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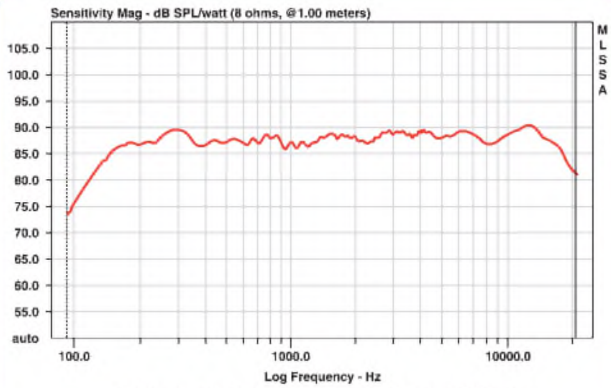
RD-40	264-706	40"
RD-50	264-702	50"
RD-75	264-700	75"



Representative Frequency Response Graph of RD Series

Carver/Genesis Replacement Element

This unique planar element was specifically designed as a replacement for the Carver planar element. The 48" ribbon was used in the Carver Amazing loudspeakers as well as in several Genesis loudspeaker models. Instructions and necessary hardware are included to make direct drop-in replacement possible.
Mfg. Part # RD48
PE Part # 264-704

Neo Series

The Neo series planar transducers feature Bohlender Graebener's latest technology in a compact and inexpensive package. The Neo series drivers differ from conventional dome tweeters in that there is virtually nothing between the electrical signal and the sound- just an almost weightless diaphragm. No wonder the Neo series drivers deliver clean, airy, transparent sound that is inherently natural and musically pleasing. The Neo technology is available in a 3" model in raw dipole form, a 3" model with faceplate, and an 8" dipole unit. All Neo models have a 4 ohm impedance and a sensitivity of up to 96dB at 2.83V/1m.


Neo3


Neo3FP


Neo8

PE Part #	Mfg #	Power Handling	Frequency Response	Dimensions (L x W x D)
264-718	Neo3	20W	2,000-20,000	3-1/2" x 2-5/8" x 1/2"
264-720	Neo3FP	20W	2,000-20,000	5-1/8" diameter x 3/4" deep
264-712	Neo8	50W	200-20,000	7-7/8" x 3-1/2" x 1/2"

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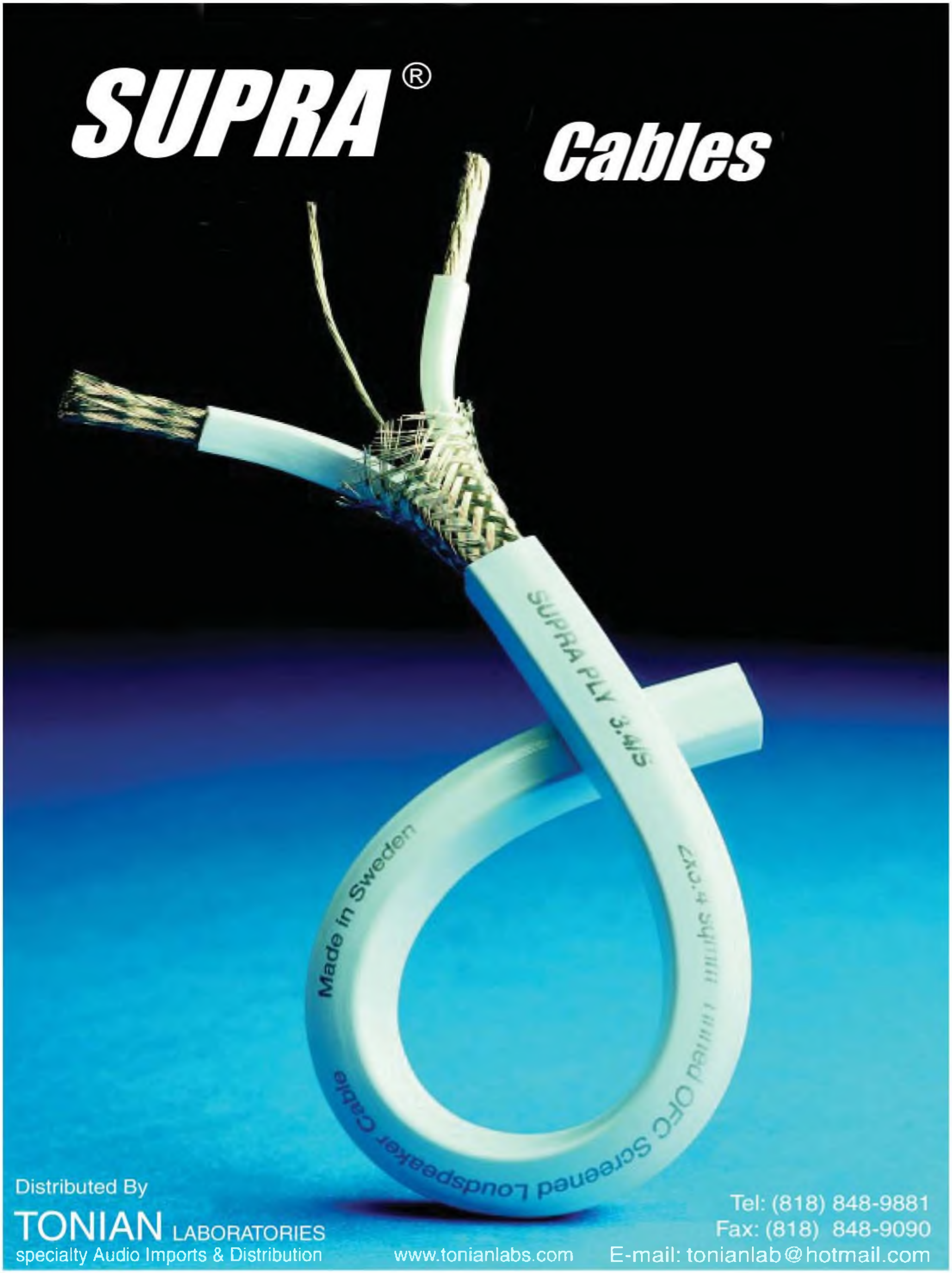
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Audio News

■ HENRY KLOSS, 72, AUDIO INNOVATOR

Henry Kloss, a well-known and admired audiophile and inventor, died at the age of 72 in Cambridge, Mass. on January 31. Mr. Kloss was born in Altoona, Penn., was educated at M.I.T., and began his rise to prominence at Acoustic Research. He was one of the founders of KLH, and also founded Advent and Cambridge Soundworks. Kloss co-invented the AR-1 speaker, built the Model 8 FM radio, and more, spawning many innovative ideas in audio, and even dabbled in video. His first priority was quality, not profit, as his products' reputation for producing quality sound, nicknamed "Boston Sound," bears out. He is survived by one son, two daughters, and seven grandchildren.

