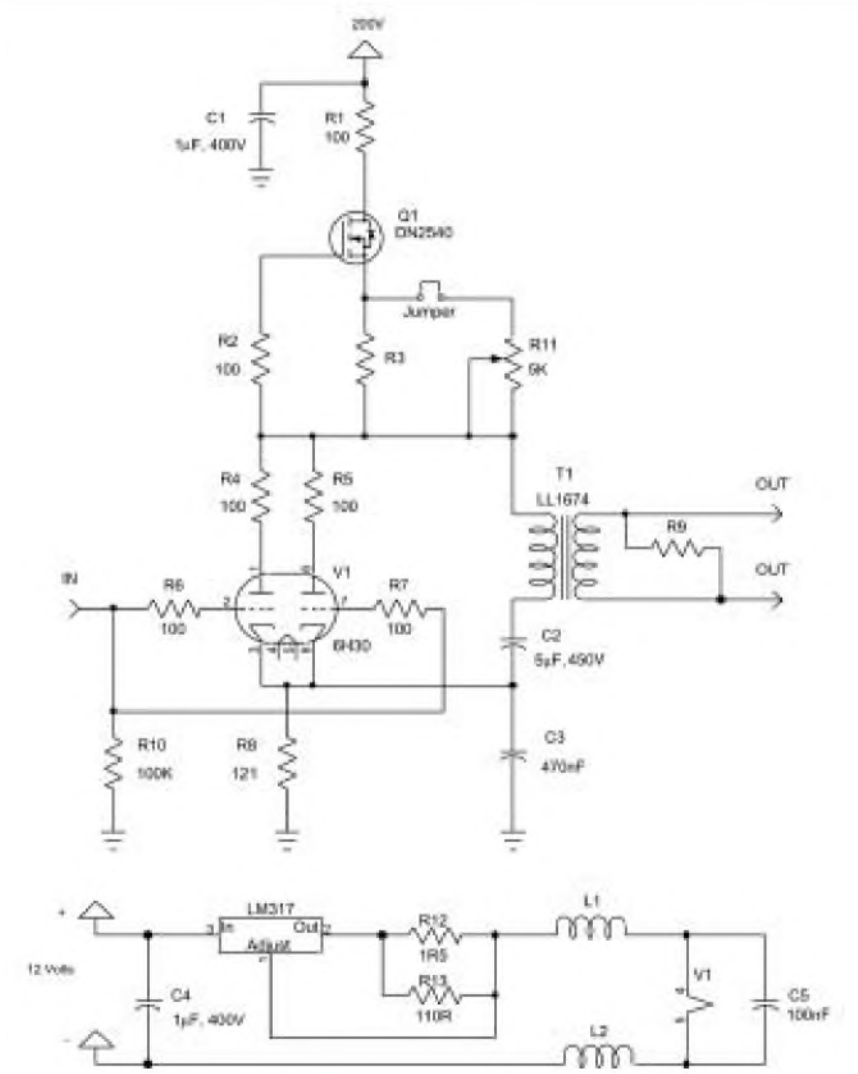


FIGURE 11: Final design for one channel of the line stage.

I had an old Mod Squad Line Drive passive preamplifier that fit the bill for inputs and switching, so I used it for my chassis. There probably is enough room for a power supply in the enclosure; however, I opted for an external supply for a couple of reasons. I am also designing a phono preamplifier, which most likely will need a stand-alone supply that could also be the supply for the line stage. Also, I still want to experiment further on the power supply, and this is easier with a separate supply. However, your situation may be different, and I'm sure that the line stage module will easily adapt to your needs.

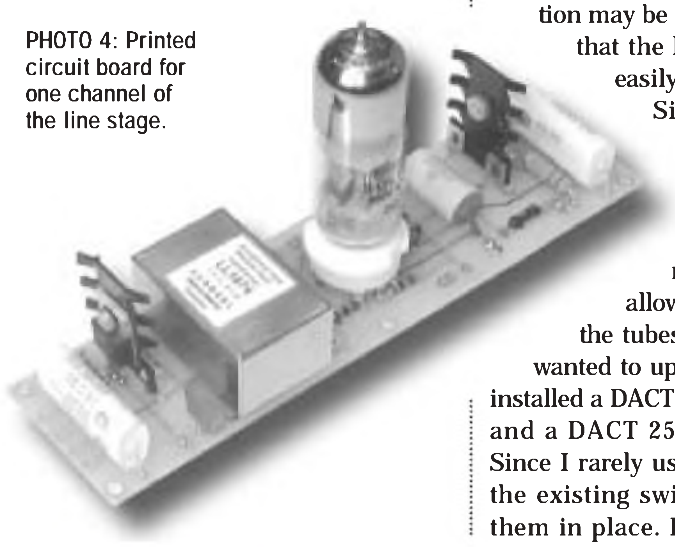

PHOTO 4: Printed circuit board for one channel of the line stage.

Since the Mod Squad Line Drive has no active circuitry to produce heat, I needed to drill a matrix of holes to allow for ventilation to cool the tubes and heatsinks. I also wanted to update the controls, so I installed a DACT 100kΩ volume control and a DACT 25kΩ balance control. Since I rarely use tape monitors, I left the existing switches and wiring for them in place. However, I wanted to switch both signals and commons for the inputs, so I replaced the original selector switch with a 4-pole Electroswitch. *Figure 10* shows the complete line stage schematic, and *Photo 3* shows the prototype line stage modules packaged in the chassis.

## COMING OUT

As luck would have it, the next meeting of the Piedmont Audio Society was held just after I had finished packaging the completed prototype boards. I wanted an independent assessment from a fresh set of ears, so I took the line stage along.

Well, how can I say it? The guys were kind in their lukewarm reception of the line stage. I was disappointed, but they were right in pointing out a couple of deficiencies that I had introduced in building the latest version. I had not done sufficient listening of my own and let them slip by.

First was a brittleness or hardness in piano crescendos which was definitely there—something I had not heard before I packaged the boards. Second was a buzz, or harsh hum under some conditions. It showed up only when the variable power supply was set at its maximum value of 250V and the volume control was at its minimum value. With the volume set to a normal listening level there was no audible buzz.

While reflecting on what I had changed in these last few steps, I realized that I had done all my earlier listening with a transformer ratio of 4:1 and then arbitrarily wired it up 8:1 for the new board because I did not need the gain. So I experimented with listening to the sound of the different ratios and found that the 2:1 ratio sounded best. The lower ratio cured the hardness in the piano crescendos. The cause of the buzz was harder to find.

## TROUBLESHOOTING

At first I thought that I had a ground problem, because even though I had been careful about a single-point star-ground system, you never know when it comes to grounds. I tried all the tricks in my book and nothing worked. In talking with Kevin about the problem, he mentioned that he had seen hums and buzzes caused by oscillations. I hadn't thought about oscillations be-

### TABLE 3
### PARTS LIST FOR *FIG. 11*

| PART | DESCRIPTION | SOURCE |
|---|---|---|
| **RESISTORS** | | |
| R1, R2, R4, R5, R6, R7 | 100Ω ¼W non-inductive | Mouser 30BJ250-100 |
| R3 | (Note 2) | |
| R8 | 121Ω Mills MRB-5 | Michael Percy |
| R9 | (Note 3) | |
| R10 | 100kΩ ¼W (Note 4) | Mouser 271-100K |
| R11 | 5k trimpot (Note 1) | Mouser 72-T93YB-5K |
| R12 | 1.5Ω 3W | Mouser 283-1.5 |
| R13 | 110Ω ¼W | Mouser 271-110 |
| **CAPACITORS** | | |
| C1, C4 | 1µF 400V GE Film | Allied Electronics 591-8015 |
| C2 | 5µF Kimber Kap | Parts Connexion |
| C3 | 470nF Kimber Kap | Parts Connexion |
| C5 | 100nF 50V | Digi-Key P4923-ND |
| **INDUCTORS** | | |
| L1, L2 | EMI bead | Digi-Key P9817BK-ND |
| **MISCELLANEOUS** | | |
| T1 | Transformer, Lundahl LL1674 | K&K Audio |
| V1 | Sovtek 6H30 | Triode Electronics |
| Q1 | MOSFET DN2540N5 | K&K Audio |
| | LM317T | Digi-Key LM317T-ND |
| | PC board | K&K Audio |
| | 9-pin tube socket | Triode Electronics |
| | Heatsink (qty 2) | Digi-Key HS241-ND |
| | Insulator, heatsink (qty 2) | Digi-Key BER102-ND |
| | Insulating bushing (qty 2) | Digi-Key 3049K-ND |
| | 4-40 screw and nut (2 ea.) | Procure locally |
| | Mounting hardware | Procure locally |

**NOTES**
1. R11 set for 4V across R1.
2. R3 closest standard value to setting of R11.
3. R9 value chosen for preference.
4. R10 present only when there is no external volume control.
5. A kit is available from K&K Audio and other selected Lundahl distributors.

cause I had a gate-stopper resistor on the DN2540 and a similar grid-stopper resistor on the 5687, as well as ferrite beads and a capacitor on its filament.

I thought I had it covered, but I was grasping at straws and ready to consider anything. I changed my tactics and started to get results. I found that when I hung my scope probe on the drain of the MOSFET the buzz stopped. However, when I connected the scope to the output of the power supply in the separate power-supply chassis, there was no effect. The two points were electrically the same; the only thing separating them was a few feet of wire.

Okay, this would be easy—simply substitute a capacitor for the scope probe. Nope. It wasn't giving up that easily.

Next I tried a ferrite bead on the high-voltage wire without success. But I did notice something interesting. When I moved the high-voltage wire to install the bead, the nature of the buzz had changed. I found that I could control the loudness of the buzz by moving the wire.

I kicked myself—I had seen this before. In contrast to the present good-looking assembly, the original test beds were a rat's nest of wires. Now everything was neat, with wires laid down and dressed. I hadn't paid attention to what I was doing and had inputs neatly running together with both outputs and power-supply wires for several inches.

It was the distributed capacitance between the wires that got me. I re-wired the signals and high-voltage wires with shielded wires, taking care to keep everything separated. This not only fixed the buzz, but also improved the overall character of the sound. High-level transients in the music had driven the circuit into momentary oscillation. This was the hardness in the sound that we heard.

While changing the transformer ratio softened the problem, it wasn't the fix. I listened to the different ratios and found that although they sounded different, there was not a problem with any of them. Difference now was a matter of taste. The 2:1 ratio was fuller, more voluminous, and a little softer. The 8:1 ratio had firmer and more bass and the sound was generally more tightly controlled. The 4:1 ratio was in between.

Earlier I had listened to various

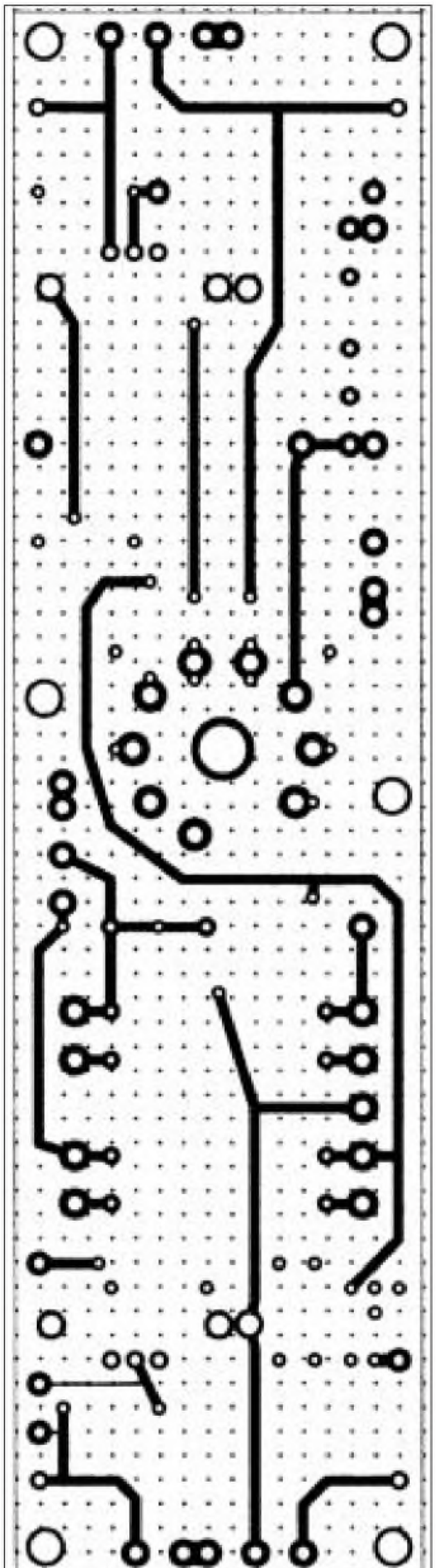

FIGURE 12: Top layer of printed circuit board. G-2191-12

5687s and several variants but had not auditioned either the ECC99 or the 6H30, both of which I had heard good things about. I had good success with my earlier method of swapping tubes without a protracted delay between lis-

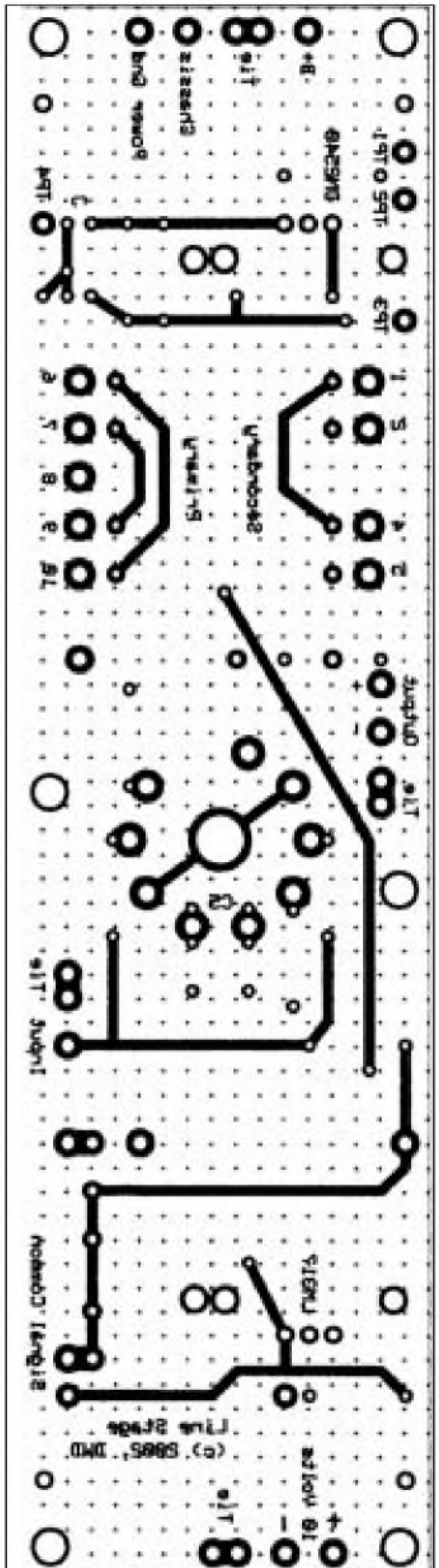

FIGURE 13: Bottom layer of printed circuit board. G-2191-13

tening sessions, so I wanted to use that procedure again. Unfortunately, these tubes have a pin configuration different from the 5687. The ECC99 has the same pin arrangement as the 12AU7, while the 6H30 is like the 6DJ8.

However, upon closer study there are some similarities in pins 1 through 5 that I could take advantage of. Pins 1 through 3 are a triode with the same pin assignments for each tube. Pins 4 and 5 are filament pins for each tube. I was only using one-half of a tube for each channel, so the fact that pins 6 through 9 were used differently for the second triode did not matter. Since I was using a current regulator for the filament, the nominal voltage did not matter, and all I had to do was rig a switch to control the amount of current provided.

It worked like a charm. There is a marked difference in the sound of the three tubes. Compared to the 5687, the ECC99 objectively has a little better detail and foundation. The big difference is that it sounds much more powerful, expressive, and lively. However, it is almost bigger than life, and I wondered whether the presentation would wear thin over time.

This is my subjective assessment, and I must say that a lot of people like this tube. The only objective downside is that my pair of tubes is slightly microphonic. In contrast, the 6H30 is more tightly controlled, provides a presentation with less fanfare, and is less dramatic than the ECC99, but is perhaps more realistic.

I really like this tube, which does everything well. I tried several variations of bias point and current and settled on 5V and 20mA. With the 5687 and ECC99 I could wire each channel to use a different half of the tube and provide the ability to get a fresh tube by swapping the tube between channels. Unfortunately, you can't do this with the 6H30 because there is a single, shared filament.

Rather than waste half a tube, I tried both triodes wired in parallel, which provided an even better presentation. First, I simply wired the two plates together as well as the two grids and the two cathodes. The circuit immediately went into oscillation, which was easy to recognize from my earlier experience. There was a grid stopper, but the extra fraction of an inch between the two grids was enough to cause a problem.

Although adding the second grid stopper fixed the problem, I did not stop there. I also added plate stoppers and a drain stopper in addition to the gate stopper on the CCS. I also added capacitors across the two power supplies and the cathode bias resistor. I expect that these precautions would have been valuable even if I had not paralleled the two sections. I ended up with the circuit shown in *Fig. 11* with 40mA shared between the two triodes at 4.85V bias.

As part of the final design process I re-assessed the parafeed capacitor. For this circuit the optimal value is 5µF. As before, use a high-quality film and foil capacitor that suits your taste. My preference is the Kimber Kap.

Happy with the design, I proceeded to lay out and etch the printed circuit board. I reduced the size of the heatsinks to be more in line with what is needed and added some holes for mounting the board as well as tie points for cable shields if needed. I wrestled with what to do about the parafeed capacitor. It took up a large portion of the board, and I wanted to make provisions for a variety of different capacitors.

My solution was to provide maximum flexibility by moving the capacitor off-board, allowing for connection to tie points on the edge of the board. The final board is shown in *Photo 4*. Layout of the board is shown in *Figs. 12* and *13*.

## SUMMARY

- A transformer-output line stage sounds very good, with parafeed taking it to the ultimate.
- The best transformer I have found for this purpose is the Lundahl LL1674.
- The preferred current source is the Supertex DN2540 FET.
- We listened to the effects of different filament supplies and found that we preferred the current-regulated DC and unregulated AC.
- Fixed-bias was compared to cathode-bias and we preferred the cathode-bias.
- Cathode-coupled provided a noticeable improvement in each case.
- Although the 5687 was good and the ECC99 provided a viable alternative, to my ears the 6H30 is the tube of choice.
- The circuit is capable of oscillation, and therefore you'll need to be careful with wire layout.
- Several items are open to personal preference: for example, the type of parafeed capacitor used, and the amount of load on the transformer secondary.
- Feel free to use these results as a starting place for your own next journey. Bon voyage! ❖

# Bass in Small Places

This author explores the relationship between room size and frequency performance. **By Dick Pierce**

A common and pervasive myth about loudspeakers goes like this: it is impossible to produce low-frequency bass in a small room. The reason, stated almost axiomatically, is because the room is too small to hold the long wavelengths.

That this is a myth without physical foundation is easy to demonstrate on several fronts. For one, if the axiomatic basis were to hold, then by the same axiom, you should not be able to hear any frequencies below several kilohertz, because these same large wavelengths could not fit within the small confines of the ear canal and thus could not excite the eardrum into motion. This article demonstrates the fallacy of this argument more analytically, and uses an acoustic measurement device as an existence proof that you can support low frequencies in small enclosures.

## A CASE STUDY: THE PISTONPHONE ACOUSTIC CALIBRATOR

The clever Danish acoustics instrument manufacturer, Brüel & Kjaer, produces a device, the type 4220 PistonPhone, which is used in the absolute calibration of microphones. The operating principle behind the PistonPhone is conceptually simple: a sealed enclosure of volume V whose volume is varied periodically by a change in volume $\Delta v$ will produce a sound level of:

$$P = \gamma \times P_0 \times \frac{\Delta v}{V} \text{ dynes/cm}^2$$

where:

$\gamma$ = the ratio of specific heats for the gas in the enclosure. For air at 20°C and at 1 atmosphere, $\gamma = 1.402$.

$P_0$ = atmospheric pressure, $1.0133 \times 10^6$ dynes/cm$^2$

$\Delta v$ = the change in volume of the enclosure

V = the total volume of the enclosure

The 4220 PistonPhone is implemented as two opposed cam-driven pistons, approximately 4mm in diameter, that work into a closed volume of approximately 19cm$^3$ (about the volume of an average ice cube). The pistons are each driven by a four-lobed cam with a sinusoidal profile, which is driven in turn by a small speed-governed motor revolving at 3750 RPM. The result is that the volume of the chamber varies sinusoidally at a rate of 250Hz (3750/60 × 4).

Now, the chamber itself is quite small. Brüel & Kjaer claims, as mentioned, that the "coupling chamber" has a volume of 19cm$^3$. Its linear dimensions are that of a cylinder approximately 3.19cm in diameter with a depth of 2.63cm. Within that volume is the piston housing itself, which occupies a small portion of the total volume.