■ CARA 2.1 *PLUS*

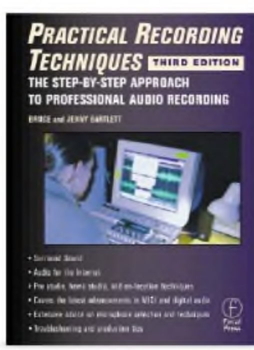
Rhintek, Inc. announces the availability of CARA 2.1 *PLUS* to the American and Canadian markets. CARA (Computer Aided Room Acoustics) is an MS Windows room acoustic modeling software package used to accurately determine the properties of sound waves in custom room designs. CARA 2.1 *PLUS* can define symmetry constraint options and calculate simple rooms 1000 times faster than before. There is also a media player to compare "before" and "after" wave files so you may hear how room elements affect the acoustics. For more, contact Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail: sales@audioXpress.com, website: www.audioXpress.com.

■ POWER SERIES SUBWOOFERS

JBL announces the release of the Power Series subwoofers. Using the same frame, cone, and surround as their flagship GTi subs, they are designed to make the best use of an amplifier's current capability. These subs use single 2Ω and dual 2Ω voice coils to extract the most power possible from amplifiers designed to provide high current into low impedance loads, such as the growing category of dedicated mono subwoofer amplifiers. The Power Series are designed for use in small sealed and vented enclosures and are available in 10″ and 12″ single voice coil and dual voice coil models. For more, contact JBL Consumer Products, 250 Crossways Park Drive, Woodbury, NY 11797, 516-496-3400.

■ *PRACTICAL RECORDING TECHNIQUES*

Focal Press announces publication of *Practical Recording Techniques* (Third Edition), by Bruce Bartlett and Jenny Bartlett. This is a hands-on practical guide for beginning and intermediate recording engineers, producers, musicians, and audio enthusiasts. It covers the latest techniques for professional studio, home studio, and on-location recording and gives advice on equipping a home studio, from low-budget to advanced. *Practical Recording Techniques* explains how to judge recordings and use the equipment available to improve them, and includes special situations and topics, such as documenting the session, recording spoken word, troubleshooting bad sound, and more. For more, contact Focal Press (an imprint of Butterworth-Heinemann), 225 Wildwood Avenue, Woburn, MA 01801-2041, 781-904-2500, FAX 781-904-2640.



■ AMPZILLA 2000

James Bongiorno, designer of the original Ampzilla, releases a new Ampzilla 2000 200W monoblock. The circuit is a full complementary four-quadrant balanced bridge weighing 52 lbs. For more, contact Spread-Spectrum Tech., Inc., phone/FAX 805-740-9902, e-mail sstinc@ earthlink.net or jamesbongiorno@ampzilla2000.com, or visit the website at www.ampzilla2000.com.

■ CHANNEL ISLANDS AUDIO

Channel Islands Audio introduces its first of many affordable, factory-direct, hi-end audio products, the VDA-1 24-Bit/96k Digital/Analog Converter and VPC-1 Passive Volume Control. For more, contact Channel Islands Audio at http://www.ciaudio.com.

■ PLITRON NBT

Plitron Manufacturing Inc. introduces a new clean-power transformer technology, named NBT or "Narrow Bandwidth Technology." These toroidal power transformers substantially attenuate differential and common-mode noises, eliminating requirement of external circuits or components. They also provide harmonic cancellation on the power line beyond any specified frequency. For more, contact Plitron Manufacturing, Inc., #8 601 Magnetic Drive, Toronto, Ontario, Canada, M3J 3J2, 416-667-9914, FAX 416-667-8928, e-mail sales@plitron.com or techinfo@plitron.com, or visit www.plitron.com.

■ 6B-ST

Bryston announces the latest in their line of high-end amplifiers, the 6B-ST three-channel power amplifier. Comprised of three completely independent modules, the 6B-ST was designed for installations requiring three channels of audio conveniently packed into a single chassis. Each of the three modular channels is capable of delivering 250W into 8Ω or 400W into 4Ω, and each module functions independently, with its own electronic circuitry, connections, and independent power supply. For more, contact Bryston LTD., 677 Neal Drive, Peterborough, Ontario K9J 7Y4, CANADA. 4, website: www.bkprecision.com.

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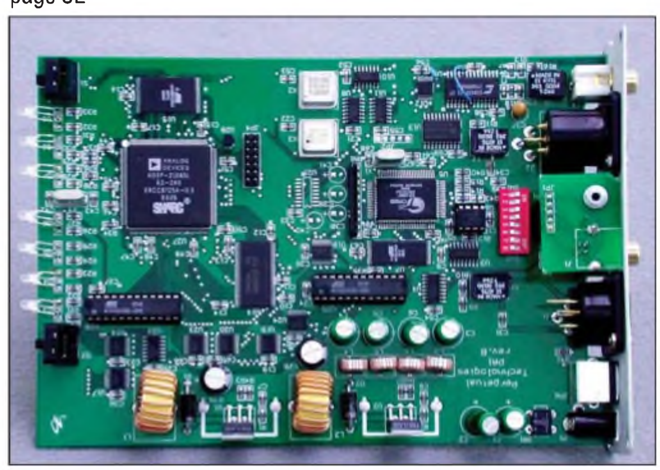

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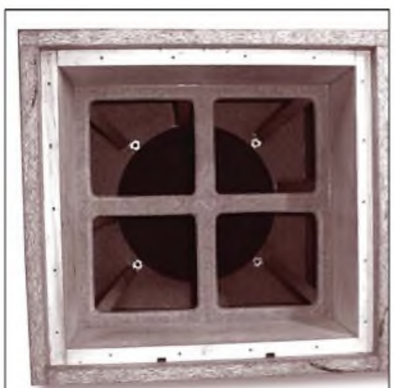

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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A New Class-B Amplifier

This tube veteran presents some groundbreaking work in amp design. **By Richard Modafferi**

Carefully designed tube circuits offer simplicity and reliability, so good tube equipment lasts a long time. For example, antique radios with tubes 60 years old work each day in my home; a vintage 1928 Philco model 87 Neutrodyne in the dining room has tubes someone installed in 1934; my grandmother's 1937 Philco model 660, now in my home, still works with its original tubes and electrolytic capacitors.

Around 1984 I began to consider the conversion of my solid-state audio system to tube electronics. This project began in 1985 with the design of a power amplifier (finished in 1987), followed by a preamp (1989) and a tuner (1991). A crucial concept in this project was the idea of simplicity—all circuitry must utilize the dumbest possible technology and have the fewest possible components. In addition, the system (especially the power amplifier) must be "bulletproof." By this I mean that the amplifier had to suffer from the user's mistakes and not blow up—open, short, or strangely reactive loads should leave the amplifier undamaged.

I brought to this project 40 years of experience in high-reliability tube circuit design, first at a company that manufactured cable TV electronics and later at McIntosh Laboratory, where reliability was (and still is) a legend. First, a discussion of a new tube audio power amplifier circuit that I have developed with the idea of maximum reliability. I modestly call it "The Modafferi Triode."

ABOUT THE AUTHOR

Richard Modafferi received his B.S.E.E. from Manhattan College in 1960 and his M.S.E.E. in 1965 and M.S. Computer Science in 1968 from New Jersey Institute of Technology. Richard worked for Blonder-Tongue Laboratories from 1960 to 1966 and McIntosh Laboratory from 1968 to 1974. He has been an independent inventor-consultant since 1974. He is the inventor of the "Infinite Slope" loudspeaker, which is currently in production by Joseph Audio. Hobbies include vigorous exercise—running, X-C skiing, bicycling—vegetarian cooking, and playing the piano.

PHOTO 1: The original Modafferi triode amplifier. ***(Photos courtesy of audioclassics.com.)***

PHOTO 2: Interior view.

THE CONCEPT

Although my circuit contains new ideas, it is based on a concept developed by someone else many years ago. In July 1936, a fellow named Charles Stromeyer published a paper entitled "General Theory and Application of Dynamic Coupling in Power Tube Design."[1] In this paper Mr. Stromeyer describes his "dynamic coupling" and discusses special tubes developed for optimal performance in this circuit.

Sadly, it never achieved the popularity it deserved, but I own what might be the only radio ever manufactured that uses a push-pull version of Mr. Stromeyer's circuit—a 1941 Zenith model 12A1 console. Dynamic coupling is described in vintage RCA tube manuals, along with the special triodes developed for its driver and output stages.

My curiosity aroused, I began to imagine a modern version of Mr. Stromeyer's dynamic coupling. In particular, I wanted to try a Class-B version; all of the early dynamic-coupling circuits had been single-ended Class A or push-pull A or AB. Class-B operation in a dynamic-coupled circuit requires that the driver stage provide the large grid current demand of the output stage. This was not practical in Mr. Stromeyer's time, but I found it easy to do with the application of modern technology.

THE CIRCUIT

Class-B dynamic-coupled output circuits work best if the output tube is a

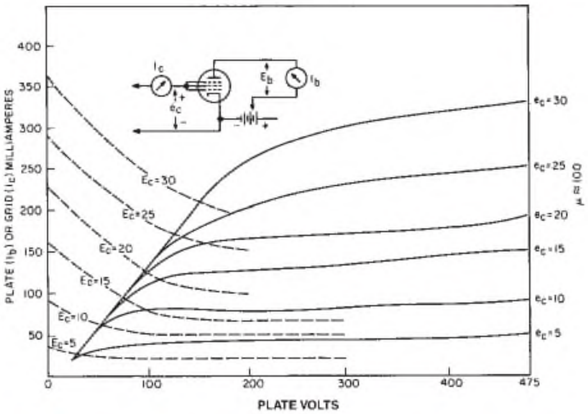


FIGURE 1: Average plate characteristics enhancement-mode triode.

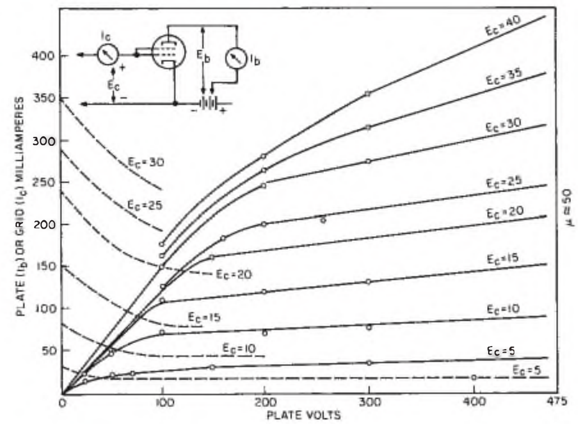


FIGURE 2: Average plate characteristics Class-B enhancement-mode operation.

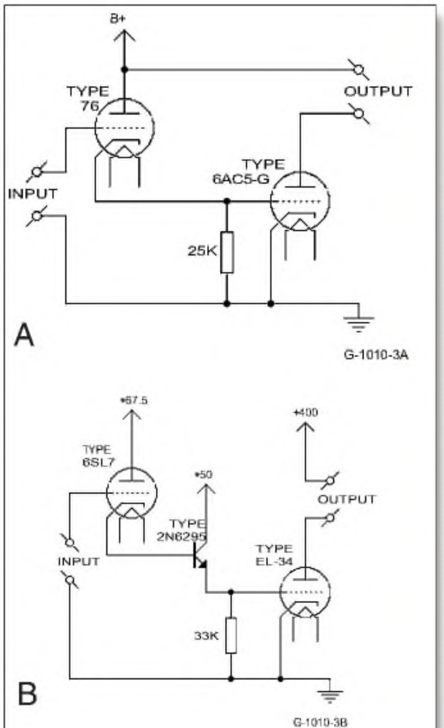


FIGURE 3: Dynamic-coupled circuits. (a) original by Stromeyer, from RCA Tube Manual, 1940 edition, page 90, and (b) Modafferi version.

hi-μ triode having large cathode-current capability. One modern pentode, the EL-34, meets this requirement fairly well. Another modern tube, the 6550 beam tetrode, works but not optimally.

Figures 1 and *2* show, respectively, the plate characteristics for the EL-34 and 6550 when these tubes are operated as enhancement-mode hi-μ triodes. Observe that all the grids are tied together—i.e., pins 1, 4, and 5 for the EL-34 and 4, 5 for the 6550. Note that the grid voltage is positive, not negative, as in conventional circuits. Thus the tube draws grid current, and for some tubes, this current can be large. The higher the μ, the lower the grid current.