The cam has a maximum radius of about 0.55cm and a minimum radius of 0.5cm. This means the total stroke of each of the two pistons is 0.05cm. The resulting displacement, or change in volume, is then

$$\Delta v = \pi \frac{d^2}{4} s$$

where d is the diameter of the piston, and s is the stroke of the piston. In the case of the measurements here, you get:

$$\Delta v = \pi \frac{(4 \times 10^{-3} m)^2}{4} \times 5 \times 10^{-4} m$$

$$\Delta v = \pi \frac{16 \times 10^{-6} m^2}{4} \times 5 \times 10^{-4} m$$

$$\Delta v = 6.28 \times 10^{-9} m^3$$

Six *billionths* of a cubic meter is a small volume, to be sure. But it is that six billionths of a cubic meter change that produces the 124dB sound pressure level. Looking at it another way, varying the volume of a sealed enclosure by:

$$\frac{6.28 \times 10^{-9} m^3}{1.9 \times 10^{-5} m^3} = 0.00033$$

a mere 0.03% change in volume! Putting it all together, you come up with a corresponding change in pressure of:

$$P = 1.402 \times 1.0133 \times 10^6 \text{ dynes/cm}^2 \times$$

$$\frac{6.28 \times 10^{-9} m^2}{1.9 \times 10^{-5} m^3} = 470 \text{ dynes/cm}^2$$

Sound pressure level in dB SPL is calculated as:

$$\text{dB SPL} = 20 \log_{10} \frac{P}{P_{REF}}$$

where $P_{REF}$ is the 0dB SPL reference sound pressure level corresponding to 0.0002 dynes/cm$^2$. In this case:

$$\text{dB SPL} = 20 \log_{10} \frac{470 \text{ dynes/cm}^2}{0.0002 \text{ dynes/cm}^2}$$

This corresponds to a sound pressure level of about 127dB and is the *peak* sound level produced. Since the cam profile and thus the waveform produced is sinusoidal, the equivalent RMS sound pressure level is 3dB less, or 124dB, essentially what Brüel & Kjaer claims.

The difficulty with reconciling this result with the notion that you must be able to fit a wavelength within the chamber is that at 250Hz the wavelength is about 1.36 meters (4½′), while the maximum linear dimension of the chamber is a mere 0.03 meters (1¼″). According to the oft-held theory stated previously, this chamber is far too small. Not by a little, but by a factor of over 40! The theory suggests that such a small chamber could not support any bass less than 10kHz.

By simply varying the drive voltage to the motor, you can change its rotational speed. When you slow the motor down so that it produces frequencies below 250Hz, you notice something very interesting: the sound pressure level actually measured in that little chamber *does not change*! Even lowering the speed of the motor corresponding to a frequency of 15Hz, you find that the sound pressure is the same 124dB.

Indeed, you can continue lowering the frequency and, as long as the response of the microphone itself holds out, you see no reduction in the sound pressure level until you get to a frequency low enough where tiny air leaks in the chamber start to dominate. Seal the chamber tight enough, and these leaks may not be significant until below a fraction of a hertz.

## WHAT IS "SOUND"?

For the purposes of this analysis, sound is the physical phenomenon of varying air pressure to a sufficient amplitude and at an appropriate frequency such that you can detect it with your ears. If you make the pressure different on one side of the eardrum versus the other, the eardrum moves in response, and, eventually, this fact is communicated to the brain and, maybe, interpreted. All that is needed is a change in pressure between the outer ear and the inner ear. There are limits to the frequency at which the external pressure changes result in differential pressure between the inner and outer ear.

Built into the ear is its own low-frequency "leak," the Eustachian tube. This small tube communicates air between the chamber behind the eardrum and the back of the throat. One of its purposes is to make sure that very low-frequency changes in pressure do not cause movement of the eardrum. These low-frequency pressure changes would, for example, include normal variations in atmospheric pressure. It's reasonable to assume that the Eustachian tube is an evolutionary response to such pressure variations: if the tube is blocked, it can lead to ear pain due to pressure variations.

Conveniently, evolution has inadvertently foreseen the role of the Eustachian tube in helping us cope with modern-day airplane travel, which causes rather large external pressure changes in a relatively short time. This is why flying can be very uncomfortable if you have a head cold: inflammation and mucus can plug the Eustachian tube, preventing pressure equalization.

That being said, any mechanism that results in a time-varying change in pressure between the outer and inner ear of sufficient magnitude in frequency will result in the potential perception of sound. It doesn't make any difference whether that pressure change is due to the passing of a wave of pressure higher or lower than ambient, or simply a raising or lowering of the overall pressure in the surrounding airspace: as far as the ear is concerned (at low frequencies, at least), they are indistinguishable. This is why headphones, for example, can convey real low-frequency information: they simply cause a difference in pressure between the outer and inner ear.

## BASS IN SMALL ROOMS

So, now we have a physical basis for refuting the notion that you cannot have low bass in small rooms. Can we show what, in fact, the effect is in reality?

Yes, however, there are some assumptions to consider. Note that the previous discussion, especially the PistonPhone analysis, assumes:

1. That the "room" is sealed such that any leaks affect only the very low-frequency behavior of the enclosure. The smaller the leaks, the longer it takes for the room to equalize (the longer the time constant of the room), the lower the cutoff frequency.
2. That the size of the enclosure is, indeed, substantially smaller than the wavelength you are attempting to reproduce.

This last requirement is very important, because in order for the phenomenon to work as it does in the case of the PistonPhone or a headphone, you must be pressurizing the chamber, in this case the room, as a whole. When the wavelengths become small compared to the size of the room, then you have some areas where the pressure is higher than others, the wave behavior of the room begins to dominate, the room begins to look more and more like a very large (maybe even "infinite") space, and overall pressurization is not possible.

What does this mean in the context of our speakers and our rooms? We can, in fact, draw some interesting assertions about achieving low-bass performance with this knowledge in hand.

Remember that the sound pressure level, where the room is reasonably tightly sealed and the wavelengths are long compared to the room dimensions, is dependent solely on the change in volume of the room (given normal air at normal pressure and temperature). That means that you can fairly easily calculate the sound level possible under these conditions from a pair of woofers of a given diameter and a given excursion capability.

Simply, the change in volume is the so-called total displacement volume $V_D$ of the woofers: that's the product of the linear excursion $X_{MAX}$ and the emissive diameter $S_D$ of the drivers. Plug those numbers into the previous equation, along with the room volume, and you now can determine the sound pressure level capabilities of the speaker system.

For example, imagine a 12″ woofer with a peak $X_{MAX}$ of ½″. In metric terms, that results in a displacement volume of about $6 \times 10^{-4}m^3$. Now, imagine a room with a volume of $31m^3$ (corresponding to a room with dimensions of 14′ × 10′ × 8′ high). Using the previous equation, one such woofer is capable of generating a sound pressure level of:

$$P = \gamma \times P_0 \times \frac{\Delta v}{V} \text{ dynes/cm}^2$$

$$P = 1.402 \times 1.0133 \times 10^6 \times \frac{6 \times 10^{-4}\,\text{m}^3}{3.1 \times 10^1\,\text{m}} =$$

$$27.5 \text{ dynes/cm}^2$$

Referenced to 0.0002 dynes/cm$^2$, this corresponds to a sound pressure level of:

$$\text{SPL} = 20\log_{10}\frac{27.5 \text{ dyne/cm}^2}{0.0002 \text{ dyne/cm}^2} =$$

$$20\log_{10} 1.38 \times 10^5 = 102.7\text{dB}$$

103dB SPL is a respectable sound level, especially at such low frequencies from a single woofer. For two such woofers operating in phase, the limit resulting from the doubling in displacement volume would be

$$P = 1.402 \times 1.0133 \times 10^6 \times \frac{12 \times 10^{-4}\,\text{m}^3}{3.1 \times 10^1\,\text{m}} =$$

$$55 \text{ dynes/cm}^2$$

$$\text{SPL} = 20\log_{10}\frac{55 \text{ dyne/cm}^2}{0.0002 \text{ dyne/cm}^2} =$$

$$20\log_{10} 2.75 \times 10^5 = 108.8\text{dB}$$

Note that these figures are the peak sound levels; the RMS sound levels will be about 3dB lower than these numbers. But that still leaves a respectable 106dB capability.

The implications of this are important: below some frequency determined by the relationship between the wavelength at that frequency and the maximum dimensions of the room, the bass frequency performance of the speaker and room, considered as a system, is dependent only on the ability of the cone to compress the air, and is independent of frequency! The limit to producing sound level is simply determined by the ratio of the displacement volume of the woofer and the volume of the room.

### DIRECT RADIATOR LOUDSPEAKER BEHAVIOR

Contrast this to the behavior required of the woofer when the wavelengths are small compared to the room dimensions, or the so-called "free-field" conditions. Here, the excursion of the cone must go as the inverse square of the frequency in order to maintain a flat frequency response. This condition is satisfied mechanically by drivers operating in the so-called mass-controlled region of operation, above fundamental mechanical resonance.

Below resonance, the system is operating in the so-called stiffness controlled region, and the excursion no longer goes as the inverse square of the frequency; rather, the excursion is constant with frequency. This is precisely the behavior needed for flat frequency response in the pressurized room case!

It would seem, then, that you need to achieve a system where the room dimensions and system resonance are coordinated in a way such that above system resonance, the system sees near free-field conditions, and below resonance, it is working to pressurize the room as a whole. That directly suggests that there is a relationship between room size and system resonance for the extended low-frequency performance you are trying to achieve. But what are the room dimension requirements?

The general assumption is that the largest room dimension must be smaller than one-quarter the wavelength of the frequency for the pressure conditions to hold (and again, this also assumes that the room is reasonably well sealed such that the cutoff due to leakage is at a significantly lower frequency). That means, for example, for a speaker system with a 30Hz system resonance, that the room's largest dimension cannot exceed one-quarter wavelength at 30Hz. That wavelength is one-quarter of 342m/ sec/30Hz = 2.85 meters, or 9½′.

Now, 9½′ is not a big room (though it is a big car!). Consider, instead, a system cutoff of 20Hz, where the required room size is now relaxed to 4.3 meters, or 14′.

Another caveat is a corollary of the requirement that the room cannot have any significant leaks: the speaker enclosure itself cannot "leak" back into the room you are attempting to pressurize. This means that sealed box/acoustic suspension systems will work, but bass reflex or dipole systems will not. The rear pressurization of these speakers is communicated directly into the room, and the speaker is incapable of generating the required differential pressures below the vent/enclosure resonant frequency. This is because the rear "chamber" is communicating directly to the room in such speakers at very low frequencies.

Now, lest you leap at the assumption that the ultimate capabilities of the system are without limit, at least as far as low-frequency limit is concerned, look at some of the other assumptions and see whether they are true.

Certainly, there is the limit imposed by the basic low-frequency time constant of the room and whatever leaks may be present in the room. However, another assumption is that the basic mechanism of how the air itself operates is unchanged at arbitrarily low frequencies. It is assumed that compressions and expansions of the kind normally encountered in sound are *adiabatic* in nature. This means that the total energy of the system remains constant.

The effect of this is that the temperature of the air varies as an inverse function of the volume, and as a direct function of the pressure. Compress the air, raise the pressure, and the temperature increases. Lower the pressure, and it decreases. But if something robs heat from the air as the temperature is raised or adds it as it is lowered, then the total energy is not constant. The temperature fluctuations are smaller and the conditions are no longer adiabatic, but isothermal (meaning "same temperature").

The adiabatic behavior of air is certainly true at normal audio frequencies. However, at very low frequencies, this may not be the case, because there can exist mechanisms that will tend to equalize the temperature and attempt to bring the system back to thermal equilibrium, i.e., approaching isothermal conditions.

Imagine, for example, the pressure varying slowly enough that the temperature of the walls tends to start to control the temperature of the air. Of course, this is not going to happen until you get to a very low frequency, but the example does illustrate that the principle is not without some fundamental limits in the very nature of the mechanisms behind sound.

## CAVEATS

In addition to meeting the fundamental requirement outlined previously, there are other considerations at work. Well above both resonance and the frequency where the wavelengths are proximal to the room size, the speaker is operating under semi-free-field conditions, and the $1/f^2$ excursion behavior of the driver ensures flat frequency response into the room. Below system resonance and below the frequency where the wavelength is proximal to room size, the pressurization effect and the frequency-independent excursion behavior of the woofer work to ensure flat frequency response into the room.

It is the region between these two that becomes problematic. This is the region where the response of the speaker/room is often dominated by narrow-band, standing-wave induced phenomenon. It is the frequency where the wavelengths are the same size (roughly) as the room, and the speaker/room system is no longer behaving as either a free-field system or a pressurized system. Rather, it is a complex resonant system.

## CONCLUSION

I have demonstrated that it is possible to generate audible very low-frequency sound in enclosures of arbitrarily small volume, utilizing the ability of a loudspeaker or other transducer to produce overall pressure changes in the room. The behavior requires that the room dimensions be smaller than the wavelength of sound being produced and that the time constant of the room resulting from leaks be large compared to the frequency being produced. I've also shown that it is possible to integrate the response of the system at higher frequencies with that at lower by considering the relative placement of room size and system resonance/cutoff frequency using sealed box systems.

Whether such information has practical application is another matter. Given practical sealed box systems (practical defined by reasonable enclosure volume and system efficiency needed to achieve a low enough cutoff frequency), the limitations then imposed on room size may be too restrictive to be usable.

However, given the small dimensions in automobiles, it is possible to utilize this low-frequency loudspeaker and "room" pressurization behavior to effectively extend the response of the system to very low frequencies, now limited only by the time constant of the leaks in the car. This is one reason why the auto sound industry has, even though possibly inadvertently, achieved the ability to produce phenomenally high sound pressure levels at very low frequencies.

The oft-obnoxious car whose incessant booming can be heard blocks away, while not the best existence proof of the concept, is unfortunate and compelling evidence of its efficacy. The only consolation is that as bad as it sounds to us hapless pedestrians, the result is far worse for the occupants of the vehicle, as we are lucky enough to be in the free-field response, while they are stuck in the pressurized response. ❖

# Passive Phono Preamps: Design Constraints

This series continues, with a look at noise, signal level, and input overload concerns when designing a passive phono preamp.

By Paul J. Stamler

In an earlier article[1] I discussed the advantages of passive equalization networks in phono preamps, with special attention to playing records with non-RIAA curves. To recap briefly, the passive design employs a resistor-capacitor network between two gain blocks with flat frequency responses (*Fig. 1*); it generates a curve with the generalized form shown in *Fig. 2*.

There are several constraints on the design of a passive preamp; the two most important are overload and noise.

## HOW HIGH THE GAIN?

First, let me talk about "nominal levels." Harking back to old analog tape days, think of a VU meter. It's an average-reading device—that is, it measures the level of an audio signal averaged over a fairly long period, ignoring momentary peaks. While it isn't perfect, the consensus in pro audio is that average levels, as measured on a VU meter that meets ANSI specs on ballistics, correlate pretty well with perceived loudness. Peak-reading meters, on the other hand, have very little to do with perceived loudness, but everything to do with overloaded recordings (especially in the digital world, where overload is neither gentle nor kind).

A "nominal level," as I use the term, is one that correlates with a "0 VU" level, as measured on an averaging meter. Typically, a nominal-level signal will have a certain amount of headroom above it, and a noise floor well below it.

There are several nominal levels currently used in audio, but for us the two important ones are −10dBV and +4dBu. The latter corresponds to 1.225V RMS into an undefined (but presumed highish) impedance; this is standard "pro" level. Most pro-level signals are balanced.

A more common level in home audio systems and home studios is −10dBV, or 316mV RMS, usually unbalanced, again into an undefined impedance.

**TABLE 1**
**SIGNAL AND SCRATCH LEVELS**

| ARTIST | TITLE | AVERAGE LEVEL (REF. 5CM/S) | AMPLIFIED PEAK (REF. 5CM/SEC) | RAW PEAK (dBU) | LABEL | REMARKS |
|---|---|---|---|---|---|---|
| Julian Rose | Levinsky at the Wedding | −1 | +19 | | Columbia | Acoustic, 1917 |
| Joe Hayman | Cohen on the Telephone | −0.5 | +21 | | Columbia | Acoustic, c. 1913 |
| | (ditto, worst bangers) | −0.5 | +24.5 | −19 | | |
| Josef Rosenblatt | ? (Hebrew letters only) | 0 | +21 | | Victor | Acoustic |
| "Mac" McClintock | The Bum Song | 0 | +22 | −19 | Victor | |
| Solomon Small | A Brivele von Russland | +1 | +15 | | Columbia | Acoustic |
| Chaim Tauber | Motl the Operator | +1 | +11 | −29 | Asch | Very bad surface |
| Aaron Lebedeff | Roumania, Roumania | +1 | +10 | | Banner | Post–WWII |
| Sara Martin | Don't You Quit Me Daddy | +2 | +15 | | OKeh | Acoustic, c. 1924 |
| | (ditto, worst banger) | +2 | +23.5 | | | |
| Miriam Kressyn | Macheteinista | +2 | +13 | | Banner | Post–WWII |
| Unknown artists | Ch Sidem | +3 | +19 | −21 | Columbia | Acoustic, c. 1900 |
| Ben Light & his Surf Club Boys | Girl from Atlantic City | +5 | +20 | −21 | Good Humor | Semi–illicit, c. 1940 |
| Pope's Arkansas Mountaineers | Birmingham | +6 | +25 | | Victor | c. 1928; very worn |
| | (ditto, worst banger) | +6 | +26 | −18 | | |
| Carson Robison | Dive for the Oyster | +6 | +18.5 | −21 | Columbia | Red label |
| Golden Gate Quartet | Do Unto Others | +6 | +20 | | Columbia | 1948 |
| Nellie Lutcher | He Sends Me | +6 | +20.5 | | Capitol | Worn; 1948 |
| Swing & Sway w. Sammy Kaye | I Thought She Was a Local | +6.5 | +24.5 | −19 | RCA Victor | 1950; Cracked |
| Parker Quartette | Swing Wide | +7 | +22 | | Decca | 1937 |
| Bagelman Sisters | Nor Dee | +8 | +14.5 | | RCA Victor | Post–WWII? |
| A few LPs: | | | | | | |
| Mickey Katz & his Orch | [no title] | 0 | +11 | | Capitol | |
| Paul Robeson | Swing Low, Sweet Chariot | 0 | +18 | −25 | Columbia | c. 1950; wrecked |
| Various artists | Sorrow Come Pass Me Around | +2 | +8 | | Advent | |
| The Who | Happy Jack | +5 | +16 | | Decca | c. 1966; v. hot |
| Worst signal (Shure test, Kogen–Jakobs–Karlov article) | | +23 | | | | 70cm/sec |

This de facto "semi-pro" standard has some advantages over the pro level; all other things being equal, a given amplifier will usually produce less distortion at a lower voltage.

Most sound cards and external A/D converters will operate quite nicely with a nominal input level of –10dBV; most home tape decks (analog or DAT) will, too. This corresponds to –7.8dBu. (Since my AC voltmeter measures in dBu rather than dBV, I'll use that scale to describe levels in this article.)

Nominal level on phonograph discs is defined by the velocity of the stylus; for many years the standard level has been 5cm/s. Most modern cartridges are velocity-sensitive devices, so a given velocity on a disc should produce the same voltage output from the cartridge, regardless of frequency. (Cartridges aren't perfect, of course, but good ones are reasonably flat within the audio band.)

How closely do real records adhere to this standard? I looked at a few representative LPs from my collection; virtually all of them were recorded at or above a nominal level of 5cm/s. The hottest disc I tried, the Who's "Happy Jack," was cut at an average level of +5dB, or 8.9cm/s.

Typically, 78s were a few dB hotter than LPs, and my hottest (the Bagelman Sisters' "Nor Dee") came in at +8dB. (See the "Average Level" column of *Table 1* for details.)

Enough recordings, especially LPs, cluster near the 5cm/s mark that I think it reasonable to design phono preamps around this figure.

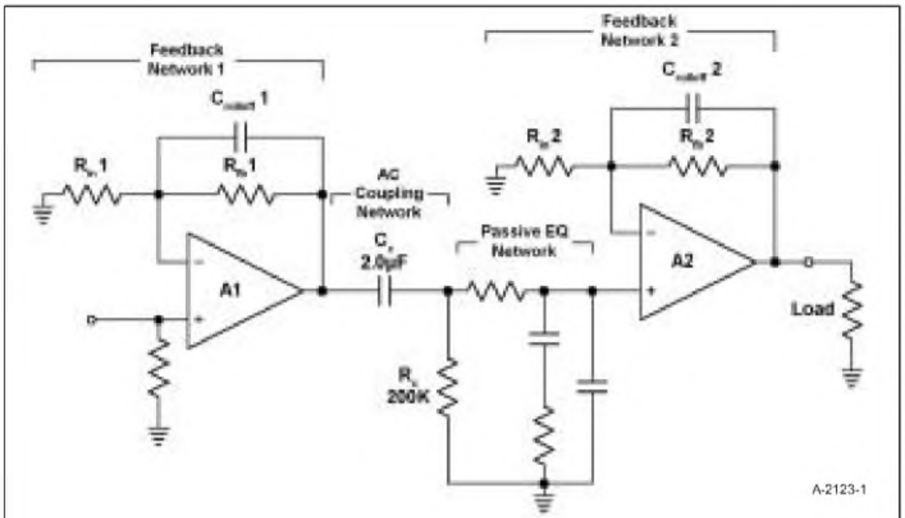


FIGURE 1: Passive preamp circuit—general form. Not all designs will use the two feedback networks, and not all will use the AC coupling network.