The EL-34 has a μ of 100 while the 6550 has a μ of 50; these values are determined from inspection of the plate characteristic curves (*Figs. 1* and *2*). Amplification factor is defined as:

Amplification factor μ = change in
Eb/change in Ec
Ib = constant

Note the almost pentode-like curves for the EL-34 (*Fig. 2*). Observe that the grid current is large, especially in the region to the left of the saturation region of the plate-current curves. (This situation is analogous to the large rise in screen current in conventional pentode operation when plate voltage is removed.) The somewhat poor saturation characteristic of EL-34 or 6550 as enhancement-mode triodes limits the maximum power and efficiency which can be obtained. The large grid current makes the enhancement-mode triode difficult to drive and accounts for the non-use of this mode of operation, at least until I developed a simple and practical drive circuit.

The Modafferi triode, while not an optimum circuit with respect to available power and distortion performance from a given set of output tubes, offers advantages with respect to simplicity and reliability. Since no grid bias is required, there is no grid bias power supply and no grid bias adjustments. Output tubes draw very small no-signal plate currents, resulting in low plate dissipation, high efficiency, and long tube life while obtaining about 60 to 70% of the power you would obtain from an optimally designed Class AB pentode circuit using the same tubes and B+ voltage.

My new drive circuit (*Fig. 3*) is based on the earlier work of Mr. Stromeyer and is, in effect, a Class B realization of the original dynamic-coupled circuit. The new concept here is the use of a high-μ triode and Darlington transistor combination to drive each phase of a push-pull output circuit. One high-μ triode-Darlington transistor combination drives each side of the push-pull pair; i.e., one tube for a push-pull pair, two tubes for push-pull parallel, three for six output tubes, and so on. This drive circuit easily supplies the required large grid current while having low nonlinear distortion.

THE AMPLIFIER

The first Modafferi triode amplifier is shown schematically in *Fig. 4*. Four more have been built after this opus 1, having power outputs ranging from 30W to 120W per channel. My original, shown here and in *Photos 1* and *2*, is rated at 50W/channel and uses a quad of EL-34 output tubes and a B+ of 400V.

All five amplifiers share the same input and drive circuitry, with each having different output stage configurations. The smallest (30W) uses a single pair of EL-34 output tubes and 400V B+. The largest (120W) uses four 6550 output tubes and a B+ of 600V.

The input circuit might look strange, which indeed it should, since it is an invention of mine (U.S. patent 4,918,394). This represents an attempt at obtaining a tweak-free gain/phase inversion stage which would work without the need for adjustment, have good gain, noise and distortion characteristics, and be as insensitive to tube aging as possible.

The novel feature of this new input circuit is the use of DC feedback (*Fig. 6*) taken from the cathode of triode V3 to the grid of triode V2. This maintains DC stability on both stages: (1) cascode voltage amplifier stage V1-V2 and (2) phase inverter stage V4 as tubes age. The operation of this circuit isn't discussed in detail here[2], but will be treated further in a future preamp article in which I use this same circuit for the phono and line gain stages—it's even used in my new tube MR-78 FM tuner!

Examination of the amplifier circuit in *Fig. 4* illustrates how I make dumb technology work. Aside from the original input circuit just mentioned, and the new output configuration, there is nothing unusual. Both aforementioned circuits, while novel, are simple and straightforward. The amplifier's power supplies use basic rectification and low-pass filtering. Only a single regulator is used in the B-supply to the driver transistors.

DESIGN DETAILS

I should discuss items that would aid anyone trying to duplicate my amplifier, or to build a similar one. Amplifiers opus no. 2 (*Fig. 5*) and 3 were built on old Audio Research chassis—service basket cases I obtained in my restoration work for Audio Classics. Rather than junk these units, I convinced the owners to let me try my new circuit in them. I gutted the chassis—the power and output transformers were good—and rebuilt them to the circuit of *Fig. 4* using all new parts.

One of these was a D-76, the other a D-160. Both are excellent platforms on which to begin work and offer an easy means for the home constructor to approach a large amplifier project. Thus I would encourage anyone contemplating the construction of my tube amplifier to first look for junk units having good transformers; these are often obtainable at audio club swap meets or hamfests, or from magazine and newspaper want-ads.

From my McIntosh days when all amps must have meters, I used power meters in opus no. 1. The Class-B output circuit makes it easy to install an "average reading" meter, as the output tubes operate true Class-B with only a small idling current of about 7mA. Opus 1 uses a 1Ω resistor to ground at the center tap of the output transformer cathode winding. The sum of all the output tube cathode currents flows in this resistor.

Since the output power of the amplifier is directly proportional to the plate current, a meter connected across this resistor will read output "power." I use quotes because what is actually being metered is output tube plate current—if no load is connected the meter will not read; if a shorted load is connected the meter will read lots of power (line fuse will blow). True power is delivered, however, to a matched load, and only under this condition can the meter be calibrated to read power.

Mine is simply a 250µA DC meter having a dB scale which I calibrated to read 0dB at 50W into a matched load. Note the use of a "meter zero" pot, which is used to adjust for the small DC offset voltage from the output stage idling current.

FEEDBACK CIRCUIT DESIGN

Feedback networks need special attention. All five opuses use similar feedback networks having one main path from an output transformer secondary load winding. Additional feedback may be taken from another secondary winding if there is more than one (*Fig. 4*), or from the primary winding if the output transformer has only one secondary. The main feedback circuit is empirically adjusted to provide not more than 15dB of feedback and no visible ringing on a 1kHz square wave into a resistive load.

Feedback is adjusted for unconditional stability—square-wave response into a high-Q pure capacitor load should be clean with minimal overshoot. There exists a value of load capacitor which will yield maximum instability; the value is remarkably constant over all transformers—usually around 220nF. The value of the capacitors used in each of the two feedback

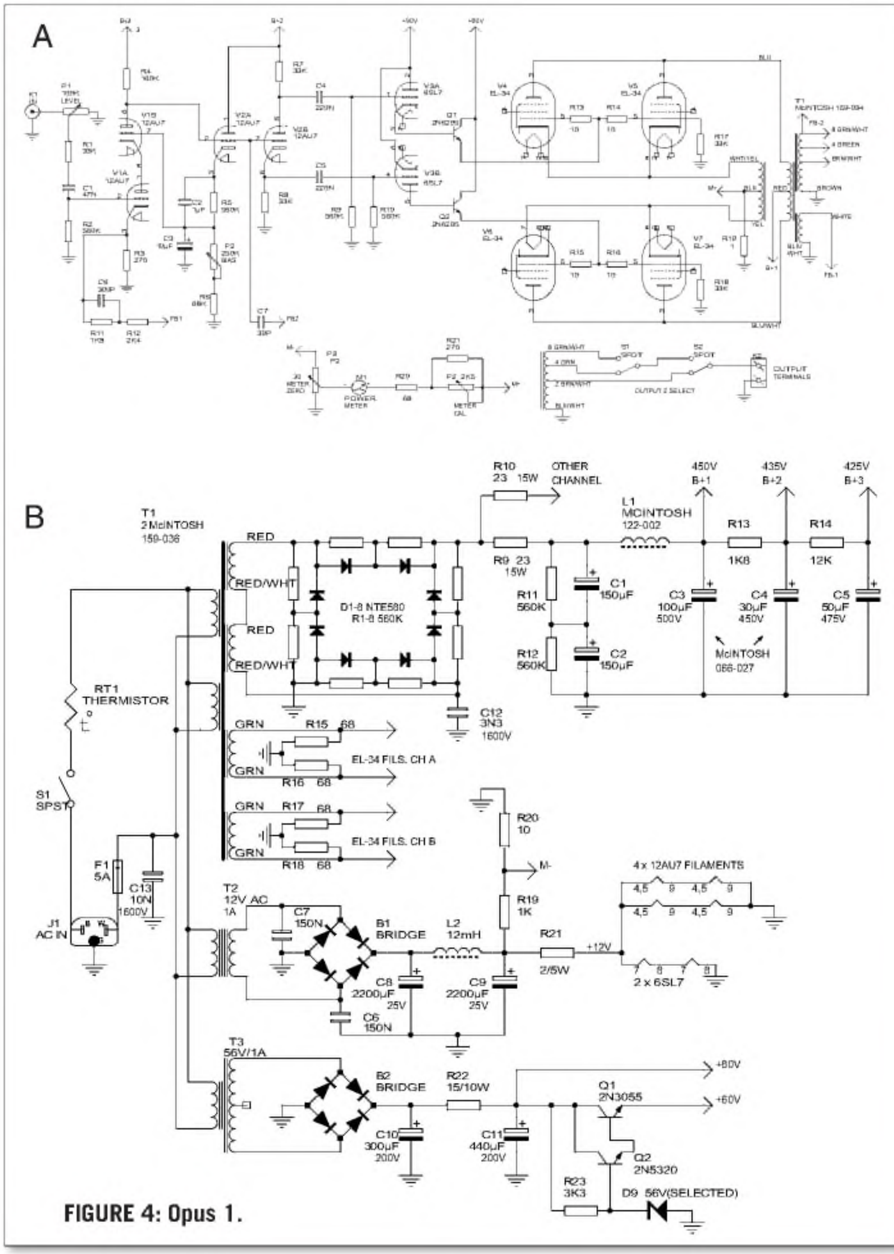


FIGURE 4: Opus 1.

paths is empirically adjusted to minimize ringing on pure capacitive loads.

Feedback is adjusted by first omitting the single capacitor (39pF in *Fig. 4*) in the "tweak" feedback loop and adjusting the capacitor (300pF) in the main feedback loop for flat frequency response, or no peaking at high frequencies, with the amplifier connected to a resistive load. Then connect a pure capacitive load and adjust the "tweak" feedback loop capacitor for minimum ringing while maintaining flat high-frequency response.

If stability cannot be obtained, reduce feedback by increasing the value of the resistors in the main feedback loop and then begin again. Note the use of two series-connected resistors with the loop compensation capacitor connected across just one resistor. This is old standard practice and is sometimes called "shelf" compensation from the shape of the feedback network frequency response.

Danger exists when a single RC parallel combination is used in a feedback network, as the feedback approaches unity at very high frequencies due to the decreasing reactance of the feedback capacitor. This isn't a problem if the output transformer rolls off at high frequencies with a minimum of spurious resonances (i.e., only one dominant pole at high frequencies), but including shelf compensation is always good insurance.

A consideration overlooked by almost everyone is immunity to RF interference. I live 600′ from a 5kW AM broadcast station; without shelf compensation, the long speaker leads act as antenna wires and couple RF energy through the feedback capacitor into the amplifier's input stage. Here the signal is rectified and causes unwanted bias shifts and sometimes audible sound output from the AM station. The unbypassed feedback resistor (2k4 in *Fig. 4*) in combination with stray capacitances forms an effective RF filter which eliminates this interference.

OUTPUT TRANSFORMER CONFIGURATIONS

Cathode coupling to the load is utilized in the output circuit of *Fig. 4*; this is standard practice by McIntosh and some other companies. It isn't necessary for the function of the Modafferi triode circuit. Opus 1, which uses McIntosh output transformers, has cathode coupling. Opus nos. 2 and 3 have Audio Research output transformers configured using cathode coupling in the original circuit, but in the new configuration they are connected conventionally. Opus 4 and 5 use Dynaco output transformers in a conventional output configuration. *Figure 6* illustrates output transformer connections and feedback networks in opus 2—built on an Audio Research D-160 chassis—and opus 5—built on a Dynaco stereo 70 chassis.

PRACTICAL CONSIDERATIONS

You will note that I have avoided describing an exact procedure for duplicating amplifier opus 1. This is because it would be unlikely that you would attempt an exact copy of it—most obviously because the chassis and cabinet would be difficult to make. Opus 1 has a scrap McIntosh MC-3500 chassis and a wood cabinet—the path I took to easily fabricate it by using material I had available.

Others will have different material available, and I encourage anyone contemplating construction of the circuit described here to exercise some imagination rather than copy. A good starting point, stated earlier, is to begin using a junk unit having good transformers. I cobbled the Dynaco stereo 70 to the Modafferi triode circuit in just one day! Since my circuit is fairly simple, it is easily adaptable to many situations.

I warn you not to destroy a valuable unit to build my circuit. Do not convert a Marantz 8 or 9, any McIntosh, or other classic of intrinsic merit—if in doubt, seek advice first. You might consider starting from a junk classic—only if it is so bad that restoration is not feasible.

Scratch-building opus 1 by experienced audio amateurs is possible, and to this end, I give the McIntosh part numbers of the power and output transformers in schematic *Fig. 4B*. A limited number of these are still available as NOS parts from Audio Classics. You can also obtain suitable power and output transformers from companies advertising in this publication and in others.