## A CARTRIDGE IN A PEAR TREE

What about cartridge outputs? Here I've deliberately restricted my universe to commonly-available and reasonably affordable moving-magnet cartridges. Low-output moving-coils are quite popular in audiophile circles, and can certainly provide excellent performance.

The cost, however, is often astronomical—and no one, to my knowledge, makes a moving-coil cartridge with a 78 stylus. (Those wishing to use these articles for designing moving-coil preamps for RIAA curves can certainly do so, of course; just incorporate a flat pre-preamp stage or transformer before the first gain block.)

Most cartridges that can be adapted for use with 78s have a nominal output of 5mV at 5cm/s; these include Grados, most Shures, and various Stantons. The Shure V15VxMR is lower; its output at nominal level is 3mV. Since 5mV = −43.8dBu and 3mV = −48.2dBu, to put out a nominal −7.8dBu, a phono preamp for a 5mV cartridge needs a midband (1kHz) gain of 36dB; one for a 3mV cartridge needs a midband gain of 40.4dB.

The passive equalization network introduces a fixed loss of 20dB at midband; therefore, total gain from both blocks must be 56dB and 60.4dB for the two respective cartridge outputs. Dividing the gain equally among both blocks (as a first approximation), this means each block has a gain of 28dB (5mV cartridge) or 30.2dB (3mV cartridge). These correspond to absolute gains of 25× and 32.4×, respectively.

## THE BIG BANGER THEORY

So far, we've discussed nominal output levels. What are the worst peaks you can expect to encounter? After all, these will determine the overload performance of the gain blocks and preamp as a whole.

Kogen, Jakobs, and Karlov, three engineers who helped develop the Shure V15 series, measured signal levels on a variety of LPs; the hottest they found was 70cm/s.[2] This is an impressive number; 23dB above nominal level, it requires considerable headroom in the reproducing system.

There are worse levels, though, if you consider scratches as well as music—especially on 78s. For my next series of tests, I measured the output of my cartridge directly, using the happy coincidence that my DAT recorder has an input impedance of 47k, perfect for cartridge loading.

Here's how I did it. First, I set up the DAT recorder using a 440Hz 0dBu tone, adjusting the input level so that the meters and headroom indicator read 0dB. This meant, in effect, that I was reading the peak of the 0dBu tone as 0dB full-scale, so that any signals I measured from the cartridge would also be the peak-equivalent. I then connected the turntable to the DAT recorder's input, using the V15 on a few LPs and a Shure M91ED, with N91-3 stylus, on 78s.

The results are listed in the column of *Table 1* labeled "Raw Peak"; this is an un-equalized signal, straight off the cartridge, measured in dBu. The worst peaks came from 78s, which makes sense (the stylus gets a heftier kick to the side when it strikes a scratch or hunk of crud at a higher speed). The worst case, a gouge that caused the cartridge to jump grooves, came on Pope's Arkansas Mountaineers' "Birmingham," a record that sounds seriously ill-treated. That scratch produced a cartridge output of −18dBu, or 25.8dB over nominal level.

To check my results, I also tested the amplified and equalized signals from the same records (and a few others). The results were broadly similar—see the column labeled "Amplified Peak." Here the results are referenced to nominal recording level (5cm/s). Note that the figure for the worst banger, "Birmingham," is +26dB, which agrees reasonably well with the raw peak. This indicates that my measurements are probably in the right ballpark.

These were my worst scratches; LPs didn't hold a candle. Note, while I'm talking about the table, that a couple of records, "The Bum Song" and "Cohen on the Telephone," require substantial headroom to reproduce the scratches; if you're listening to the music or talk at 83dBA (a common monitoring level in studios, and the one I use most of the time), you'll need a speaker and amplifying chain that can reproduce 105–108dBA cleanly.

I'm calling that the worst case: −18dBu at the preamp's input. While I didn't find any LPs with similar scratch levels, I think it's prudent to plan for them.

## THE FIRST STAGE

What sort of levels does the first stage need to handle? With a 5mV cartridge, and a stage gain of 28dB, worst-case output would be +10dBu equivalent, or

3.46V pk. (A 3mV cartridge has 4.4dB lower output for the equivalent scratch, while the stage gain is 2.2dB higher, so the first stage's output is +7.8dBu. I did my calculations for the 5mV cartridge, which stresses the stage more.)

A value of 3.46V isn't particularly high; a solid-state circuit (IC or discrete) operating on ±15V rails can easily produce that level cleanly. So can most tubed stages.

## CURRENT AFFAIRS

What about current draw? To calculate that, you need to know the total load presented to the amplifier.

Look at *Fig. 1* again. The passive equalization network presents an absolute minimum load to A1 that is equal to the network's series resistor (44.2k for the adjustable-curve version). If A1 incorporates a feedback network ($R_{in}1$–$R_{fb}1$), it will appear in parallel with the equalization network. If the stage is RC-coupled, the coupling resistor ($R_C$) will also appear in parallel.

How big is the feedback network, if used? Grossly oversimplifying, a typical moving-magnet cartridge presents a DC source resistance to the preamp of about 1k. This generates thermal noise (unless you cool it to absolute zero, which is not practical). Any other resistance on the input stage, such as the feedback net-

**TABLE 2**
**RIAA CURVE, V15VMR (3mV SENSITIVITY)**

| AMPLIFIERS | A1 $E_N$ (nV/RTHZ) | A1 $I_N$ (pA/RTHZ) | A1 $R_N$ (OHMS) | A2 $E_N$ (nV/RTHZ) | A2 $I_N$ (pA/RTHZ) | A2 $R_N$ (OHMS) | NOISE A1 (µV) | NOISE A2 (µV) | NOISE A1+A2 (µV) | S/N (dB) | AMPLIFIERS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Noiseless | 0 | 0 | 0 | 0 | 0 | 0 | 73.7 | 26.0 | 78.1 | –72.1 | Noiseless |
| Pass | 3.2 | 0 | 253 | 2.26 | 0 | 253 | 82.8 | 29.5 | 87.9 | –71.1 | Pass |
| 5534/5534 | 3.5 | 0.5 | 253 | 3.5 | 0.5 | 253 | 91.8 | 32.8 | 97.5 | –70.2 | 5534/5534 |
| 5534/5534 | 3.5 | 0.5 | 253 | 3.5 | 0.5 | 1000 | 91.8 | 36.6 | 98.9 | –70.1 | 5534/5534 |
| PassLS | 5.9 | 0 | 253 | 4.2 | 0 | 253 | 95.7 | 33.7 | 101.5 | –69.9 | PassLS |
| 6SL7_487 | 0 | 0 | 2100 | 0 | 0 | 2000 | 96.5 | 37.1 | 103.4 | –69.7 | 6SL7_487 |
| 5534/604 | 3.5 | 0.5 | 253 | 10 | 0 | 253 | 91.8 | 53.6 | 106.4 | –69.5 | 5534/604 |
| 5534/604 | 3.5 | 0.5 | 253 | 10 | 0 | 1000 | 91.8 | 56.0 | 107.6 | –69.4 | 5534/604 |
| 12AX7_1k | 0 | 0 | 2886 | 0 | 0 | 2786 | 103.8 | 40.6 | 111.4 | –69.1 | 12AX7_1k |
| 1028/1028 | 1 | 1 | 752 | 1 | 1 | 753 | 113.4 | 34.3 | 118.4 | –68.5 | 1028/1028 |
| 1028/1028 | 1 | 1 | 752 | 1 | 1 | 1499 | 113.4 | 38.3 | 119.7 | –68.4 | 1028/1028 |
| 12AX7_2.2k | 0 | 0 | 4369 | 0 | 0 | 4269 | 116.3 | 46.6 | 125.2 | –68.0 | 12AX7_1k |
| 1028/604 | 1 | 1 | 752 | 10 | 0 | 253 | 113.4 | 53.6 | 125.4 | –68.0 | 1028/604 |
| 1028/604 | 1 | 1 | 752 | 10 | 0 | 1000 | 113.4 | 56.0 | 126.5 | –68.0 | 1028/604 |
| 604/1028 | 10 | 0 | 253 | 1 | 1 | 753 | 123.6 | 34.3 | 128.3 | –67.8 | 604/1028 |
| 604/604 | 10 | 0 | 253 | 10 | 0 | 253 | 123.6 | 53.6 | 134.7 | –67.4 | 604/604 |

work, will generate more noise.

To keep this added noise below 1dB, the resistance of the feedback network should be less than 25.9% of the cartridge's resistance. This works out to about 250Ω. Since the stage gain needs to be 25×, $R_{fb}1 = 24 \times R_{in}1$. If you want the combined resistance appearing at the preamp input to be 250Ω, then $R_{in}1 = 255\Omega$ and $R_{fb}1 = 6.19k$ (rounding off to the nearest E96 values).

In a differential amplifier, the same signal (well, nearly so) appears across $R_{in}$ as across the amplifier input. The "Birmingham" scratch therefore drops −18dBu, or 137mV pk, across 255Ω, requiring 540µA. The network and coupling resistor present a worst-case load of 36.2k, requiring (for 3.46V output) another 96µA. Under absolute worst-case conditions, then, the first gain block must produce 636µA of current.

This doesn't sound like much, but the output devices of many IC op amps are biased at about 200µA. If you use an IC, and want it to operate in Class A at all times, you'll need to bias the outputs to a higher quiescent current by using a resistor or constant-current source to V−. A 1.6mA bias ought to do fine.

What about slewing? Walter Jung's rule of thumb suggests that an amplifier slew rate of 1V/µs for each peak volt of output is sufficient, so a slew rate of 3.5V/µs or more should work fine. Many IC op amps (including a unity-gain-compensated 5534) can easily achieve that. So can discrete solid-state circuits. Vacuum tube circuits are seldom slew-limited, so the issue doesn't arise with them.

How fast will the signal actually be moving? Assume for the moment that you're using a Grado cartridge; these are unusually wideband, with a specified high-frequency rolloff point of 50kHz. If the rolloff constitutes a fourth-order Butterworth low-pass filter (a fair approximation), the rise time will be 9.3µs. The amplifier's maximum rate of voltage change ("signal slope") will be 0.37V/µs, which ought to keep most decent op amps from ever slewing.

Of course, there's no law that says the first amplifier must use feedback; it's entirely possible to build an open-loop circuit that does the job. Such a circuit only sees a coupling resistor and the network, easing current requirements considerably. Circuits without global feedback don't usually slew—one less thing to worry about.

Given these numbers, I think it's safe to say that the danger of input stage overload in a passive preamp of this type may be exaggerated.

## THE SECOND STAGE

With either a 3mV or a 5mV cartridge, the output of stage 2 at nominal level will be −7.8dBu, by stipulation. What will it do on bad scratches?

My worst-case scratch measured 25.8dB higher than nominal level when unamplified, and 26dB higher with amplification and equalization. Taking the worse figure, the maximum output level will be +18dBu, or 8.77V pk.

That's a pretty hefty voltage—but still within the range of an IC op amp, running on ±15V rails. Some tube circuits will also be fine at this level, given a friendly load impedance; others won't.

How much current now? Assume a feedback amplifier with the same resistor values as those in the first stage; the feedback network draws 1.38mA. A 50k load will draw an additional 165µA, a 10k load 877µA, and all three together 2.42mA. (This might represent a 50k level control plus a 10k tape deck hanging on the amplifier's output, all unbuffered. That's an extreme case, of course; normally I'd include a buffer for the tape deck.) The amplifier might have its output biased at 2.6mA (50k load) or 3.4mA (10k + 50k).

By Jung's criteria, a feedback amplifier should have a slew rate of at least 8.77V/µs. The signal's maximum rate of change, however, is more limited than it was in the first stage, because it has passed through the equalization network. In the worst case, the signal is bandlimited to 17kHz with a 6dB/octave rolloff; this corresponds to a rise time of 9.36µs (by coincidence, almost exactly the same as the first stage). The actual maximum rate of change, therefore, will be 0.94V/µs. The 6V/µs slew rate of a compensated 5534 (or anything faster) should provide sufficient margin to make the preamp slew-proof.

## MAKE A JOYFUL QUIET

Calculating noise performance in phono preamps is tricky. A typical moving-magnet cartridge usually has an impedance which, in the audio range, rises steeply due to the cartridge's inductance, hits a resonant peak as the inductance meets load capacitance, then drops as capacitance takes over (*Fig. 3*). Applying the RIAA (or another) equalization curve complicates the issue.

After a few false starts, I devised a brute-force method of calculation that I think yields reasonably accurate results. (I'd like to thank Bob Stanton for prodding me into using a realistic model of cartridge impedance instead of an oversimplified one, and also for providing the rise time figure for a 50kHz low-pass filter I cited earlier. He is not, of course, responsible for my conclusions, nor for any errors on my part!)

First, I modeled various cartridges (including recommended load resistance and capacitance) using PC-ECAP, an AC circuit simulation program. Next, I divided the audio spectrum into several regions, representing either portions of the RIAA curve or areas where the cartridge/load impedance showed a particular behavior. For each region, I computed the average impedance, calculated the noise the cartridge would generate when used with various amplifier circuits, then applied the appropriate amount of gain for that region of the RIAA curve. (If I keep this up, I might invent calculus.) I did the

**TABLE 3**
**RIAA CURVE, GRADO (5mV SENSITIVITY)**

| AMPLIFIERS | A1 $E_N$ (nV/ RTHZ) | A1 $I_N$ (pA/ RTHZ) | A1 $R_N$ (OHMS) | A2 $E_N$ (nV/ RTHZ) | A2 $I_N$ (pA/ RTHZ) | A2 $R_N$ (OHMS) | NOISE A1 (µV) | NOISE A2 (µV) | NOISE A1+A2 (µV) | S/N (dB) | AMPLIFIERS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Noiseless | 0 | 0 | 0 | 0 | 0 | 0 | 26.7 | 20.0 | 33.4 | –79.5 | Noiseless |
| Pass | 3.2 | 0 | 245 | 2.26 | 0 | 245 | 35.7 | 22.7 | 42.3 | –77.5 | Pass |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 245 | 37.2 | 25.2 | 44.9 | –76.9 | 5534/5534 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 744 | 38.4 | 26.3 | 46.6 | –76.6 | 1028/1028 |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 1000 | 37.2 | 28.2 | 46.6 | –76.6 | 5534/5534 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 1499 | 38.4 | 29.5 | 48.4 | –76.3 | 1028/1028 |
| PassLS | 5.9 | 0 | 245 | 4.2 | 0 | 245 | 46.9 | 25.9 | 53.6 | –75.4 | PassLS |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 245 | 37.2 | 41.2 | 55.5 | –75.1 | 5534/604 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 245 | 38.4 | 41.2 | 56.3 | –75.0 | 1028/604 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 1000 | 37.2 | 43.1 | 56.9 | –74.9 | 5534/604 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 1000 | 38.4 | 43.1 | 57.7 | –74.8 | 1028/604 |
| 6SL7_1K | 0 | 0 | 2812 | 0 | 0 | 2712 | 52.6 | 31.0 | 61.0 | –74.3 | 6SL7/1k |
| 12AX7_1k | 0 | 0 | 2866 | 0 | 0 | 2866 | 53.0 | 31.5 | 61.6 | –74.2 | 12AX7_1k |
| 604/1028 | 10 | 0 | 245 | 1 | 1 | 744 | 68.2 | 26.3 | 73.1 | –72.7 | 604/1028 |
| 604/604 | 10 | 0 | 245 | 10 | 0 | 245 | 68.2 | 41.2 | 79.7 | –72.0 | 604/604 |

same for the passive network.

I looked at three popular cartridges. I chose the Shure V15VxMR, popular in various versions for a couple of decades now, as an example of a cartridge with moderately high inductance and lower output (3mV at nominal 5cm/s). The Grado cartridge (no model specified, as most of their current units have identical electrical characteristics) represents low inductance and resistance, plus higher output (5mV), while the Stanton 500 is a high-inductance, higher output cartridge (again, 5mV). It's not often used for LPs except by club DJs and radio stations, but it's popular for playing 78s.

My results for the RIAA playback curve are summarized in *Tables 2–4*. First, note the significantly better signal-to-noise ratios theoretically attainable with a Grado cartridge. This improvement is mostly due to the Grado's lower inductance, leading to lower self-noise generation (Johnson noise) as well as lower current-generated noise when the cartridge is used with bipolar-input amplifiers. The higher output helps, too, but it's a secondary factor; the Stanton cartridge, with the same nominal output level, is still noisier than the Grado, due to the Stanton's high inductance.

**TABLE 4**
**RIAA CURVE, STANTON 500 (5mV SENSITIVITY)**

| AMPLIFIERS | A1 $E_N$ (nV/RTHZ) | A1 $I_N$ (pA/RTHZ) | A1 $R_N$ (OHMS) | A2 $E_N$ (nV/RTHZ) | A2 $I_N$ (pA/RTHZ) | A2 $R_N$ (OHMS) | NOISE A1 (µV) | NOISE A2 (µV) | NOISE A1+A2 (µV) | S/N (dB) | AMPLIFIERS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Noiseless | 0 | 0 | 0 | 0 | 0 | 0 | 61.4 | 20.0 | 64.6 | –73.8 | Noiseless |
| Pass | 3.2 | 0 | 245 | 2.26 | 0 | 245 | 66.4 | 22.7 | 70.1 | –73.1 | Pass |
| PassLS | 5.9 | 0 | 245 | 4.2 | 0 | 245 | 74.1 | 25.9 | 78.5 | –72.1 | PassLS |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 245 | 76.0 | 25.2 | 80.1 | –71.9 | 5534/5534 |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 1000 | 76.0 | 28.2 | 81.1 | –71.8 | 5534/5534 |
| 6SL7_1K | 0 | 0 | 2812 | 0 | 0 | 2000 | 76.3 | 28.5 | 81.5 | –71.8 | 6SL7_1K |
| 12AX7_1k | 0 | 0 | 2886 | 0 | 0 | 2786 | 76.7 | 31.2 | 82.8 | –71.6 | 12AX7_1k |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 245 | 76.0 | 41.2 | 86.5 | –71.3 | 5534/604 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 1000 | 76.0 | 43.1 | 87.4 | –71.2 | 5534/604 |
| 12AX7_2.2k | 0 | 0 | 4369 | 0 | 0 | 4269 | 83.4 | 35.8 | 90.8 | –70.8 | 12AX7_2.2k |
| 604/1028 | 10 | 0 | 245 | 1 | 1 | 744 | 91.4 | 26.3 | 95.1 | –70.4 | 604/1028 |
| 604/604 | 10 | 0 | 245 | 10 | 0 | 245 | 91.4 | 41.2 | 100.3 | –70.0 | 604/604 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 744 | 98.8 | 26.3 | 102.2 | –69.8 | 1028/1028 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 1499 | 98.8 | 29.5 | 103.1 | –69.7 | 1028/1028 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 245 | 98.8 | 41.2 | 107.1 | –69.4 | 1028/604 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 1000 | 98.8 | 43.1 | 107.8 | –69.3 | 1028/604 |

**TABLE 5**
**WORST-CASE 78 CURVE, GRADO (5mV SENSITIVITY)**

| AMPLIFIERS | A1 $E_N$ (nV/RTHZ) | A1 $I_N$ (pA/RTHZ) | A1 $R_N$ (OHMS) | A2 $E_N$ (nV/RTHZ) | A2 $I_N$ (pA/RTHZ) | A2 $R_N$ (OHMS) | NOISE A1 (µV) | NOISE A2 (µV) | NOISE A1+A2 (µV) | S/N (dB) | AMPLIFIERS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Noiseless | 0 | 0 | 0 | 0 | 0 | 0 | 64.8 | 33.5 | 73.0 | –72.7 | Noiseless |
| Pass | 3.2 | 0 | 245 | 2.26 | 0 | 245 | 75.4 | 35.2 | 83.2 | –71.6 | Pass |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 245 | 78.2 | 37.9 | 86.9 | –71.2 | 5534/5534 |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 1000 | 78.2 | 39.9 | 87.8 | –71.1 | 5534/5534 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 245 | 78.2 | 49.2 | 92.4 | –70.7 | 5534/604 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 744 | 82.8 | 41.5 | 92.6 | –70.7 | 1028/1028 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 1000 | 78.2 | 50.8 | 93.2 | –70.6 | 5534/604 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 1499 | 82.8 | 43.6 | 93.6 | –70.6 | 1028/1028 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 245 | 82.8 | 49.2 | 96.3 | –70.3 | 1028/604 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 1000 | 82.8 | 50.8 | 97.1 | –70.2 | 1028/604 |
| PassLS | 5.9 | 0 | 245 | 4.2 | 0 | 245 | 90.4 | 37.3 | 97.8 | –70.2 | PassLS |
| 6SL7_1K | 0 | 0 | 2812 | 0 | 0 | 2712 | 96.0 | 41.0 | 104.4 | –69.6 | 6SL7_1K |
| 12AX7_1k | 0 | 0 | 2886 | 0 | 0 | 2786 | 96.7 | 41.2 | 105.2 | –69.6 | 12AX7_1k |
| 12AX7_2.2k | 0 | 0 | 4369 | 0 | 0 | 4269 | 109.6 | 44.8 | 118.4 | –68.5 | 12AX7_2.2k |
| 604/1028 | 10 | 0 | 245 | 1 | 1 | 744 | 121.6 | 41.5 | 128.5 | –67.8 | 604/1028 |
| 604/604 | 10 | 0 | 245 | 10 | 0 | 245 | 121.6 | 49.2 | 131.2 | –67.6 | 604/604 |

I calculated noise figures for a wide range of amplifiers (I almost said "a whole slew," but thought better of it.) Here's a decoding chart:

Pass: This is the JFET version of Nelson Pass's discrete op-amp circuit.[3]
PassLS: The same circuit, but with a Linear Systems LS-843 dual JFET substituted at the input.
5534: Signetics NE-5534a (also sourced by other manufacturers; the NE 5532a should perform identically).
1028: Linear Technology or Maxim LT-1028A.
604: Burr-Brown OPA-604 (OPA-2604 should perform identically).
6SL7_487: A 6SL7 tube with a cathode resistor of 487Ω. This can be an open-loop circuit, perhaps direct-coupled to a cathode follower, or the first stage of a feedback pair. Open loop, it provides the proper gain for a 3mV cartridge.
6SL7_1k: The same, with a 1k cathode resistor. This gives lower gain when used open-loop, appropriate for a 5mV cartridge, and an easier load when part of a feedback pair.
12AX7_1k: A 12AX7 tube, biased as in the Dynaco PAS preamp's phono stage ($R_{plate}$ = 150k, $R_{cathode}$ = 1k). Again, this could be open-loop or the first stage of a feedback pair.
12AX7_2.2k: The same, biased as in Audio Research's SP3/SP6 series of preamps ($R_{plate}$ = 301k, $R_{cathode}$ = 2.2k).
Noiseless: A hypothetical perfect amplifier, adding no noise of its own; all of the noise is thermally generated by the cartridge and the passive equalizing network.