PERFORMANCE MEASUREMENTS AND COMMENTS

Power output, frequency response, and harmonic distortion test results for opus 1 are shown in *Figs. 7*, *8*, and *9*, respectively. Square-wave responses into resistor and capacitor loads are shown in *Fig. 10*. A waterfall plot is shown in *Fig. 11*. This test is always done on loudspeakers, but am I the only one to test electronics this way? I regularly do this on FM tuners because it clearly shows faults such as spurious output from poor or malfunctioning stereo decoders. Amplifiers seldom show bad waterfall plots, but if a problem such as instability exists, it will be clearly evident.

There are no remarkable performance characteristics for opus 1. Its measurements could be easily exceeded by almost any transistor amplifier. In fact, I recently built a simple 50WPC stereo transistor unit in order to test loudspeakers and it bests the tube amp in all measurement respects (frequency response, distortion, damping factor, rise time, and anything else you can think of to test). Opus 1 may be an ordinary tube amplifier regarding performance, but it has high intrinsic reliability, stability, and long-expected tube life. These three characteristics are often missing in tube amplifier designs.

In this age of good cheap and long-lasting solid-state electronics, does the existence of overcomplex, expensive, tweaky tube electronics make sense? Would you put a precious set of NOS Mullard EL-34 tubes in an amp that would burn up these gems in one or two thousand hours? Do you shudder every time you fire up a tube system as you wait for sparks and fire?

I asked these questions back in 1985 when I began the conversion of my

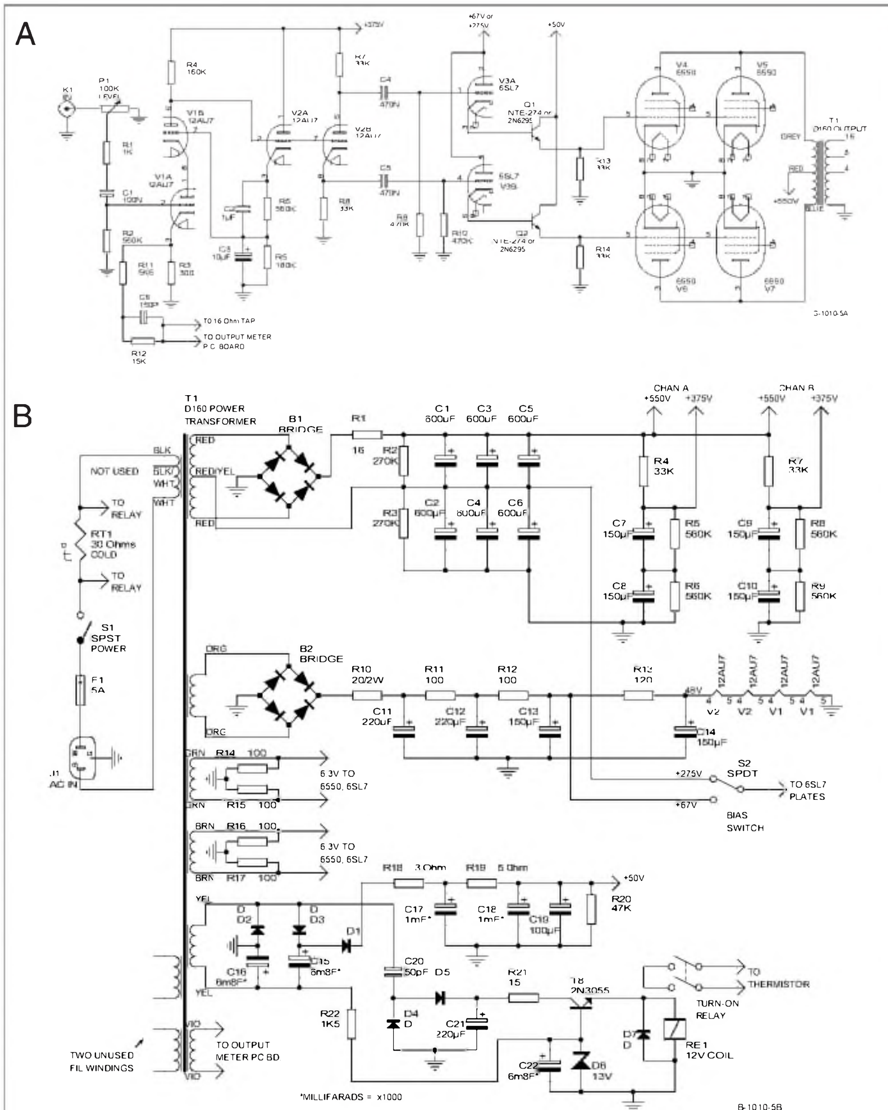


FIGURE 5: Modafferi triode amplifier built on an Audio Research D-160 chassis. One stereo channel shown (other is identical and shares common power supply). Note that each channel has its own B+ filter system.

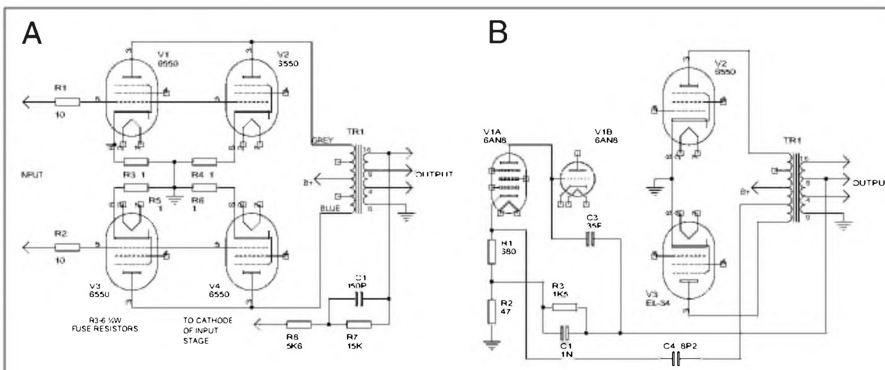


FIGURE 6: Output configurations and feedback networks for Modafferi triode amplifiers. (a) opus 2 on Audio Research D-160 chassis, and (b) opus 5 on Dynaco stereo 70 chassis.

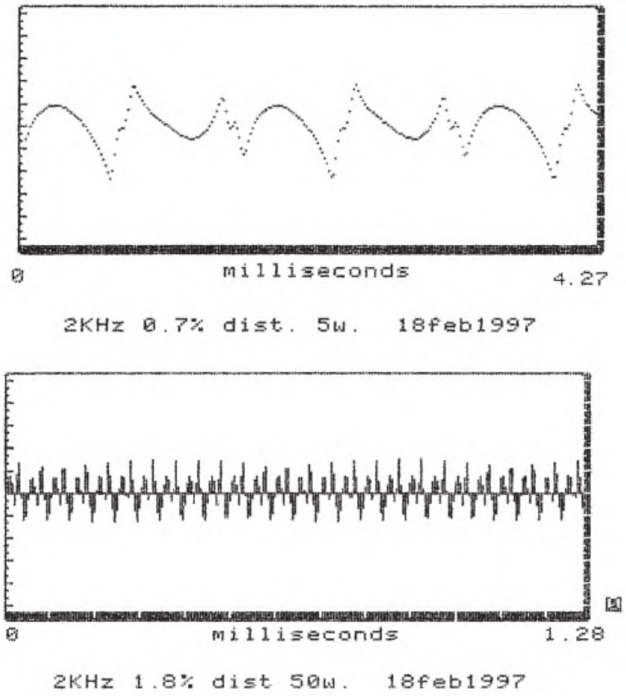


FIGURE 7: Low-power distortion.

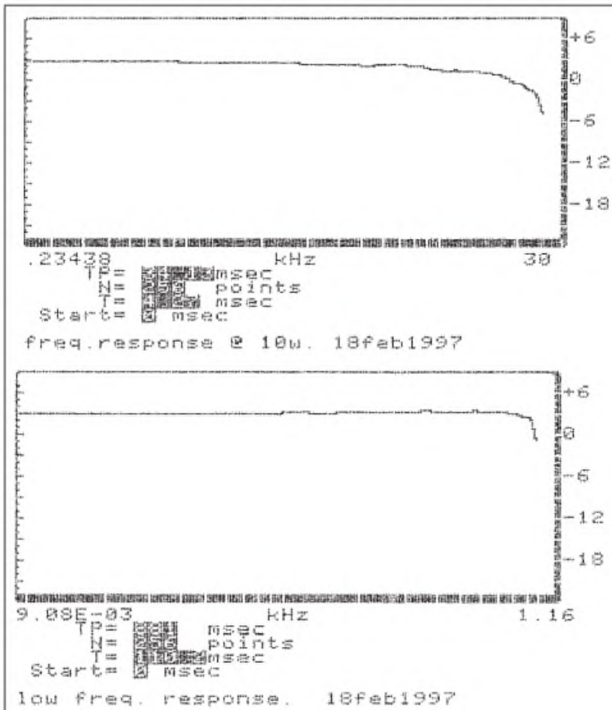


FIGURE 8: Opus 1 8Ω load on 8Ω output tap.

audio system to tubes. If 60-year old radios continue to work flawlessly in my home, then, I reasoned, audio gear, which is built using better components, should last centuries!

Thus the only justification for building a tube audio system is based on two ideas: (1) It must work as well, or better (if possible), than the solid-state system it replaced, and (2) It must work with a vanishingly small probability of failure. With regard to (1), the tube amp opus 1 sounds as good as the solid-state unit it replaced—at least on an easily driven dynamic speaker load. With regard to (2), I expect at least a 50,000 hour tube life; opus 1 should easily last the rest of my life (I'm 64) and beyond the end of my son's life (he's 16).

I base this estimate of tube life on my long experience doing tube life tests. I learned that if you run a tube at low electrode dissipations and small average cathode current, tube life can be very long.[3]

Only time will tell whether I did things right. Opus 1 has about 15,000 hours on it now, and so far no tube aging is evident. The output tubes run so cool, you can hold your hand on them without discomfort. It is immensely gratifying to turn on a tube audio system every day that always works. Perhaps the first element to fail will be the line cord.

There has been one field failure in a Modafferi triode amplifier so far. Opus 3, built in 1994 on an Audio Research D-76 chassis, melted an output tube after a few weeks of service. The 6550 shorted screen to plate, a rare occurrence. Plugging in a new tube and replacing the 1Ω cathode fuse resistor fixed the problem. The 2N6295 driver transistor and 6SL7 driver tube were undamaged, but I replaced both for insurance. Both Audio Research conversions are in daily use and working at the time of this writing.

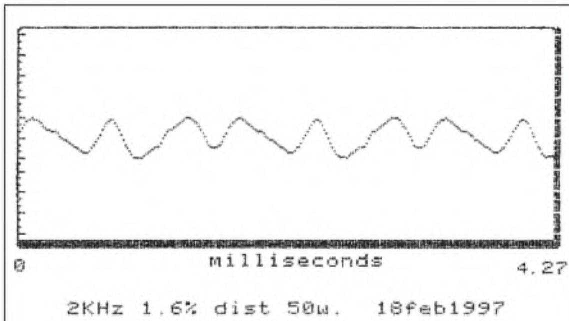


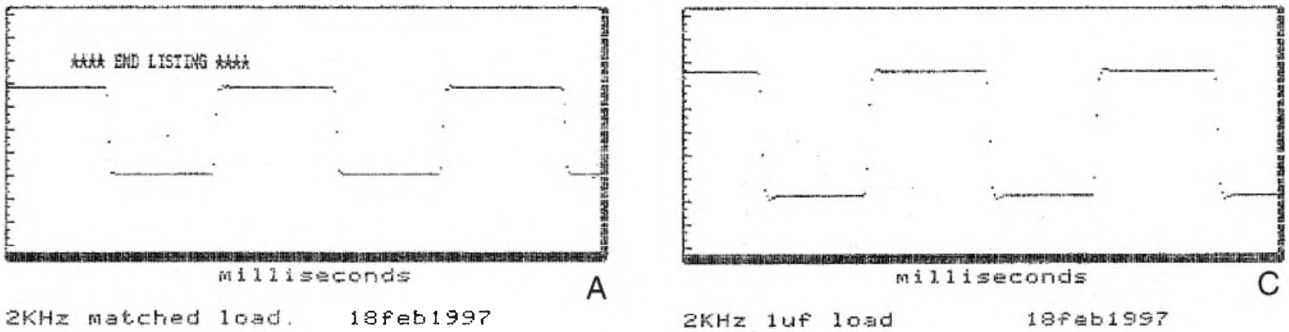
FIGURE 9: High-power distortion.

OUTPUT TRANSFORMERS

"A tube amplifier is only as good as its output transformer," said someone long ago. I discovered this truism once again during testing of opus 1. I was performing a full-power test at 20kHz; power output was 50W and distortion was 6%. Hardly sterling performance.