The "$R_{noise}$" columns denote, for solid-state amplifier circuits, actual resistances (other than the source resistance) contributing to noise. These include feedback networks and, in the case of the LT-1028A, a resistor designed to increase stability (more later). I've set up the feedback networks using the nearest available E96 values.

With tubes, "$R_{noise}$" means something else. This number includes cathode resistors, and the first tube's plate resistor (divided by the square of the gain), but it also includes the "equivalent noise resistance" of the tube itself. Triode noise may be modeled as a resistance with the value 2.5/gm, where gm is expressed in siemenses (formerly mho).

A few clarifications are in order. First, note that some op amps are listed twice, with different values in the "A2 $R_{noise}$" column. I did this to see whether raising

**TABLE 6**
**WORST-CASE 78 CURVE, STANTON 500 (5mV SENSITIVITY)**

| AMPLIFIERS | A1 $E_N$ (nV/ RTHZ) | A1 $I_N$ (pA/ RTHZ) | A1 $R_N$ (OHMS) | A2 $E_N$ (nV/ RTHZ) | A2 $I_N$ (pA/ RTHZ) | A2 $R_N$ (OHMS) | NOISE A1 (µV) | NOISE A2 (µV) | NOISE A1+A2 (µV) | S/N (dB) | AMPLIFIERS |
|---|---|---|---|---|---|---|---|---|---|---|---|
| Noiseless | 0 | 0 | 0 | 0 | 0 | 0 | 205.4 | 20.0 | 206.3 | –63.7 | Noiseless |
| Pass | 3.2 | 0 | 245 | 2.26 | 0 | 245 | 208.9 | 22.7 | 210.2 | –63.5 | Pass |
| PassLS | 5.9 | 0 | 245 | 4.2 | 0 | 245 | 214.8 | 25.9 | 216.4 | –63.3 | PassLS |
| 6SL7_1K | 0 | 0 | 2812 | 0 | 0 | 2000 | 217.2 | 28.5 | 219.1 | –63.2 | 6SL7_1K |
| 12AX7_1k | 0 | 0 | 2886 | 0 | 0 | 2786 | 217.6 | 31.2 | 219.8 | –63.2 | 12AX7_1k |
| 12AX7_2.2k | 0 | 0 | 4369 | 0 | 0 | 4269 | 223.6 | 35.8 | 226.4 | –62.9 | 12AX7_2.2k |
| 604/1028 | 10 | 0 | 245 | 1 | 1 | 744 | 229.7 | 26.3 | 231.2 | –62.7 | 604/1028 |
| 604/604 | 10 | 0 | 245 | 10 | 0 | 245 | 229.7 | 41.2 | 233.3 | –62.6 | 604/604 |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 245 | 263.9 | 25.2 | 265.1 | –61.5 | 5534/5534 |
| 5534/5534 | 3.5 | 0.5 | 245 | 3.5 | 0.5 | 1000 | 263.9 | 28.2 | 265.4 | –61.5 | 5534/5534 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 245 | 263.9 | 41.2 | 267.1 | –61.5 | 5534/604 |
| 5534/604 | 3.5 | 0.5 | 245 | 10 | 0 | 1000 | 263.9 | 43.1 | 267.4 | –61.4 | 5534/604 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 744 | 386.6 | 26.3 | 387.5 | –58.2 | 1028/1028 |
| 1028/1028 | 1 | 1 | 744 | 1 | 1 | 1499 | 386.6 | 29.5 | 387.7 | –58.2 | 1028/1028 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 245 | 386.6 | 41.2 | 388.8 | –58.2 | 1028/604 |
| 1028/604 | 1 | 1 | 744 | 10 | 0 | 1000 | 386.6 | 43.1 | 389.0 | –58.2 | 1028/604 |

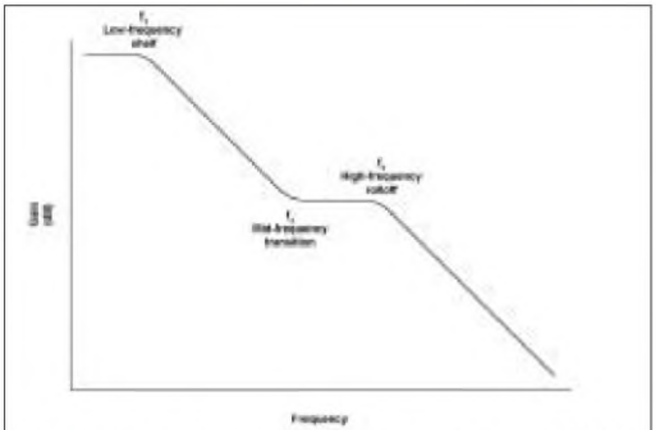

FIGURE 2: Generalized phono equalization curve. A-2123-2

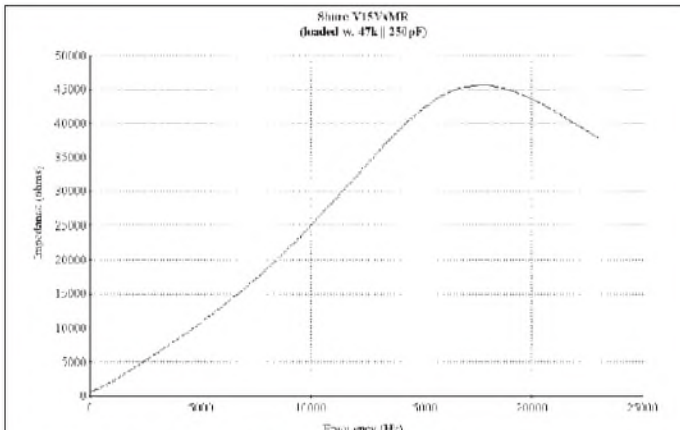

FIGURE 3: Impedance of a moving-magnet cartridge. A-2123-3

the resistance of the second-stage's feedback network would have serious effects on noise. It didn't; at worst (Grado cartridge with twin 5534s or 1028s) the signal-to-noise ratio worsened by only 0.3dB. Therefore, raising the resistance of the second stage's feedback network may be a viable option if you want to minimize current draw.

Second, note that for both of the higher-inductance cartridges, noise from the cartridge and A1 predominates over noise from the network/A2 combination. With these cartridges, changing the second amplifier from a quiet IC (5534 or 1028) to a noisier one (604) degrades S/N by only 0.7dB at worst, again a negligible amount. Because the Grado inherently produces less input noise, it's more sensitive to changes in A2, but even here the worst-case degradation is 1.8dB, audible but not terrible.



Although it's not shown in the tables (I had to stop someplace), you can similarly mix and match tubed circuits. For example, using the 12AX7 with a 1k cathode resistor for stage 1, but a 2.2k resistor for stage 2, decreases the signal-to-noise ratio by only 0.3dB from using 1k in stage 2, again a negligible amount.

That's a good thing for preamps designed for switchable equalization; using a FET-input amplifier for A2 (or a tube) minimizes input bias currents, which could cause popping as network components are switched in and out. The fact that cartridge and input-stage noise predominate also implies that the theoretical degradation in noise performance inherent in passively-equalized preamps is, for well-designed circuits, of little practical importance.

Note, in passing, that the worst S/N listed for the Grado (604 op amps in both positions) is within 0.1dB of the figure for the Shure V15 with a perfect, noiseless amplifier—and both are still quieter than any record surface in my experience.

Surprisingly, the LT-1028A comes out noisier than the venerable NE-5534a. You wouldn't expect that, based on the chips' specs, but there's a kicker. The 1028 is not guaranteed to be stable at unity gain. In this circuit, the band-limiting capacitor in the feedback network forces the circuit to unity gain at high frequencies; to maximize stability, the manufacturer recommends inserting a 500Ω resistor in series with the non-inverting input. I included that resistor in the $R_{noise}$ columns for the 1028, and its noise pushes the 1028 below the 5534 in noise performance.

It's worth noting, too, that even the poorer combinations are remarkably good; the quieter designs, used with

Grado cartridges, approach the noise performance of a 16-bit digital system (S/N = 80dB, assuming a nominal level of −16dB ref. full-scale).

## STRIKE UP THE BANDWIDTH

I applied the same analysis to a "worst-case" 78 equalization. This consists of a bass boost with breakpoints of 50Hz and 500Hz (standard RIAA frequencies), but a treble curve that doesn't begin to roll down until 17kHz, essentially a flat response. This curve might be used in playing European electrical 78s; without the bass boost, it would be appropriate for acoustic recordings. (In practice, I suspect you'd seldom use an EQ this extreme—but that's why it's a worst case.)

The results, for the Grado and Stanton cartridges (nobody in my experience uses a V15 for playing 78s), are summarized in *Tables 5–6*. Because the treble is rolled off at a much higher frequency than it is in the RIAA curve, amplification of high frequencies is much greater, and signal-to-noise ratios are correspondingly worse. For this configuration, input noise (cartridge- and amplifier-generated) predominates, while the contribution of A2 is minimal.

As a result, the high inductance of the Stanton exacts a severe penalty; with all listed amplifiers, S/N is significantly poorer. The Grado, on the other hand, performs well; in fact, in this worst-case 78 setup, it performs about the same with any given amplifier as the Shure, RIAA-equalized! Which means, in practice, that it should work fine.

So should the Stanton, provided the input amplifier uses tubes or FETs rather than bipolar transistors. Interestingly, the Stanton is at its noisiest when used with "low-noise" bipolar-input ICs. While these chips have low voltage noise density ($e_n$), their current noise ($i_n$) is high enough to generate considerable noise when fronted by the high impedances a high-inductance cartridge shows in the top octaves.

## TUBES, TRANSISTORS, AND THE REAL WORLD

Tubed circuits have a reputation for poor noise performance in phono preamps, but, as these tables show, they can perform quite respectably. When used with RIAA equalization, the 6SL7 or 12AX7 yields S/N ratios in the ranges of 69dB (V15) or 74dB (Grado). Even on worst-case 78s, with the bandwidth wide open, S/N with a Grado cartridge is 69.6dB. These figures are better than any LP in my experience, and way better than any 78.

How relevant are these figures? If you listen at 83dBA, the worst RIAA preamp, with a Shure V15, would generate an acoustic noise level of 15.6dBA, barely audible in a quiet room. The worst-case 78 preamp, with a Grado cartridge, would generate 15.4dBA, about the same. Even with the Stanton 500, the worst-case 78 preamp would generate 24.8dBA, still darned quiet—certainly quieter than any 78 on earth.

(Lest anyone think I'm knocking the Stanton cartridge, or other high-inductance cartridges used to play 78s, rest assured I'm not. While I usually use Grados for transferring 78s, I keep a Shure M91ED or Stanton on hand as well; they are designed with lower compliance, at least when used with 78 styli, and will stay in the groove on warped records that give Grados the shim-me-sha-wabble.)

## CONCLUSION

Well-designed passive preamps, using good ICs, discrete transistors, or tubes, can be quiet enough for professional use or serious listening, with either LPs or 78s. Properly designed, with due attention to voltage levels and loading, they should also operate cleanly, and without overload, even on the worst signals or scratches.

This article, and my previous one, have developed enough information to enable readers to design the core of a passively-equalized phono preamp, with or without variable equalization curves. There are other bits that need to be considered in a real-world design, however, such as subsonic filtering, summing circuits, and buffers. I'll talk about them, with some worked examples, in a future article. Enjoy!

And now, back to the lab . . . ❖


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# Techniques and Gadgets for Audio Experimenters

Learn from the experiences (and miscues) of this audio veteran. **By Larry Lisle**

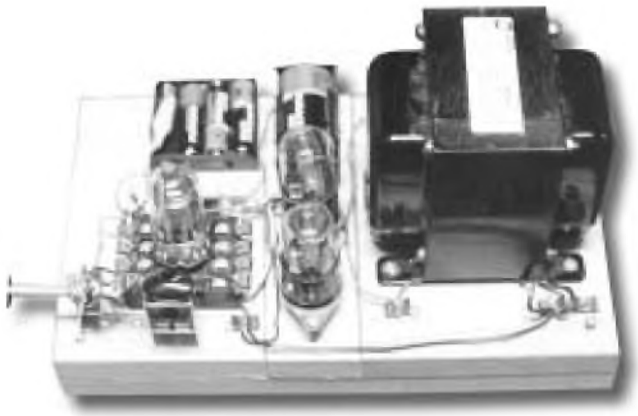


PHOTO 1: An example of experimental breadboarding. There are sub-boards for the input stage, the output tubes, and the transformer. Sometimes I use only a baseboard and one sub-board for the component or stage I'm experimenting with.

I started out doing things the hard way. I built my projects in aluminum boxes because that's the way they were built in the magazines I saw. I worked much harder on mechanical construction than on the electronics.

Since then, I've built many projects and eventually learned ways to overcome some of the difficulties in the construction and debugging of electronic circuitry. In this article I'd like to pass along a few ideas that may help make "do it yourself" audio more fun and less expensive.

## DON'T START WITH A METAL CHASSIS

This was my biggest blunder, and I'm ashamed to admit how long I kept starting projects on a metal chassis. Unless you're building a kit or copying a proven circuit from an article or book, starting with a brand new and expensive chassis is not a good idea. Wiring a chassis can be laborious, especially when you reach down inside a box and can't see what you're doing because your fingers and the soldering iron (ouch!) are in the way.

Changing the wiring later is difficult, even substituting one resistor value for another. It's also hard to keep the wiring looking neat and the top of the chassis free from scratches and extra holes. Too often you'll wind up with a project that doesn't look very good, or worse, one that you know could sound better if you had made a few changes.

## BREADBOARDS AND SUB-BREADBOARDS

I have come to prefer breadboarding and working in stages or modules. Look at *Photo 1*. In building this experimental amplifier that will produce less than 1% IM distortion without feedback, I used a baseboard and three separate "sub-breadboards." I can alter or completely change the input stage, output tubes, or the output transformer without disturbing the rest of the amplifier.

You can use the technique in many different ways. Sometimes I'll use just the baseboard and one sub-breadboard for a part or stage that I plan to change frequently. If shielding is required, I'll put a small metal enclosure around the particular part or module that needs it.

## BREADBOARDING WITH METAL CHASSIS

*Photo 2* shows another technique. I built a proven input stage on one chassis, a proven output stage on another, and a proven power supply on a third. In this case, I used an Electra-Print input transformer to drive a pair of triode-connected 3Q4s in parallel. The output stage consists of a parallel-fed Electra-Print 1:1 interstage transformer driving a single-ended type 45 tube and a Hammond output transformer.

A conventional power supply using a 6X4 is on the chassis on the right. In front of the amplifier is an alternate push-pull output stage using classic transformers by Thordarson and RCA and a pair of 45s. Each chassis has a single ground point, which are connected together both for safety and to avoid audio problems.

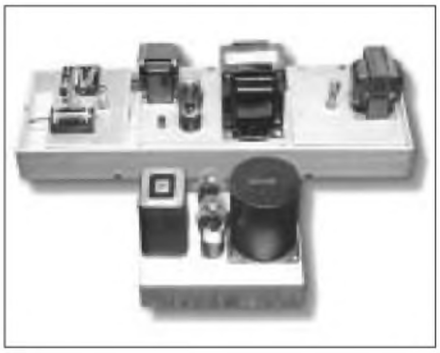


PHOTO 2: Another style of breadboarding. From left to right are chassis with a proven input module, a proven output module using a single-ended type 45 tube, and a proven power supply. In front of the amp is another output stage using push-pull 45s. I can experiment with any of the modules without disturbing the rest.

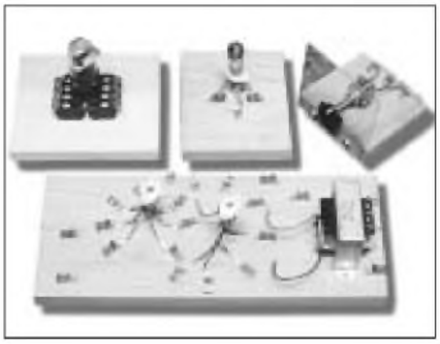


PHOTO 3: Some gadgets for making experimenting easy. Their use is explained in the text.

## BACK TO METAL

I could now transfer the amplifiers shown in *Photo 1* or *2* to a single metal chassis and be reasonably certain they would work with few changes. If you're a beginner, I suggest that a big chassis is a lot easier to wire than a small one. Lay out the project in a line from input to output with the grid terminals of each tube socket oriented toward the input while the plate terminals are toward the output. Use a single ground point.

Wiring carrying AC should be twisted together, shielded with braid from a piece of coaxial cable, and run close to the chassis. Orient transformers for minimum hum before making the installation permanent. Build the power supply on a separate chassis if possible.

A copy of *The ARRL Radio Amateur's Handbook* from the tube era has much useful information about chassis construction practices in general, and the section on building audio amplifiers is well worth reading. Current editions are a good source for solid-state construction ideas.

## OTHER TECHNIQUES

There are many other ways of using the breadboarding philosophy. *Photo 3* shows an octal relay socket with Fahnestock clips fastened under the screw terminals. Changing wiring or auditioning parts is a breeze with the clips. Another sub-breadboard holds a single seven-pin miniature tube socket supported on stiff pieces of solid hook-up wire. Of course, the setup is microphonic, but with care it can give good information.

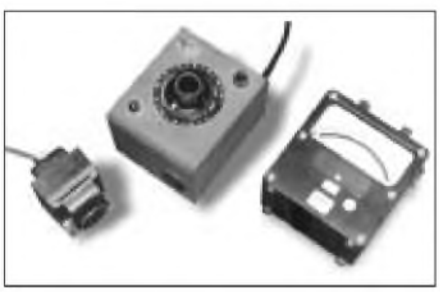


PHOTO 4: An isolation transformer, variable voltage transformer, and lab type meter are handy items for the experimenter.

PHOTO 5: Test equipment is very useful for the audio experimenter. This Heathkit intermodulation distortion meter usually sells for about $25 at flea markets.

Also shown in the photo is a similar sub-board with two sockets and a multi-impedance transformer. Finally, there's a potentiometer mounted on a piece of Plexiglas for safety. You can adjust the "pot" until the circuit is optimum and then substitute the nearest fixed value resistor. I've also used small boards for solid-state components and printed wiring sub-assemblies.

## MORE GADGETS

In *Photo 4* you can see some useful items. A variable voltage transformer can serve many functions. For instance, if you're bringing to life a piece of audio gear that hasn't been fired up for several years, use the variable transformer to raise the voltage slowly and save the filter capacitors.

It's also handy when testing a brand new project. Start it at a low voltage and watch for smoke or out-of-norm meter readings before slamming it with the full input. If you build a bench power supply with separate input for filament and plate circuits, you can vary the primary of the high-voltage supply and dial the volts you want for many different needs.

The isolation transformer is an important piece of safety equipment to have when you're testing antique audio gear or brand new projects. Make sure variable voltage or isolation transformers have a more than sufficient power rating for any expected load.

Finally, meters are very useful. One of the first pieces of non-audio equipment you should buy is a good multimeter for measuring voltage and current. As you proceed in the hobby, keep an eye out for one function meter like the one shown that you can leave connected to an experimental amplifier as you make changes.

Test gear designed for audio can save you a lot of time. For example, *Photo 5* shows a Heathkit intermodulation distortion meter that can also include an audio vacuum tube voltmeter (VTVM). These and other instruments may be old, but they will still tell you much about what is going on inside your amplifier. They might not be as accurate as newer instruments in the absolute sense, but you can see the effects of adjustments and compare different amps. The one shown was purchased for $25 at an electronic flea market.

In addition to the material mentioned here, documentation is also important. Start by labeling tubes with a piece of masking tape, for example, "RCA 6L6 #1." As you use tubes, their markings can often become smeared or disappear.

There are also slight variations between tubes of the same type. If you balance a push-pull stage, for instance, you want to make sure the tubes get back in the same socket. It's a good idea to try several tubes in each socket during the breadboard phase. If you don't, you might inadvertently be using a tube that's special in some way, such as having unusually high transconductance or low noise. Should the circuit someday stop working, replacing that tube with another might not be satisfactory, and you won't know why.

You should also make a clean schematic diagram of your finished project with directions for any adjustments that aren't obvious. Also include voltage, current, power, and distortion readings, along with the conditions under which they were taken and the type of instrument used.

A laboratory type notebook is also a good thing to start. It doesn't need to be formal, just diagrams and notes on your equipment and the changes you make as they develop. I often find myself looking back at notes on experiments I've made in the past.

Building amplifiers and other electronic gear has been a lot of fun for me over the years. I hope these suggestions will make it more enjoyable for you, too. ❖

# Product Review

## Beyond Thiele/Small

## DUMAX and Klippel Driver Measurement Systems

By Thomas Perazella

PHOTO 1: The measurement part of DUMAX. Note the ability to rotate the test chamber.

Anyone who is serious about designing a loudspeaker using dynamic drivers is either familiar with, or has used, Thiele/Small parameters to model those drivers. They are generally accepted as ways to predict driver behavior in different enclosures.

One factor that is often overlooked when working with T/S parameters is that they are small-signal parameters. When making small signal measurements of a driver, the displacement of the voice coil is kept to a minimum to provide a linear model. Having a linear model allows the measured parameters to be effective in the design process. All is then well with the world until the first time you use the resulting speaker to reproduce music at any level above a whisper.

### NONLINEARITIES

Then, the demons of driver nonlinearity rear their ugly heads. Do you have a T-Rex tromping through the house that sounds more like a drunken cat, or a space shuttle launch that sounds more like your neighbor's misfiring lawn mower? Most likely, the sound you're hearing is the result of driver nonlinearity manifesting itself as distortion. It's interesting to watch people agonize over minute differences in speaker small-signal parameters when choosing a driver for a new speaker and yet totally disregard the fact that using a driver out of its linear operating area will radically change the character of the reproduced sound in a very negative way.

Nonlinearities that typically affect dynamic drivers are:

1. Force factor (Bl) changes with distance from a rest position
2. Compliance changes with distance from a rest position
3. Asymmetrical voice-coil inductance, that is inductance that changes differently as the voice coil moves in different directions from the rest position
4. Variable voice-coil impedance with temperature
5. Flux modulation
6. Change of frequency response with position (typically due to surround resonance, resulting in IM distortion)
7. Doppler distortion
8. Suspension hysteresis and thixiotropy

There are other nonlinearities that can affect the sound of a finished speaker, including but not limited to:

1. Enclosure leaks or flexing (may not be nonlinear)
2. Mechanical or acoustical noises from the driver, port, and so on
3. Decentering of the diaphragm motion, or "oil-canning"
4. Gross suspension problems such as inversion of the surround
5. Drive amplifier current limitations due to extreme impedance excursions

The good news is that there are measurement methods you can use to determine and quantify the first four, and sometimes other nonlinearities just mentioned. Two methods that have reached the practical stage are the DUMAX® system (**D**rive **U**nit **M**easurements **A**t e**X**cursion) produced by DLC Design of Wixom, MI, and the Klippel Analyzer System by Klippel GmbH of Dresden, Germany. They use very different approaches to measure the nonlinearities. However, both have produced results that successfully model driver behavior under large-signal conditions. If you wish to get into the technical details on these methods, information on them is available in the form of AES papers listed later.

### DISTORTION

How do these nonlinearities produce distortions? Results depend to some degree on whether the driver is being operated above resonance where it is mass loaded or below resonance where it is compliance loaded.

For force factor, whether in the mass loaded or compliance loaded range, in order for diaphragm motion to be linear, the driving force must be linear across the desired range of movement. If the driving force produced by the

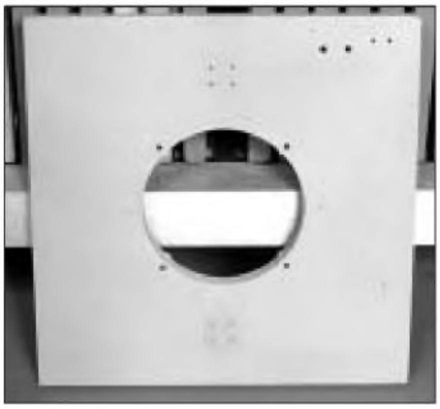


PHOTO 2: A wide range of baffles is necessary to accommodate various drivers.


PHOTO 3: A reflective mark placed in the center of the dust cap provides a reference spot for the front measuring laser.

voice coil/magnet structure for a given drive signal changes as the coil moves through its excursion range, the force on the coil, and therefore the diaphragm, will not be linear, resulting in distortion.

For compliance, you might think that because the driver is mass loaded above resonance, compliance nonlinearities would only matter below resonance. While they are certainly very important in this range, they can also affect the driver above resonance.

To visualize this, think of a driver above resonance with a magnetic structure having unlimited linear excursion capability. Also think of the suspension having a compliance that is totally linear throughout most of its range, but also having a "brick wall" change in compliance when it reaches a certain distance, much like having a mechanical stop on both sides of the voice coil. It is easy to visualize that, above resonance, until the "brick wall" compliance limit is reached, as drive level is increased, the mass being constant and dominant will result in linear output. However, once the artificial stops are reached, the output will certainly be distorted, much the same as when an amplifier goes into clipping.

In reality, the compliance changes are not that dramatic unless the voicecoil former strikes the back plate. However, as the excursion from the rest point increases, the compliance decreases. This additional resistance to movement adds to the inertia of the mass in a nonlinear way with excursion, resulting in distortion.

Although not related to the driver mass, to help visualize what is happening, you can think of suspension nonlinearity as adding a mass delta to the system with increasing excursion. Unlike mass, the compliance force is always in the direction of the rest position and is largely affected by distance rather than acceleration. Therefore, it is really not a true analogy. With Einstein, you gain mass as you approach the speed of light.

In this technically incorrect, but hopefully enlightening analogy, you can think of the driver as gaining mass as you approach the suspension limits. In reality, the compliance limits can produce a force opposing diaphragm movement that is greater at some point in the excursion range than the effect of the driver's mass. In practice, you can see the effect of this "added mass" as a decrease in $F_S$ with increasing drive level.

It should now be clear that any evaluation of a driver is not complete without information that indicates the limits of excursion that will result in a relatively linear output. Although there are no absolute guidelines, generally accepted limits for force factor and compliance changes are a reduction of Bl to 70.7% of the rest position value and a reduction of compliance to 25% of the rest position value.

Traditionally, $X_{MAX}$ is determined by a physical measurement of the voice coil and gap heights. This is a very simplistic approach that is at best a rough approximation of Bl limited excursion and does not take into account any compliance limits.

I've worked with drivers that were limited by Bl changes where there was plenty of linear compliance left and also the opposite case, where compliance limits strangled an otherwise very competent motor structure. An effective $X_{MAX}$ specification will take into account both factors separately, and for both directions of travel. In each direction, the smaller excursion where either limit is reached should be used. The two excursion values should be added, and $X_{MAX}$ specified as half of the total value, even if the excursion to one side is greater.

## DUMAX®

The DUMAX system[1] uses a special chamber to control offset from the rest position of the driver by applying a static pneumatic pressure. *Photo 1* shows the DUMAX system prior to mounting a driver.

Measurements of driver electromechanical parameters are then made at various offsets. These measurements can be mapped to T/S parameters if desired, but can be used in a nonlinear model, which the T/S parameters cannot. The minimum set of parameters includes:

1. $M_{MD}$ = moving mass of diaphragm (kg)
2. $C_{MS}$ = suspension mechanical compliance
3. $R_{MS}$ = mechanical resistance (N . s/m)
4. Bl = motor force factor
5. $R_E$ = resistance of voice coil (ohms)
6. $S_D$ = diaphragm area (m$^2$)

The measurement system used by DUMAX includes:

1. The test chamber
2. Laser position transducers for the diaphragm and magnet positions
3. A chamber pressure transducer
4. A test microphone inside the test chamber with a preamplifier
5. A known non-magnetic mass
6. A voltage-controlled pressure source
7. A current-source amplifier with remote sense
8. A Windows computer with data and acquisition cards and various measurement and analysis software

With DUMAX, primary emphasis is placed on changes in force factor and compliance, as these errors can occur repeatedly even when drivers are operated well within maximum power ratings due to the dynamic nature of music compared to test signals. Testing a driver is done in several stages.

## DRIVER MEASUREMENT

Driver preparation begins with selection of an appropriately sized baffle. *Photo 2* shows an example of one of the many baffles needed to test the great range of drivers that pass through DLC Design.

Two reference marks are put on the driver, one on the dustcap (*Photo 3*), and the other on the rear of the magnet structure (*Photo 4*). These marks are targets used by the laser positioning system. One laser is inside the chamber to measure the rear of the driver (*Photos 5* and *6*). The other laser is on the outside of the chamber to measure the position of the dustcap (*Photos 7* and *8*).

Precise measurement of the diaphragm position in relation to the frame is necessary in several of the tests for reasons I describe here. Changes such as bowing can occur in the shape of the test chamber housing as the static pressures are changed, affecting measurement accuracy. The driver is then mounted to the baffle and secured to the test chamber. Once sealed in the chamber, pneumatic force is used to extend the diaphragm to both extremes of excursion and then back and forth with 1mm less excursion for each cycle until back at the rest position. This "break-in" eliminates any offset in the normal rest position due to storage effects.

Next, force factor (Bl) is measured both at rest and as a function of excursion. After measuring Bl at the rest posi-

PHOTO 4: A plate with a reflective mark placed in the center of the magnet structure provides a reference spot for the rear measuring laser.

PHOTO 5: A frame inside the test chamber mounts the internal laser.

PHOTO 6: The internal laser determines the precise position of the rear of the motor structure.

tion, acoustic pressure is used to offset the diaphragm to the new desired position, and a test tone in the mass-controlled frequency range is inserted from a current-source amplifier. A current-source amplifier is used to eliminate the effects of any changes in voice-coil inductance at various positions. *Photo 9* shows some of the controls and meters on the custom current amplifier.

The resulting acoustic output measured by an internal microphone is proportional to the Bl, with little influence from stiction, and other nonlinear effects (*Photo 10*).

Measurements at different positions continue until a decrease in Bl to at least 50% of rest value—and even further if suspension compliance allows—is reached in both directions. The position in each direction where Bl reaches 70.7% of the rest value is designated as $X_{MAG}$ for that direction. *Photo 11* shows David Clark of DLC Design "at the console" making a measurement.

Moving mass ($M_{MD}$) can be measured in one of two ways. The first involves using the drivers' magnetic force to balance the force of gravity acting on the mass while keeping the diaphragm in the same position. Very high accuracy in measuring the diaphragm position in relation to the driver basket is required to ensure that errors due to suspension movement are eliminated. The test chamber with the driver mounted is rotated to the upward vertical position while maintaining the diaphragm position by injecting current into the voice coil. The amount of current needed to maintain that position is then measured.

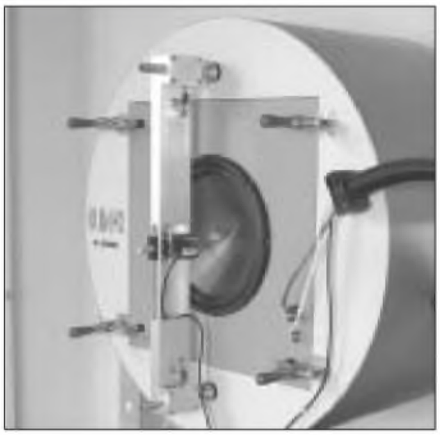


PHOTO 7: The mounting frame for the external laser provides micro positioning capability.

PHOTO 8: The external laser determines the precise position of the dust cap and, therefore, the diaphragm position.

PHOTO 9: The constant-current amplifier provides drive signals, both AC and DC to the voice coil of the driver being tested.

PHOTO 10: Measurements of acoustic output are done by means of an internally mounted microphone.

PHOTO 11: Mr. David Clark, the inventor of DUMAX, has conducted large numbers of measurements using this test setup.

PHOTO 12: The Klippel Distortion Analyzer 1 is the heart of the various modules that make up the Klippel system.

The chamber is then rotated to the downward vertical position while again using current injected into the voice coil to maintain the diaphragm position. The amount of current to maintain the rest position is then measured. The difference between the two drive currents and the Bl force factor is used to calculate the driver mass.

The second method of measuring moving mass is the traditional added mass system, in which a known mass is added to the diaphragm and the change in resonance frequency measured. The use of a laser-pneumatic servo allows for a constant position of the diaphragm even when the mass is added, eliminating the compliance change induced errors that can occur when the suspension is displaced by the added mass in a more typical setup.

Measurement of compliance ($C_{MS}$) is a bit more complicated than the previous parameters. There are actually six possible test methods that can be used, depending on the characteristics of the driver being tested. Some include driving the voice coil of the driver being tested, while others require no connection to the voice coil at all, enabling compliance measurements with the voice coil completely out of the gap, a position where driving the coil would obviously be of no use. These include:

1. $F_S$ oscillation
2. Impedance curve
3. Acoustic maximum
4. Mechanical maximum
5. Voice-coil maximum
6. Acoustic transmission

Compliance is measured at various test points in both directions using static air pressure to position the diaphragm at each of the test points. A more complete description of the tests and results is given in the AES reprint. It is interesting to note that in one case, measurements of driver compliance were made to ±10mm, where the voice coil was only able to provide acceptable drive to a distance of –6mm to +8mm.

Mechanical resistance ($R_{MS}$), which is often assumed to be linear with velocity, is also measured. Nonlinearities that can disturb the normal assumptions can be due to friction, plastic deformation, and aerodynamic drag. In the DUMAX scenario, this parameter can be measured at different diaphragm positions.

Measuring voice-coil resistance ($R_E$) is pretty straightforward. The most accurate method is to use separate measurements of the current through the voice coil and the voltage across the coil at the same time. You can do this yourself with two digital multimeters and a variable low voltage power supply.

Insert one DVM set to the DC current mode in series between the power supply and the voice coil. Next connect the other DVM set to the DC voltage mode across the voice-coil contacts. Slowly increase the voltage to the driver until a few hundred milliamps is measured with the DVM set to the current mode. Note the voltage at that point. You might notice a slight drift in current as the voice coil heats up from even that low level of power.

To determine the resistance, divide the voltage reading by the current reading, being careful to allow for decimal positions if using milliamps or millivolts instead of amps and volts. This is a far more accurate method than trying to use a DVM in the usually least accurate part of the resistance range and also trying to allow for lead resistance. With DUMAX, this four-wire measurement is made while measuring Bl with added mass.

Effective diameter ($S_D$) is determined by measuring the diameter of the diaphragm and one-third of the surround. Measuring an oval speaker becomes a little tricky, but using a graph paper tracing and counting the squares that are more than 50% inside the one-third surround limit can help. For square or triangular shaped diaphragms, the calculations are easy. For unusual shaped diaphragms, measurements of force or pressure may be used.

Other parameters versus diaphragm position that can be measured using the DUMAX equipment, include:

1. Inductance
2. Semi-inductance
3. Diaphragm acoustic leakage resistance
4. Flux modulation by voice coil current
5. Thermal capacity

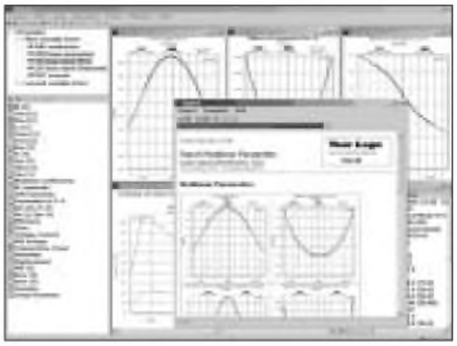
Parameters that are also sensitive to pressure can be measured, including:
1. Diaphragm flex
2. Surround or dome collapse
3. Magnetic fluid stability

Certainly, subjecting a driver to DUMAX analysis will result in far more useful modeling information than just the basic T/S parameters. Not only will limits beyond which the driver will be nonlinear be revealed, but more useful values for modeling parameters will be available, reducing the surprises that occur when going from T/S parameters to finished designs. In addition to graphical reports, measured parameters can be fed into other programs that can predict driver performance.

## KLIPPEL

Like DUMAX, the Klippel Analyzer system enables measurement of both linear and nonlinear parameters of a driver.[2] However, the system is quite different in hardware, software, and operation. While DUMAX measures parameters from normal to extreme operating conditions, Klippel operates the loudspeaker under normal working conditions and finds parameter values that are used to model the driver by digital realtime processing using a DSP.

Although DUMAX and Klippel both produce equivalent values of basic nonlinear parameters, DUMAX actually measures the parameters at the extremes of operating conditions, while Klippel uses the model to predict results under those conditions. Both methods have advantages, but for the purpose of evaluating basic nonlinearities, they both provide the information amateur speaker builders need to evaluate different drivers.



B-2154-1

FIGURE 1: dB-Lab software provides HTML output of the various measured parameters.

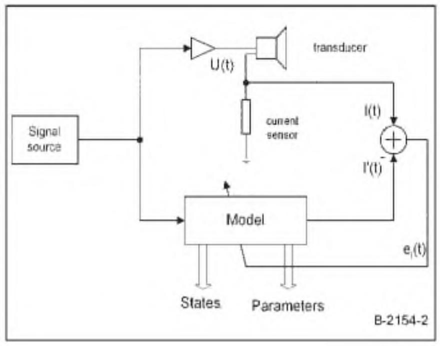



FIGURE 2: Measured parameters of a driver under test are compared to predicted values of a model, which is adjusted to match the measured results.

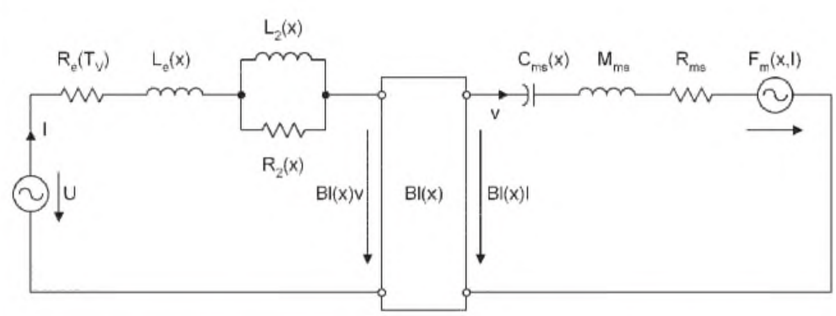



FIGURE 3: The nonlinear elements of a driver can be represented by this equivalent circuit.

B-2154-3

In addition to the basic analyzer, the Klippel system has several different software modules that extend test capabilities to large-signal and linear-signal measurements, numerical simulations of speaker systems in different enclosures, auralization of modeled drivers using a high-quality speaker system to reproduce the linear and nonlinear aspects of the driver under test, and power testing which allows testing of drivers up to the point of destruction with different duty cycles while monitoring various variables and then producing a "death report." All pertinent data from the various tests is stored in a database for analysis and presentation.

The heart of the system is a digital processing unit called the Distortion Analyzer 1 (*Photo 12*) that can be operated in stand-alone or computer-controlled modes. dB-Lab is the master software that controls the distortion analyzer and additional software modules (*Fig. 1*).

Output is in the form of HTML files with standard templates for report generation. Since this software is project oriented, projects may be organized in different workspaces, and custom setups, including comments, logos, and pictures, can be saved as templates for other reports.

Unlike DUMAX, the Klippel system does not use static air pressure to move the diaphragm to various positions in the driver excursion range. Rather, normal music or an audio-like signal (noise) is used to excite the driver, and measurements are taken from the coil current and voltage and, in some cases, position of the diaphragm using a laser for position information. All measurements are taken with the driver in free air or in a sealed enclosure.

PHOTO 13: A driver mounting stand with laser mount allows diaphragm position information to be collected and sent to the Distortion Analyzer 1.

As measurements are taken, the modeling software receives voltage input from the driver terminals and predicts the voice-coil current that will result. The actual measured current is fed into the modeling program, which then adjusts the model until the differences in predicted and actually measured currents are minimized (*Fig. 2*). The resulting model parameters then most accurately reflect the driver parameters under those test conditions.

Testing begins in the small-signal domain and then proceeds in steps until the maximum amplitude is reached based on the identified parameters and thermal and mechanical protection limits. The Klippel system uses software that can make it an automated system with both short-term and long-term tests that can extend for days.

The driver is modeled by an electromechanical equivalent circuit described by the following parameters called state quantities:

1. *x(t)* displacement of the voice coil
2. *v(t)* velocity of the voice coil
3. *I(t)* the input current
4. *u(t)* the driving voltage at the driver terminals
5. *P(t)* real electrical input power
6. $T_V(t)$ voice-coil temperature
7. $R_{TC(v)}$ thermal resistance representing convection cooling
8. $T_M(t)$ magnet structure temperature
9. $T_A$ ambient temperature of the driver before testing
10. $\Delta T_V(t)=T_V(t)-T_A$ increase of voice-coil temperature
11. $\Delta T_M(t)=T_M(t)-T_A$ increase of magnet structure temperature

The relationship between the state quantities is represented by electromechanical and thermal equivalent cir-

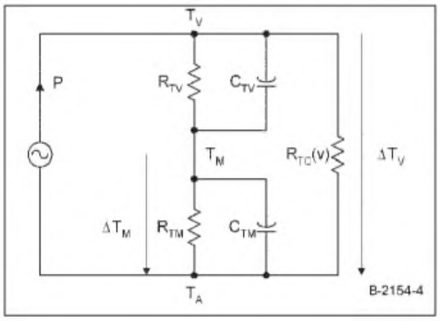
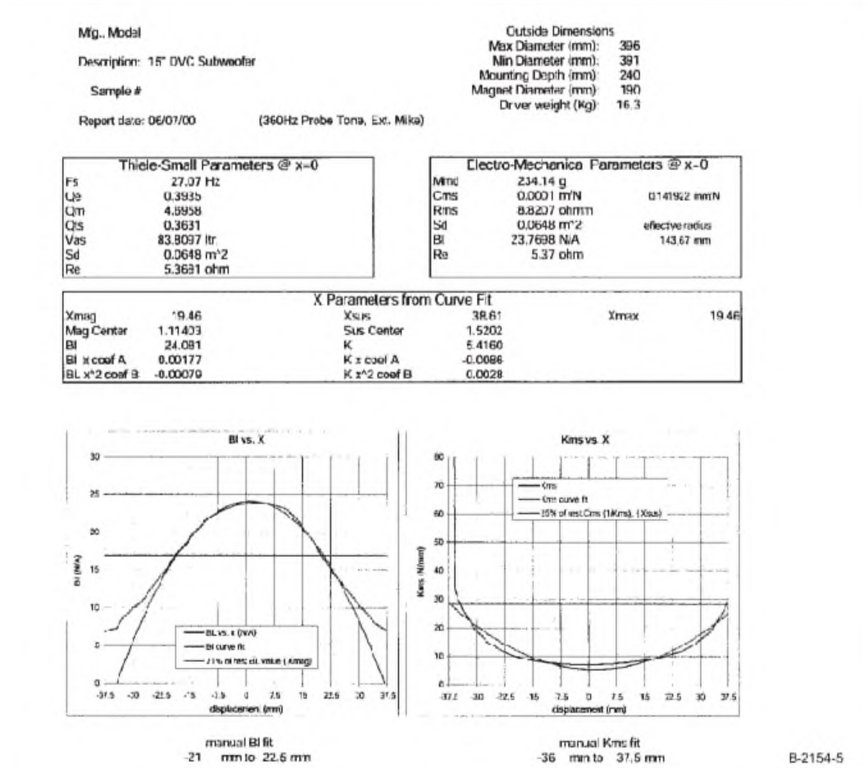



FIGURE 4: Thermal behavior of a driver can be represented by this equivalent circuit.



Mfg., Model

Description: 15" DVC Subwoofer

Sample #

Report date: 06/07/00 (360Hz Probe Tone, Ext. Mike)

Outside Dimensions
Max Diameter (mm): 396
Min Diameter (mm): 391
Mounting Depth (mm): 240
Magnet Diameter (mm): 190
Driver weight (Kg): 16.3

| Thiele-Small Parameters @ x=0 | |
|---|---|
| Fs | 27.07 Hz |
| Qe | 0.3935 |
| Qm | 4.5958 |
| Qts | 0.3631 |
| Vas | 83.8097 ltr |
| Sd | 0.0648 m^2 |
| Re | 5.3631 ohm |

| Electro-Mechanica Parameters @ x=0 | | |
|---|---|---|
| Mmd | 234.14 g | |
| Cms | 0.0001 m/N | 0.141922 mm/N |
| Rms | 8.8207 ohmm | |
| Sd | 0.0648 m^2 | effective radius |
| Bl | 23.7698 N/A | 143.67 mm |
| Re | 5.37 ohm | |

| X Parameters from Curve Fit | | | | | |
|---|---|---|---|---|---|
| Xmag | 19.46 | Xsus | 38.61 | Xmax | 19.46 |
| Mag Center | 1.11403 | Sus Center | 1.5202 | | |
| Bl | 24.081 | K | 5.4160 | | |
| Bl x coef A | 0.00177 | K x coef A | -0.0086 | | |
| BL x^2 coef B | -0.00079 | K x^2 coef B | 0.0028 | | |



FIGURE 5: A basic DUMAX report provides both graphical and numerical data on the measured parameters of a driver.

cuits. Both linear and nonlinear elements are modeled.

Linear elements include:

1. $M_{MS}$ mechanical mass of driver diaphragm assembly including voice-coil and air load
2. $R_{MS}$ mechanical resistance of total-driver losses
3. $R_{TV}$ thermal resistance of path from coil to magnet structure
4. *RTM* thermal resistance of magnet structure to ambient air
5. $C_{TV}$ thermal capacitance of voice coil and nearby surroundings
6. $C_{TM}$ thermal capacitance of magnet structure

Nonlinear elements include:

1. *Bl(x)* instantaneous electrodynamic coupling factor (force factor of the motor) defined by the integral of the magnetic flux density B over voice-coil length l
2. *CMS(x,t)=1/ KMS(x,t)* compliance of driver suspension + air load (the inverse of stiffness*)*
3. *LE(x)* part of voice-coil inductance which is independent of frequency
4. *ZL(x,s)* electric impedance representing the influence of eddy currents

Other elements such as the resistance of the voice coil that varies with the voice-coil temperature are also derived. The equivalent circuit for nonlinear behavior is shown in *Fig. 3*. The equivalent thermal circuit that describes the heating properties of the driver is shown in *Fig. 4*.

Hardware involved in making the measurements includes:

1. The Distortion Analyzer 1 with AD and DA converters, current and voltage sensors, and DSPs
2. An audio power amplifier
3. Cables for connecting the driver and amplifier
4. An IBM-compatible computer with USB interface
5. Analyzer and simulation software

In addition, a laser can be used to measure the displacement of the diaphragm (*Photo 13*).

## DRIVER MEASUREMENT

Measurements start by mounting the driver in free air or in a sealed box, free air being preferred. A noise signal as specified in IEC 60268-1 is applied to the amplifier powering the driver under test. Part of the function of the Distortion Analyzer is to limit the drive signal to a level that represents a safe working range for the driver in terms of maximum power and displacement. There are five steps in the test:

1. All of the drive and measuring parts of the system are tested before a signal is applied to the driver under test.
2. Small signal measurements are made.
3. Operating range of the driver is determined by slowly increasing the drive signal until one of the limits of safe operating range is reached.
4. Thermal parameters are measured.
5. The learning speed of the update algorithm is reduced to minimize the

effects of measurement noise on the parameter estimates. Longer term measurements are then done to monitor variations of driver parameters and thermal resistance and capacitance of the magnet structure.

Resulting data is then used to provide both small- and large-signal parameters, including nonlinear parameters and distortion analysis.

Two of the Klippel system capabilities I found particularly interesting were the ability to simulate a complete system including driver and enclosure and the ability to take the modeled behavior of a driver and pass it through a high-performance speaker system to hear the various distortion components. The two modules are called Simulation (SIM) and Auralization (AURA).

At first, SIM may sound like any of the box tuning programs available. Not so. It provides a spectral analysis of proposed combinations. Dominant nonlinearities of the driver, enclosure, and radiation are considered. For example, air compression and port noise can be considered. In addition, each nonlinearity can be switched off to examine its effect on the system.

AURA is even more fascinating. By transferring all the measured driver parameters into this module, either test signals or music can be passed through the "driver." Actually, as the signal is passed through the module, it is modified by the nonlinearities measured for the driver.

The resulting signal is passed through a high-quality audio system to allow listening of the effects of the driver nonlinearities. A high-quality tweeter simulation fills out the test signal to provide full-range output for the listening test. As with SIM, the various nonlinearities can be removed from the "driver" to see what effect reducing those nonlinearities would have on the resulting sound. These tests can be run both as open A/B or blind comparisons.

## CONCLUSIONS

I hope that you have been able to get a flavor of the depth of information that is available from these systems. Should you, as an enthusiast, rush out and buy one of them? I don't think so, unless you plan to spend upwards of the middle five figures and also a lot of time learning and doing analyses.

More realistically, many suppliers are now starting to specify realistic $X_{MAX}$ values as measured by DUMAX or Klippel. If your vendor does not, you might wish to drop them a note saying they are only providing half a loaf when it comes to their data. T/S parameters are fine, but they don't cut it under real-life drive situations. Not knowing what the driver is going to do when faced with higher-than-whisper-level signals is a severe limitation in any speaker design prcject.

Both companies will also provide testing of drivers you supply. At the time of this article, the going rate for a basic DUMAX report is $100 and for a basic Klippel report $310.

*Figure 5* is an example of a basic DUMAX report on a high-quality 15″ driver. As you can see from the results, this driver is limited by the length of the magnetic field rather than the linear travel of the suspension. $X_{MAG}$ is 19.46mm, while $X_{SUS}$ is 38.61mm. The resulting $X_{MAX}$ is 19.46mm.

DLC has extended an offer to *audioXpress* readers and clubs who want to test drivers. For a period of six months from the publish date of this article, if you mention *audioXpress* when you send a driver to them for testing, the price for a basic test report will be $75. Send the driver with a check for $75 plus shipping costs. Be sure to include your e-mail address.

After testing, the test report will be sent to you via e-mail and the driver shipped back to you. If you have a

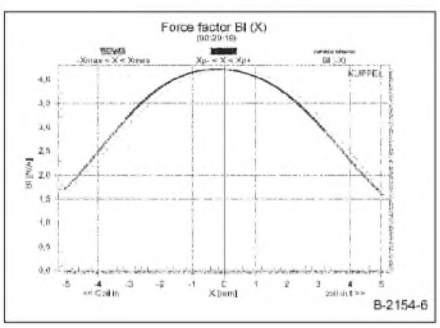



FIGURE 6: This Klippel graph shows the driver Bl product vs displacement.

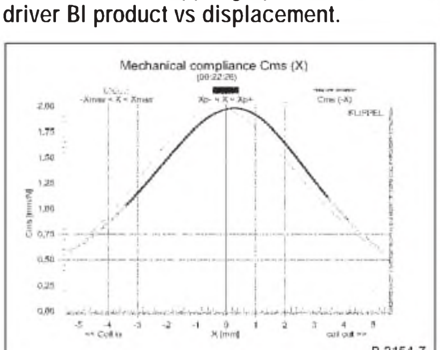



FIGURE 7: Driver compliance vs displacement from Klippel.

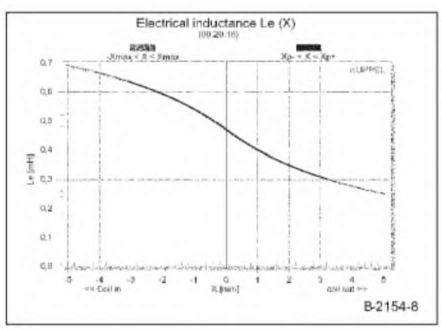



FIGURE 8: A graphical presentation of inductance nonlinearities that will cause distortion.

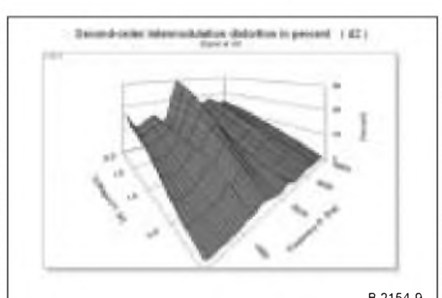



FIGURE 9: This Klippel 3-dimensional graph shows IM distortion vs both drive level and frequency.

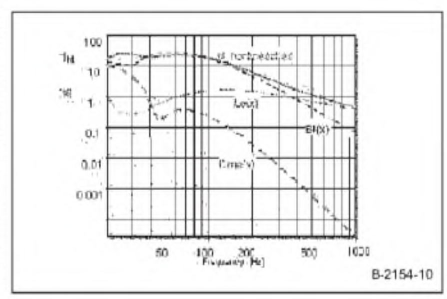



FIGURE 10: The Klippel SIM module predicts the total distortion of all nonlinearities of a driver in a specified enclosure as well as the contributions of each separately.

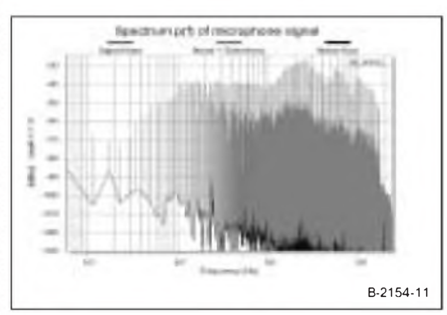



FIGURE 11: Measurement results using a multitone excitation signal show the signal, distortion + noise, and the noise floor. Distortion at higher frequency is mostly IM, which would not be visible in harmonic distortion measurements.

group of people in your club who are interested in a particular driver or if you are doing a project yourself such as line arrays where multiples of the same driver will be used, this would be a good value.

Klippel has also made a special offer for *audioXpress* readers. They will provide an HTML format basic report showing linear, nonlinear, thermal parameters, and results of distortion analysis for $250. Again, freight costs are at your expense. Klippel reports can vary widely due to the large number of tests that can be performed. *Figures 6–8* show just a few of the results you can expect from a basic test. Three examples of some of the results obtainable from the optional modules are shown in *Figs. 9–11*.

The Klippel website also has a wealth of information in the form of papers and descriptive information you can download. Just reading his papers can give you a much broader understanding of how dynamic drivers work and what problems are encountered. The product information is broken down into an overview and then individual pieces on each module and the hardware.

Both of these systems have been designed by people who know how to deal with the foibles of dynamic drivers. In addition to their own work, their papers point to a great deal of reference material done by other people. My suggestion is to get through as much of the Clark and Klippel work as you can handle to give yourself a better understanding of dynamic drivers before making choices for your next project. The time will be well spent. ❖

## REFERENCES

D. Clark, "Precision Measurement Of Loudspeaker Parameters," J Audio Eng. Soc. Vol. 5, pp. 129–141 (1997 March).

W. Klippel, "Distortion Analyzer—a New Tool for Assessing and Improving Electrodynamic Transduction," presented at 111th Convention of The Audio Eng. Soc., February 19–22, 2000, Paris, France, preprint 5109.

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# Constructing the SEAS Froy Mk. 3

*By Edward T. Dell, Jr.*

The tiny vented SEAS Froy Mark 3 design turned up on the Madisound website (www.madisound.com) while I was searching for details on THOR[1]. The Froy (*Photo 1*) is a small MTM utilizing the SEAS Excel Millennium technology drivers. The specs looked impressive. The structural drawing (*Fig. 1*) is from SEAS. Its only unusual feature, other than small size, is the ½″ slot port centered in the rear panel which also adds rigidity to the one-square-foot enclosure.

## FEATURES

The drivers for each Froy include two W15CY001 mid/woofers and one T 25 CF 002 tweeter. The 5″ mid/woofers use the same type of machined magnesium cones as the larger ones used in the THOR design. SEAS says the rear vent directs any possible port noise away from the listener.

The crossover, set at 2200Hz, includes a Zobel (L2 + C2) in parallel with the paralleled mid/woofers. The crossover schematic is in *Fig. 2a*; the layouts for the crossovers in *Fig. 2b* (low-pass) and *Fig. 2c* (high-pass). The latter two drawings are full-scale and work as templates.

The Froy design was originated by Murray Zeligman, including this latest Mk. 3 iteration. Frequency range is claimed to be 40–25000Hz, with a maximum short-term power of 300W and long-term power of 140W and 4Ω impedance. On special order through a dealer, SEAS can provide magnetically shielded drivers for this design where it might be used as a center channel speaker for a home theater system.

**PHOTO 1:** Finished Froy Mk. 3 system.

## MAKING THE CUTS

Building stock for the Froys is ¾″ MDF (medium-density fiberboard) requiring about a third of a 4 × 8′ sheet. The cutting guide is *Fig. 3*. If you make the four vertical cuts as suggested, then the short cuts form the sides, fronts, backs, tops, and bottoms. The remaining pieces may be cut to form the vent walls.

Adding to the SEAS instructions, I used a ⅛″ roundover bit to shape the inner edges of the two sides of these port walls (*Photo 2*), which may make for a smoother transition into the port and possibly ease some diffraction turbulence. I decided to leave cutting the ½″ slot until after box assembly, since otherwise the MDF is soft and likely to break (*Photo 3*).

You might consider making the panels which form the vent ¼″ longer (4¼ × 6 × ¾″) and cutting two ⅛″ deep × ¾″ wide dadoes in each side wall to make mounting the port walls easier. As an alternative, I mounted the walls using #8 × 1½″ flathead screws (*Photo 4*)—one to each of the three vent mating edges—and sank them ¼″ below the outer surfaces of the box (sides and back). I used a plug cutter in scrap plywood stock to plug the screw heads, sanding the plugs flat after the glue dried. Plywood works better because the MDF plugs break too easily.

As with my THOR project[1], I used biscuits for assembling every joint except the vent walls (*Photo 5*). I used far fewer this time, making doubly certain that the fit of each joint was snug. I labeled each of the parts indicating left and right, inside/outside, back, front, sides, top, and bottom. This is enormously helpful in keeping track of where you are during assembly.

I marked mating edges to simplify assembly, and also to keep track of the biscuit locations. It is a good

**PHOTO 2:** The Froy port is unusual located midway between top and bottom of the cabinet. The inner sides of the port walls have been rounded to smooth diffraction in the port. Screws were used to fasten the port walls, and pilot holes are visible for cutting the back opening later with a jig-saw.

**PHOTO 3:** Opening the rear port after the box is assembled.

**PHOTO 4:** Countersunk screws hold the port walls in place during assembly.

idea to mark these with either an "E" for the edge cuts or an "F" for face cuts. The Froy front and back panel biscuit cuts are all face cuts, the sides are all edge cuts, and the top and bottom require two of each—face cuts for the sides and edge cuts for the front and back.

**PHOTO 5:** Dry assembly of the box before gluing ensures a good fit of joints. Biscuits were used on all joints except the port walls.

**PHOTO 6:** Threaded brass inserts were used for mounting the drivers.

I used a router to cut both the dadoes for the driver rims, using the Jasper® jig, and carefully made test cuts for both the correct diameters and the respective depths. I find these easier to do in each panel (*Photo 6*), before assembling the box. I used a ¼″ bit for the driver holes with the jig as a guide.

During construction of the Froys, I ordered and received a new DeWalt router which has a dust collection port you can attach to your shop vacuum. Although slightly more difficult to manage with a 1¼″ hose attached, it is virtually dustless during operation, in sharp contrast to the standard unvented router (*Photo 7*).

## MOUNTING THE DRIVERS

One of the major problems I encountered in building the THORs was attempting to use brass inserts and #8 machine screws to mount the drivers. The problem is marking the locations for the ¼″ holes for the inserts. The holes in the rims of the drivers can accept a ³⁄₁₆″ bit, but nothing larger. How do you make sure that you drill the necessarily larger ¼″ hole ex-

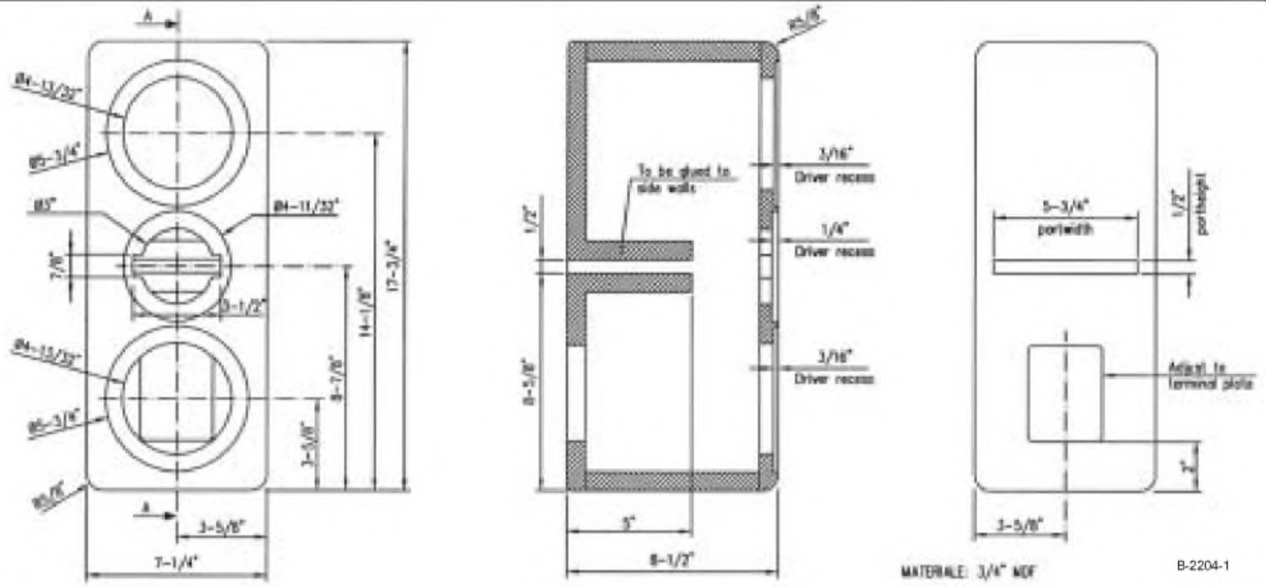



**FIGURE 1:** The SEAS cabinet drawing. The original was drawn in metric, which accounts for the tight diameters for the driver openings. The outer dimensions for cutting the dadoes are reasonable using the Jasper guide. The inner diameters are not so critical since they must only clear the cast frame of the drivers. Note that the sides of the cabinets fit inside the fronts, backs, tops, and bottoms.

actly centered on the pilot hole?

Thinking hard about this one day, far from my shop, I found a solution, which popped unbidden into my head. I placed the driver in its dadoed mounting shelf, face downward. I placed the front panel in the drill press and carefully drilled one ⁵⁄₁₆″ hole, through the mounting flange of the driver as guide. I removed the driver, leaving the bit in the chuck.

Then I lowered the bit into the pilot hole and clamped the panel to the drill press table. I removed the ⁵⁄₁₆″ bit and installed the ¼″ one, and drilled the mounting hole for the threaded brass insert. I then installed the first brass unit.

Next—and I admit I'm being super fussy about this—I repositioned the driver, face down, and fastened it to the baffle with the first machine screw. I replaced the ¼″ bit with the ⁵⁄₁₆″ one, and drilled a second pilot hole, again using the driver as a guide, followed by the ¼″ hole. I then installed the second threaded insert, and again remounted the driver, now with two screws.

Next I drilled the remaining four ⁵⁄₁₆″ pilot holes. I removed the driver and inserted the bit into each pilot hole, clamped the panels to the drill press table, and drilled the remaining ¼″ insert mounting holes, one by one. I followed the same procedure for the tweeters. After installing the threaded inserts, the drivers were all mounted perfectly.

Take care in installing the brass inserts. Be sure to drive them in at 90° to the face of the panels. I found this easiest with a very large screwdriver, strangely enough. The final two or three turns must be done with a narrower-bladed tool, driven slightly below the surface of the stock. Clean up any expanded material with a sharp knife or chisel.

I found a tip in an old copy of *Shop Notes*[2], which suggested using a small three-cornered file to make a relief trench across one side of the insert threads, at right angles. Clamp each insert in your bench vise for this procedure. This apparently helps relieve expansion pressure avoiding bulges around the top of the insert. I discovered this suggestion after installing all my inserts, of course, but it does sound as though it should alleviate the bulges.

## ASSEMBLY

I assembled my crossovers on ⅛″ hard masonite® bases with copies of the paper templates (*Figs. 2a* and *b*) attached with spray adhesive (*Photo 8*). I drilled holes for

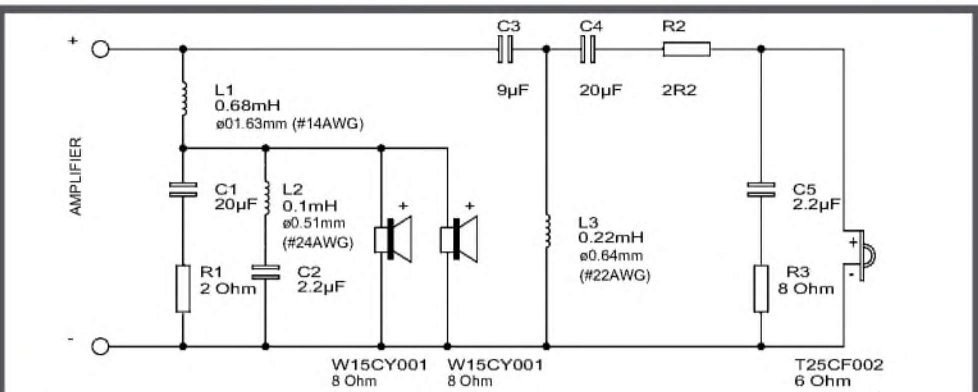



**FIGURE 2A:** Froy crossover schematic. The coils are all air core; the capacitors are either polypropylene or polyester foil; the resistors are all metallic oxide or non-inductive wire wound.

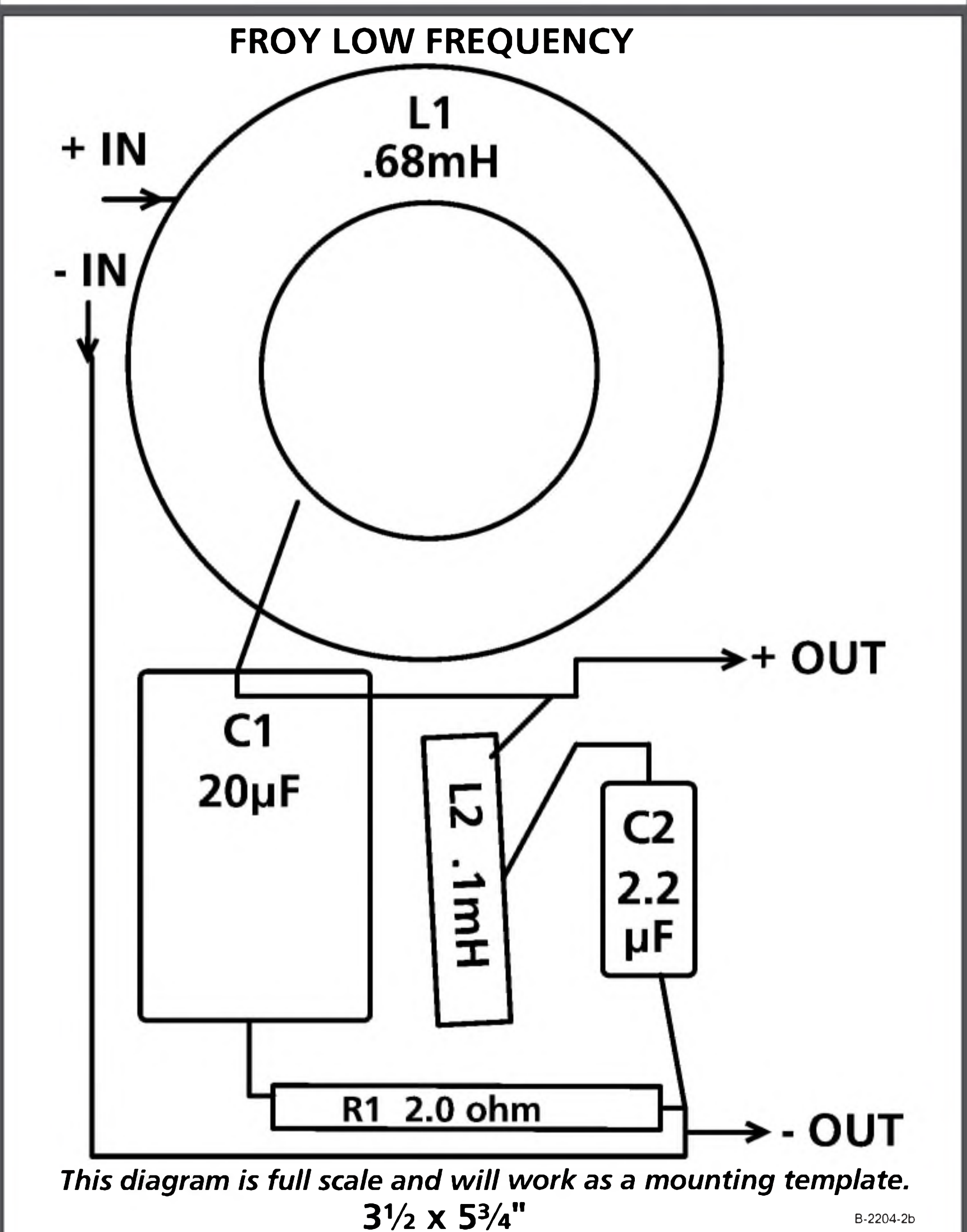



**FIGURE 2B:** Froy low-frequency crossover template, which can serve as a mounting guide. Minimum size is 3½ × 5¾″.

the cable ties to mount each component. I used staked terminals for input and output connections. I put the high-pass and low-pass components on separate boards to allow for bi-wire and bi-amp connections.

The input terminal plate with separate HF and LF is rather large, so I turned it sideways at the bottom of the rear panel (*Photo 9*) to leave room to mount the tweeter board. It seemed sensible to mark mounting hole locations for the crossover boards and drill pilot holes for them (two each) inside the rear panel before assembling the box. The LF board fits easily on the rear wall (*Photo 10*) of the upper half of the box.

Gluing the box together should be preceded by a complete, dry assembly. A few of the biscuits may need "encouragement," or the slots re-sawed. Vacuuming the slots ahead of time helps, as well.

Once everything is fitted, if you do as I did, clamp the dry assembled box firmly, fitting the port side pieces in their proper locations. Mark their positions on the sides and back panel, both inside and outside, and use a tool that cuts a pilot hole for the screw and its countersink as well as a recess for a glued plug atop the screw. Attach the vent sides to the back panel first, then to the sides. Remove the screws and disassemble the box.

Using screws to hold the vent walls is also helpful in the first stages of the gluing process. Begin with glue on the back panel and on one vent wall edge, and attach with its pre-drilled screw, making sure the wall is accurately located. Next, attach the sides, then the ends and, finally, the front panel. Clamp the entire assembly. I used 12 clamps (*Photo 11*), cleaning off the glue oozing from each joint with a damp paper towel.

## FINISH WORK

After each box has dried overnight, it is time to seal any gaps with plas-

**PHOTO 7:** All box edges were rounded with a router, 1/8" bit. The router is a new style DeWalt equipped with a dust port for connection to a vacuum.

**PHOTO 9:** The opening for the terminal block is opened after box assembly, and mounting screw locations marked for drilling.

**PHOTO 11:** Gluing uses all 12 of my clamps.

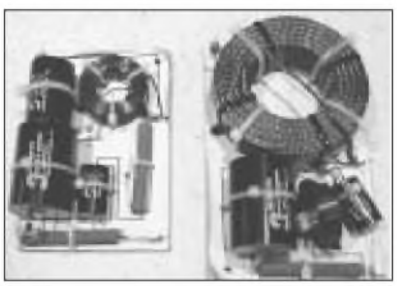


**PHOTO 8:** The two assembled crossover networks, HF on the left, LF on the right. Components are held in place with cable ties. Stud terminals are press-fit into the 1/8" Masonite.

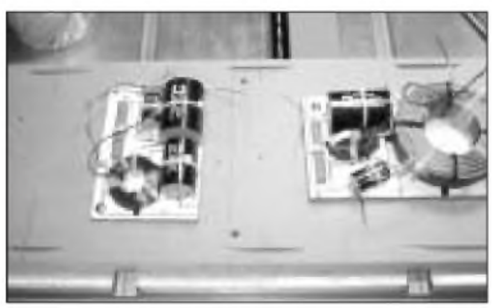


**PHOTO 10:** Crossovers were pre-mounted on the back panel prior to box assembly, then removed prior to the gluing. This eases later crossover installation.

**PHOTO 12:** A 1/4" bit rounds the exit of the port. Note the plywood plugs covering the screws which secure the port walls.

tic wood and sand them to smooth any mismatched joints. If you use a belt sander, which is a potent tool, equip it with 80 or 120 grit and keep the device moving at all times. It can reshape the panels in ways you won't like if you're not very attentive. Finish the job with a palm sander.

Use the ⅝″ roundover cutter in your router to round each of the cabinet's edges, except around the back edges. If you have a router table, the job is much easier.

A jig saw completes the job of opening the port tube, after having pre-drilled ¼″ holes in opposite corners of the marked area before assembly. The same method works in opening the space for the input terminals. I used the router with a ¼″ roundover bit to run around the port exits (*Photo 12*), again guessing that rounded edges of a port, however modest in size, might minimize diffraction effects.

Before putting two coats of oil-based primer (on the advice of a local paint vendor, even though I thoroughly dislike cleaning oil-based paint from brushes), I covered each of the brass mounting inserts (*Photo 13*) with masking tape. My three coats of finish paint was water-based and mixed a near-black shade slightly darker than the primer. Each coat was followed by palm-sanding with 180 grit paper, after coats 1–3 and 400 after coat 4 (*Photo 14*).

Attaching the interwiring on the inputs and outputs of the crossover boards is next (*Photo 15*). I used #14 AWG stranded copper, leaving enough slack to reach the three drivers in each box. A bit of tape identified the tweeter lead. Putting these in place and installing the mounting screws with a shorty Phillips-head driver is a bit tricky, but do-able. Thread the connecting wires to their respective driver opening and prepare the stuffing.

SEAS recommend 80 grammes of Dacron® wool to lightly stuff each box. Thickness is about 5cm, or 360 grammes per square meter. I used 40 grammes of high loft quilting in each compartment behind the mid/woofers, avoiding blocking the port. Cut the leads to a length making access to the drivers possible and solder each to their respective terminals, being sure to observe polarity, of course (*Photo 16*).

I placed a single bead of Mortite® beneath the rims of each of the mid/woofers before installing the drivers with #8 machine screws. The tweeter openings were very snug and did not need sealant. Be careful with the tweeter because the dome protrudes beyond the mounting frame.

The shipping packing includes a protective insert which you can use in the mounting process. SEAS offers a replacement for the tweeter diaphragm, coil, and connector assembly (which they call "a butterfly"), if you damage yours. Mount the mid/woofers last. It is a

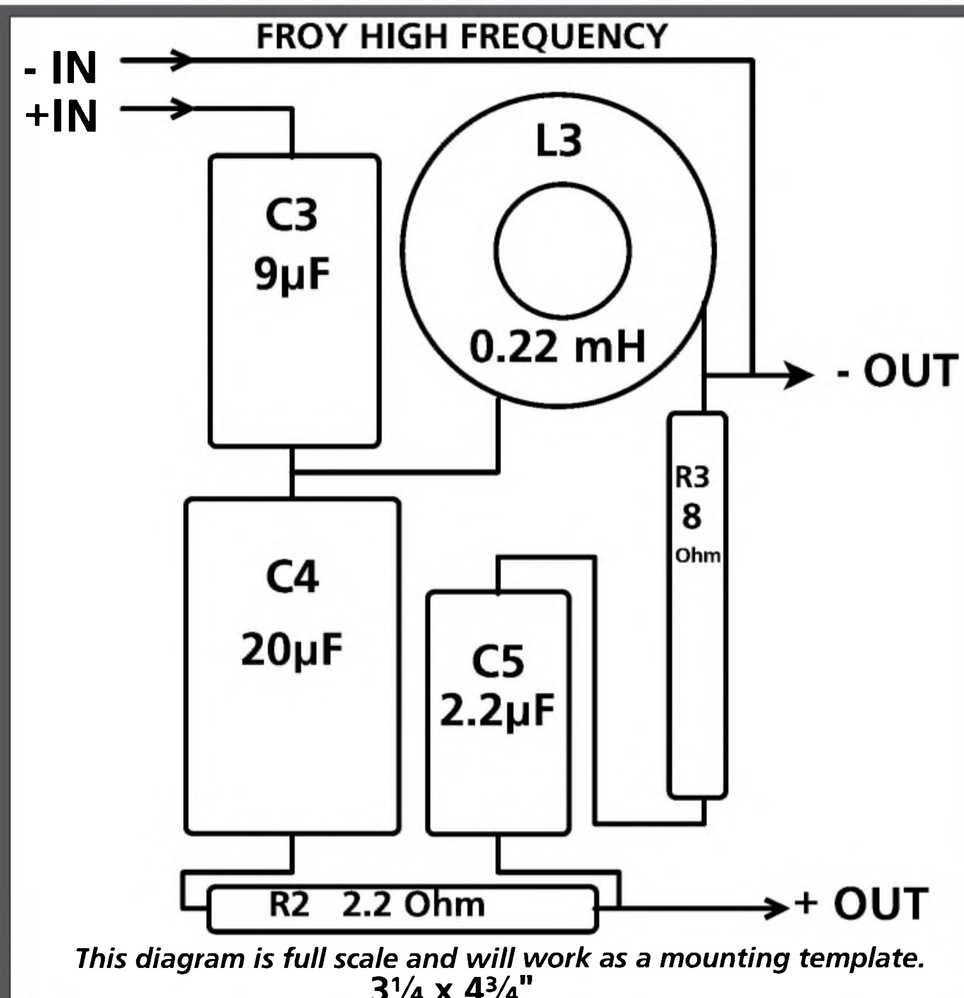



**FIGURE 2C:** High-frequency crossover template. Minimum size 3¼″ × 4¾″.

**PHOTO 13:** Second primer coat is complete, all mounting inserts covered by masking tape.

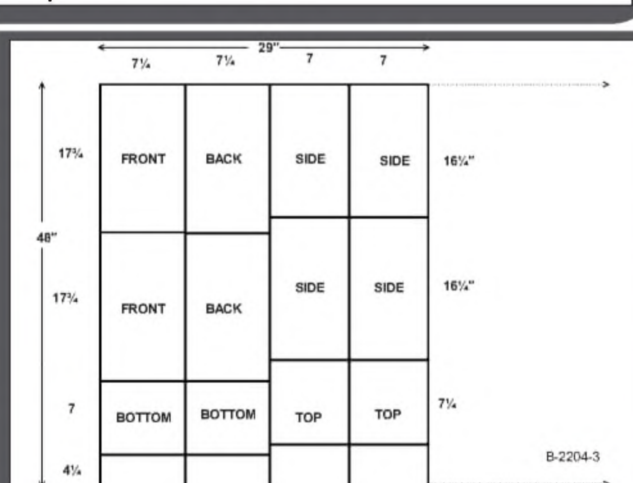



**FIGURE 3:** Cutting guide. After making the four vertical cuts and the cross-cuts for the fronts, backs, sides, bottoms, and tops, cut the remaining small scrap pieces to 4¼″ × 5¾″ to form the sides of the ports.

good idea to test continuity and polarity of each driver by using a battery on each one, positive to positive, negative to negative. The motion of the cones should be outward in every case.

Next issue, you will find Joe D'Appolito's measurements of Froy Mk. 3, accompanied by a listening evaluation from Dennis Colin. How did I react to them? Before I saw Joe's report I was frankly surprised how close they were in sound profile to the THORs. I noticed a slight edginess in the upper end, which is surprising since they have the same tweeter as the THORs. They do require stands, which should be 26″ high, to lift the tweeters to ear level. They have excellent horizontal dispersion, although like the THORs, the vertical sweet spot is quite narrow.

Building this pair was a very great pleasure, as speaker construction always is for me. ❖


## REFERENCES

1. "Building the THORs," *audioXpress*, Sept. 2002, pp. 14–23.

2. *Shop Notes*, Vol. 6, #34, p. 5, July 1997, www.augusthome.com.

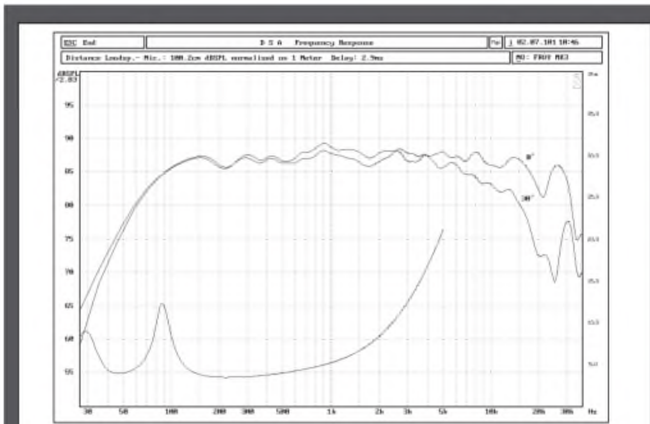



**FIGURE 4:** SEAS frequency response chart for the Froy system. B-2204-4

## TABLE 1
### SEAS FROY MK. 3 BILLS OF MATERIALS

| QUANTITY | ITEM | DETAIL |
|---|---|---|
| 4 | W15CY001 | SEAS mid/bass driver |
| 2 | T25CF002 | SEAS tweeter |
| 2 | L1 | 0.68mH, 1.63mm #14AWG |
| 2 | L2 | 0.10mH, 0.10mH #24AWG |
| 2 | L3 | 0.22mH, 0.64mH, #22AWG |
| 2 | C1 | 20$\mu$F 250V Polypro or Polyester foil |
| 2 | C2 | 2.2$\mu$F 160V " |
| 2 | C3 | 9$\mu$F 400V " |
| 2 | C4 | 20$\mu$F 400V " |
| 2 | C5 | 2.0$\mu$F 400 " |
| 4 | R1 | 2.0Ω, 10W metal oxide or non-inductive WW |
| 2 | R2 | 2R2Ω " " |
| 2 | R3 | 8.0Ω " " |
| 8 | Dual terminals for crossovers | |
| 2 | Twin cup input with gold 5-way posts | |
| 8′ | #14 cable | |
| 160 grammes | Quilt polyester | |
| 32 | #8 threaded brass inserts | |
| 32 | #8/32 × ½″ stainless round head machine screws | |
| 1 | ⅓ sheet ¾″ medium density fiberboard | |
| Elmer's carpenter's glue | | |
| Sandpaper, Mortite caulk | | |

A number of vendors offer a variety of crossover variations, as well as a variety of complete parts kits. Prices for full kits vary from $850 to $950, depending on options. Some offer finished cabinets for the Froy.

**PHOTO 14:** Fifth coat of paint completed with the box ready for hardware installation.

**PHOTO 15:** One pair of crossovers with connecting wires in place, ready for installation.

**PHOTO 16:** Tweeter is connected, stuffing installed and machine screws ready for final mounting of the drivers.

# Xpress Mail

## MISMATCH AND AUTOFORMER

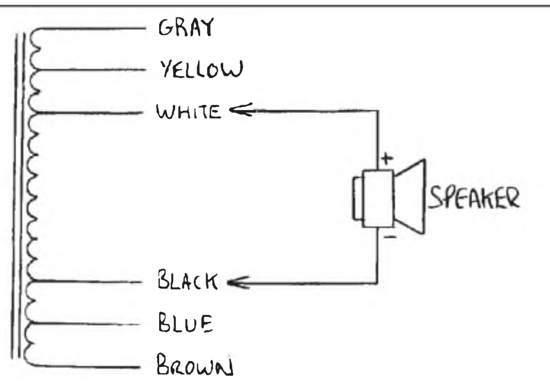
Earlier this week I got the call from my wife that I had long awaited. "Your *audioXpress* came today, and the ZERO autoformer is on the cover!" A quick clock check. I still had two hours left in the day before I could get home to see it.

Thank you for giving my manuscript the opportunity to be published in the January 2003 issue of *audioXpress*. Being a poor writer as a child, I look at this as overcoming one of life's many obstacles. Your staff, including Amity, Marianne, and Peter, has been a pleasure to work with. I thank them for their assistance as well.

A technical note about the article: Figure 1 (marked as document number B-2118-1 on page 40) is a bit goofed up. The original rev 1 proof (dated Aug. 13) showed a transformer with three separate windings. The ZERO autoformer has only one winding with multiple taps. The corrected figure along with a corrected table is shown in *Fig. 1*.

Thanks again for your assistance.

Paul Speltz
Woodbury, Minn.




| SPEAKER'S IMPEDANCE MULTIPLIED BY | CONNECT SPEAKER TO + | − | CONNECT AMPLIFIER TO + | − |
|---|---|---|---|---|
| 2 | WHITE & | BLACK | YELLOW & | BLUE |
| 3 | WHITE & | BLACK | YELLOW & | BROWN |
| 4 | WHITE & | BLACK | GRAY & | BROWN |

FIGURE 1: Corrected speaker impedance multiplier.

B-Spelt-1

## MARANTZ REPLICA

The following is not intended to denigrate Mr. Kobayashi or his article ("A Marantz 8B Replica," July '02 *aX*, p. 14) in any way. I read this article, and found it very interesting; however, the measured performance of his design, using the latest modern components, including the Plitron toroidal output transformers, was inferior to the earlier replica, which he built, using Mr. Uesugi's design.

Please correct me if I'm wrong, but the only logical explanation I can see for this is the use of cathode bias for the new design, as opposed to fixed bias for the earlier design, and the lack of the components mentioned in the article, used for compensation for frequency and phase aberrations in the feedback and gain circuits.

I wish the author could confirm or refute my speculations on this, as this is a project I would seriously consider building, and if I am correct, would prefer to build the earlier (Uesugi) design, rather than the later design in the article.

Jerry Boncer
Carlsbad, Calif.

*Satoru Kobayashi responds:*

*Thanks for your interest in my article. Your comment is "true," as you can see from the circuit diagram. My design is not intended to copy the well-known Marantz 8B amplifier, but to make a simple EL84pp amplifier with a state-of-the-art toroidal transformer, which might exceed the performance of this famous amplifier. As you indicated, the amplifier might not inherit the essence and/or concept of the original Marantz 8B amplifier, so it should be called, "A simply designed EL84pp amplifier using a Plitron toroidal transformer."*

*In terms of performance comparison between my amplifier and Mr. Uesugi's Marantz 8B replica amplifier design, there are several ways to characterize the amplifier performance. I focused on the power bandwidth brought by Mr. Van der Veen's toroidal transformer design rather than a low level of distortion as Mr. Uesugi's replica amplifier showed using a lot of NFB. I am very proud of being an evangelist for the use of toroidal transformers, which offer a much wider frequency response than a conventional E-I cored transformer.*

*So I would like to maximize this feature with non-NFB that was recommended by Mr. Van der Veen as well. I knew that this might change a direction or a concept to replicate the original Marantz 8B. In addition, I would have liked to simplify the circuit so that any beginner might be able to replicate this amplifier without any extra labor to adjust an idling current.*

*If you are experienced in the assembly of this kind of amplifier, you might modify the grid bias circuit from a self cathode bias to a fixed bias using extra components to increase the linearity and decrease distortion a little bit . . . and also to increase NFB, as Uesugi did to decrease the distortion. Then it might achieve better performance than this.*

*So far I have communicated with Mr. Van der Veen through e-mails, and am very confident because his design criterion was well accepted and verified by non-NFB with this toroidal transformer, which brings a wider frequency response without NFB than an E-I transformer with NFB.*

*Finally, it would really be up to you whether to build the earlier Uesugi designed amplifier, with the circuit optimized with the well-known Japanese-made LUX OY series output transformer, featuring a good 1960s-*

*era performance. However, if you have never used such a toroidal transformer before, I strongly suggest you try this transformer, no matter what circuit you choose. You might achieve the sound clarity, which really exceeds a conventional E-I cored transformer.*

*I strongly believe that such a wide frequency range amplifier using this toroidal transformer will work with state-of-the-art wide frequency speaker systems such as B&W and so on. If you have old classic-style speakers, then Mr. Uesugi's designed replica might fit into those.*

*As of today, I have received at least two letters from Japanese enthusiasts who have copied this amplifier, and are very pleased with the results. One has modified a power supply from my original to his own design using a couple of toroidal power transformers, though I believe the sound might be identical, but he mentioned he has a little problem with hum due to a difference between his floor design and my original design.*

## TUBE ALTERNATIVES

I have some alternative tube types to suggest to readers who may wish to design power amplifiers.

The 300B and 2A3 family (6A3 and 6B4G) of tubes are among the most popular of the triodes. All can provide sufficient output power at low distortion in Class A or AB1 for many applications. All have good plate dissipation ratings and low plate resistance, both to the good. However, they also have low amplification factors, which makes them relatively difficult to drive.

The type 8417 beam pentode triode connected is outstanding. Its plate dissipation rating triode connected is 40W, with amplification factor about 16, plate resistance around 700Ω, and Gm approximately 23000 siemens. The 8417 as a triode would be a good successor to the 300B; as a pentode it can deliver up to 100W.

The 6BX7 dual triode is a good candidate for lower power applications. Its plate dissipation rating is 10W/section, 12W maximum for both (temperature limited). Amplification factor is 10, plate resistance about 1300Ω (650Ω with both sections in parallel), and Gm about 15000 siemens for the parallel arrangement. While the 6BX7 has a slightly lower plate dissipation rating than the 2A3—12W versus 15W for the 2A3—it can deliver as much power depending on operating conditions. Since the 6BX7 is a dual triode, a single tube could also be used in a push-pull stage at lower output power, of course.

Both the 8417 and 6BX7 are much easier to drive than the types they might supplant, which is a significant advantage. The 8417 is not widely listed but is still available, according to inquiries I have made, and is about the price of the 300B. The 6BX7 is listed in current ads and is relatively cheap.

J. L. Markwalter
Port Charlotte, Fla.

## PROJECT COSTS

It baffles me why you don't include any cost-to-build data in the projects you publish. Many of your target audience readers are very interested in lower cost quality. You could help with better info if you wanted.

Bruce Powers
Chelan, Wash.

*This is a good idea which we too often neglect—but not always. Sometimes authors use surplus, in which case an accurate estimate is difficult.–ETD*

## TRUTH IN PUBLISHING

I was pleased to find sections of your magazine available on the web. I was thinking of subscribing, until I did a little reading of the recently published material.

I read a review of the HCA1000A published in June 2001, written by Duncan and Nancy MacArthur. The review was full of statements referring to the "sound" of this amplifier. By avoiding the use of any scientific method, for instance, A B X testing, these people have fooled themselves into thinking that they can listen to and accurately describe amplifier sonic characteristics such as "presence," "sound staging," and "ambience." This is totally ridiculous.

Why your editor allowed an article filled with so much unsubstantiated fantasy listening impressions is beyond me. The technical specifications of this amplifier indicate that no human could hear a sonic signature of this equipment. There is no way the statements made in this article would hold up under simple blind testing. By publish-

**ing this fiction, you have greatly compromised the value of your publication.**

**I do hope you make an effort to return to publishing the truth.**

**Dana Olson**
**Dana.j.olson@medtronic.com**

*Duncan and Nancy MacArthur respond:*

*Double blind testing has already been debated at great length in other venues. If you dislike subjective evaluations, by all means do not read beyond the test results.*

## MORE THOR

**With reference to "THOR: A D'Appolito Transmission Line (May '02 *aX*), I would like to know whether by using a W18EX001 woofer (larger magnet with greater sensitivity—88dB SPL versus 86.5dB SPL of WE18E001), I'm getting better overall sensitivity. In addition, would this change be a one-to-one replacement, or will it affect some other parameter?**

**Atto Rinaldo**
**Tambre, Italy**

*Joe D'Appolito responds:*

*Yes, exchanging the W18EX001 for the W18E001 will increase your sensitivity above 500Hz by 1.5dB, but there is a price to pay. You will lose low-frequency extension. Low-frequency $f_3$ will increase from 47Hz to about 59Hz. As always, there is no free lunch.*

*The problem with using the EX version is its low Qt. This leads to a very overdamped system. The EX $Q_{TS}$ is 0.24 versus 0.34 for the E version. Transmission lines have a very large resistive component loading a driver that adds greatly to mechanical damping. If you use a driver with too low a $Q_{TS}$, the overall damping is much greater than critical. Remember $Q_{TS}$ for the W18E001 in the THOR transmission line is already 0.55. Substituting the EX driver will drop $Q_{TS}$ to about 0.4, which is too overdamped.*

*An example here will help. This example uses a different driver in a transmission line I designed for a different application, but the results illustrate the point I want to make. The first column of Table 1 lists the free-air parameters for this driver.*

*You know when the driver is placed in a closed box the resonant frequency will increase and all the Qs will increase by a similar amount. Column two of Table 1 shows the change in all driver Qs when you place it in an unlined closed box that increases $F_{SA}$ from 37.6Hz to an $F_{SB}$ of 51.1Hz. Notice in particular that $Q_{MS}$ increases from 6.9 to 9.5. All other Qs increase by the same ratio. In particular, $Q_{ES}$ increases from 0.37 to 0.51.*

*The situation is quite different when you place the driver at the input end of the transmission line. Looking at column 3 of Table 1, you see that $f_s$ increases to 51.1Hz and $Q_{ES}$ rises to 0.51 as it does in the closed box, but $Q_{MS}$ actually decreases to 2.25 and $Q_{TS}$ is 0.42 versus 0.48 for the closed box. The resistive loading on the woofer by the transmission line decreases $Q_{MS}$, further damping the woofer.*

**TABLE 1**

| | FREE-AIR | CLOSED BOX | TLINE |
|---|---|---|---|
| $F_{SA}/F_{SB}$ | 37.6 | 51.1 | 51.1 |
| $Q_{MS}$ | 6.9 | 9.5 | 2.25 |
| $Q_{ES}$ | 0.37 | 0.51 | 0.51 |
| $Q_{TS}$ | 0.35 | 0.48 | 0.42 |

## SAVE THE HORN

**It occurred to me after reading Robert Roggeveen's article ("Build Your Own Axisymmetric Tractrix Horn," Sept. '02 *aX*) that there might be a solution to the destruction of the horn mold when removing it from the mortar. Instead of adding the individual cone segments from the outside insert them on the inside of the previous ones. This would mean that the steps would be facing toward the top of the completed mold and would not act like a "fish-hook." Provided the completed mold is made of impervious material or treated in some way to repel moisture and also to discourage adhesion of the mortar, then I would imagine that the mold would be re-usable.**

**Ross Herbert**
**Carine, WA**
**Australia**

*Robert Roggeveen responds:*

*I thank Ross Herbert for his problem-solving efforts to ease my emotional pain of destroying a beautiful horn mold. My dear father, who helped me edit the article, suggested the same, and, indeed, in principle the fish-hook effect is reversed when cones are inserted. However, doing so will change the disk-cutting procedure markedly and add layers of construction complexities to gener-*

*ating a working prototype.*

*Most difficult would be the building of the flange, as it would now be produced with large nearly flat ribbons of paper of little stability. The inner circle would now approach the circumference of the flange. Oversizing the disk would increase stability, but I did not pursue this venue of horn-shape making.*

*One of the main criteria for this project was to make a simple prototype for the horn enthusiast to make at home. To see a beautiful mold metamorphosed into a solid horn is not unlike sculpture casting. (Kowal, D., & Meilach, D. (1972).* Sculpture Casting, Mold Techniques and Materials: Metals, Plastics, Concrete. *New York: Crown Publishers.) It's functional art!*

## MIKE PREAMP

I just finished reading Ron Tipton's article ("A High-Performance Microphone Preamp," Oct. '02 *aX*). He might like to know that Burr-Brown/TI has introduced the INA217—a low-noise, low-distortion pin-compatible replacement for the discontinued SSM2017 mike preamp.

I enjoyed every article this month, and was glad to see Martin Colloms writing for *aX*. Another ex-pat from *Stereophile*, it seems.

Chuck Hansen
Ocean, N.J.

*Ron Tipton responds:*

*Thanks, Mr. Hansen, for telling me about the INA217. I missed it when doing my search. I have downloaded the INA217 data sheet from the TI website and I've requested a couple of samples. The original Analog Devices SSM2017 had a noise spec of 0.95nV per root Hz at 1kHz. The TI part is listed as 1.3nV per root Hz at 1kHz, so it's not quite equivalent.*

*We will build yet another version of the 401 and compare performance. However, it may be a moot point. A professional recording studio has been evaluating the 401 for about seven months and the owner reports "excellent" performance. He is writing a review, which I hope to see in the not-too-distant future.*

## NOSTALGIA

The two volumes of *Audiocraft* reprints have given me more pleasure than anything that I have read for quite a while. Some of my enjoyment is simply nostalgia—for companies (e.g., Heathkit) long since vanished and for the enterprise of those who adapted an open reel tape deck to their car. But of more lasting value are Norman Crowhurst's columns on design fundamentals that are as relevant today as when they were written. I hope that you are able to reprint the 1958 issues.

Radley M. Smith
wesmiths@comcast.net

*Thanks for your kind words. The 3rd series is in production.–Ed.*

## HELP WANTED

I am trying to locate a replacement laser for my Philips CD 650 CD player. I have tried the obvious places like Philips but to no avail.

I rebuilt this machine from an article from the *Audio Amateur* in the late 80s. The laser transport number is cdm2 part number 4822 691 30211.

Please if you can help I would appreciate it very much.

John Weegenaar
weegenaar@xtra.co.nz

A group of audiophiles I know are quite enthusiastic about near-field listening, because it focuses so many audible felicities!

The only article I've read about this was written by the late Peter Mitchell about 20 years ago. It was so potently convincing that I still prefer this listening mode.

The title of the article was, approximately, "Champagne Sound on a Beer Budget." Does anyone have a copy of this article to send me? Thank you! ❖

Carlos E. Bauzá
PO Box 810
Guayama, PR 00785-0810
bauzace50@yahoo.com

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

# Classifieds

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## AD INDEX

# Yard Sale

## FOR SALE

Four 10″ 831727 (8Ω) CC Line Peerless woofers, 20Hz–500Hz, $35 ea; Four 5¼″ 55–1580 MCM midbass, fiberglass cone, phase plug, 50Hz–5kHz, $20 ea; Two 5¼″ 850488 Peerless midbass, HDS series (no phase plug), 60Hz–3kHz, $30 ea; Two ribbon tweeter Hi-Vi Research RT1c, extend above 20kHz, $35 ea; Four Radio Shack/Lineaum tweeter ("360° wide angle") $15 ea., or $50 for all 4. Some never used, others barely used and like new. Please consider 10% additional to cover UPS ground shipping costs. Bank check (official or cashier's check) only please, no personal checks. For detailed specs, go to www.madisound.com, www.mcmelectronics.com, www.partsexpress.com, www.meniscusaudio.com, or other supplier/distributor websites. Dan Martin, 336-274-7809, dtmajmemm@earthlink.net.

Two Plitron power transformers 625VA 35V/35V 8.93A, never been used, $60 each. Two Plitron output transformers PAT-008, never been used, $170 for pair. ashby123@attbi.com.

Sprague 0.27μF 400V (8pcs) $48; Astron 0.25μF 400V (8pcs) $48; Dearborn 0.15μF 200V (8pcs) $24; Westcap 0.1μF 200V (8pcs) $24; Aerovox 0.2μF 200V (8pcs) $24; Vitamin Q 0.039μF 200V (8pcs) $24; NOS, mil-spec, hermetically sealed, axial, signal capacitor. Matched within 1% of each other: free shipping, 201-780-5132.

## WANTED

*Speaker Builder* 1987 and 1989 back issues, have available 1981 through 1984, and 1986 issues. Contact degu99@netzero.net.

Schematic of Phase Linear preamp—Model 2000 Series Two Stereo Console. Audio output transformers for use in Electrostatic Speakers. High powered amplifiers. Carver M400 Magnetic Field Power Amplifier. Angel Rivera, (941) 798-9227 (Florida), e-mail: alrivera@tampabay.rr.com.

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# Audio Aid

## A Balanced Op-Amp Driver

Using half of the OP275 as an inverter with a gain of 1× will result in a perfectly balanced signal to drive the 6SN7 double triode, which is used as a driver for an output stage (*Fig. 1*). This driver outputs a wideband signal of about 100V p-p with a 3V AC signal on points C and D, which should do for most applications. The maximum undistorted output voltage depends a bit, of course, on the plate and supply voltage on the driver.

This circuit works extremely well, and you should not use any additional feedback, which will absolutely kill its musicality. The opamp will need a small ±15V DC supply, but it eliminates a phase inverter with the DC plate and filament supplies. If you fancy a regular volume at the input, replace the potentiometer with a 4k7 resistor and omit the 1k2. You will still enjoy the music.

A. J. van Doorn
The Netherlands

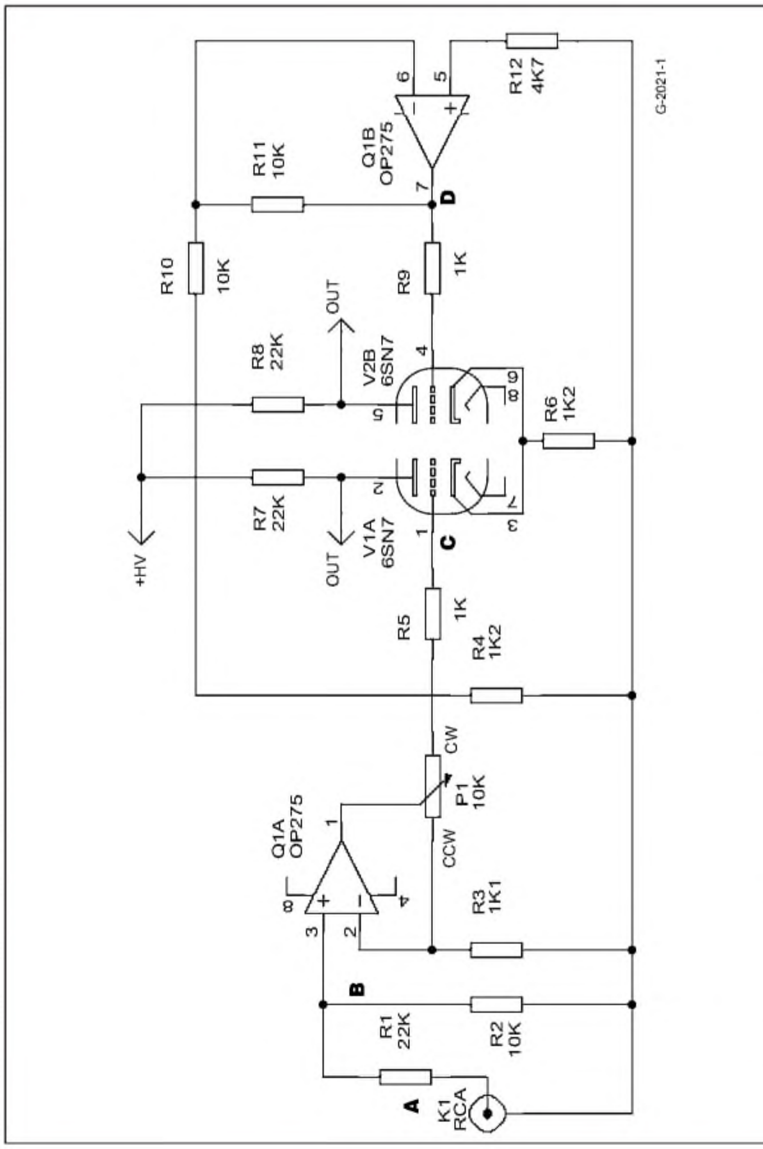


FIGURE 1: Balanced driver with op-amp input and inverter.