DC power input to the power output stage was almost 200W! Twenty-five percent efficiency from a Class-B circuit that should yield at least 60% efficiency! Where was the power going? When I noticed the EL-34 plates glowing dull red, I shut things off and pondered the problem.

One idea made sense: The McIntosh output transformer in opus 1 is Mac's smallest unit; it's from an integrated amp and rated at 30W. I was pushing it to nearly double its power rating. Investigation of the output stage tube current and voltage revealed an elliptical load line (*Fig. 12* and "Output Stage Tests" sidebar). Without going into complex theory, let me state that unless the load line (the locus of the output tube plate current and plate-cathode voltage under signal conditions) is straight and angled so as to yield optimum power output and efficiency (*Fig. 11*) things won't work optimally. There could even be a meltdown!



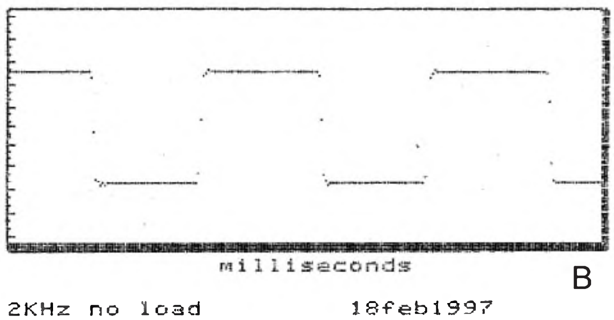


FIGURE 10: Opus 1. Square-wave response at 10W. Feedback switch set for 9dB feedback. 8Ω load in (a).

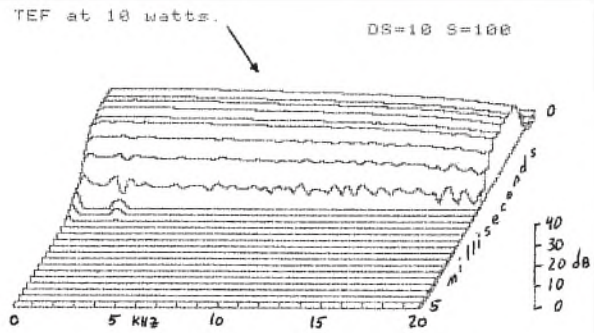


FIGURE 11: Opus 1 TEF. 8Ω load on 8Ω output tap.

OUTPUT STAGE TESTS

I use a simple but powerful test that yields useful information on output stage performance. Oscilloscope connections to the circuit are shown in *Fig. 12(f)*. The resistor R samples the output stage plate current. Output stage voltage is taken from a transformer secondary winding.

An ideal Class A push-pull output stage yields the straight line shown at (a). Since the total plate current is unvarying, a horizontal line appears since the net vertical input is zero. The Class AB waveform is shown at (b). The horizontal portion at the bottom center represents the "Class A" region where both tubes are conducting.

As the drive signal swings into the "Class B" region, one tube approaches cutoff and the other conducts plate current proportional to the input signal. Each side of the push-pull output stage produces an upward sloping line on the scope. One side slopes up to right, the other to the left.

Ideal Class B operation yields the waveform at (c). The bottom of the "V-wave" is the point at which the input drive signal swings through zero where neither side of the push-pull output stage is conducting. As the input signal swings through a complete cycle, each half of the push-pull output stage conducts on alternate half cycles, producing the two halves of the "V" shaped waveform.

Opus 1 produces the waveform at (d) when driven to 50W at 2kHz. It's close to the ideal shown at (c) and is very satisfying. DC input power to the output stage is 85W and AC output power is 50W. Efficiency is a good 59%. Distortion is a fair 1.6%.

The situation at 20kHz is a different matter. Here DC input to the output stage is 200W for 50W output and distortion is now 6%. The tube plates glow dull red. What's wrong? The clue lies in the strange waveform at (e).

With output tube plate-cathode voltage and plate current in phase, the test setup of (f) yields scope patterns in the form of lines. No area is enclosed. When tube current

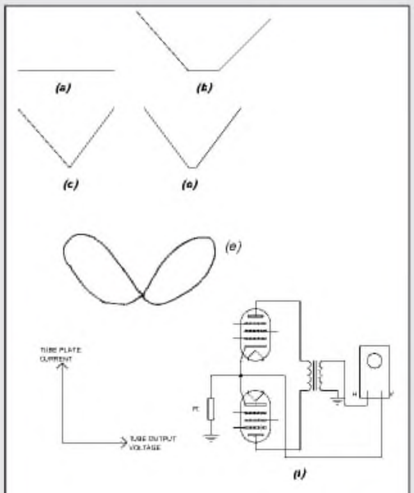


FIGURE 12: Output stage test waveforms. (a) ideal Class A, (b) ideal Class AB, (c) ideal Class B, (d) Opus 1 at 50W and 2kHz, (e) Opus 1 at 50W at 20kHz, (f) scope connections to output stage.

I believe the cause of the problem in opus 1 (besides overloading the transformer) is mainly transformer core loss. This loss has two separate modes: (1) hysteresis and (2) eddy currents. Hysteresis is minimized by using "good" iron in the core, and eddy currents are minimized by making the core laminations as thin as possible and coating them with very thin insulation.

As frequency increases, these losses become predominant. Clever transformer designers can minimize these losses but not eliminate them. Opus 1's transformer core soaks up considerable power at high frequencies—even during the short duration of this test, it warmed noticeably.

The situation is better at low and mid frequencies. Output stage efficiency is about 60% at 1kHz, a reasonable result. I note here that large McIntosh amplifiers have plate efficiencies around 70%, a remarkable result; the theoretical maximum for a Class-B power stage is 78%.

In a nearly desperate attempt to improve things, I tried a bigger output transformer in opus 1. The original, McIntosh part no. 159-034, became the model for a new transformer, with double the power rating. I had the original manufacturer (Magnetic Windings—note here that even Mac did not make all of its own transformers) copy 159-034 in a bigger 75W size. It worked; opus 1 now delivered 70W at 2kHz instead of 50W, but the problem at 20kHz remained—an elliptical load line, high distortion, and poor efficiency. Defeated, I put 159-034 back into opus 1 (saving the new transformer for another project) and went back to more pressing matters: speaker design and vintage audio equipment restoration.

You could blame my Class-B output configuration for the problem at high frequencies. There is a very easy experiment to check this. Replacing the 6SL7 driver tube with a 6SN7 rebiases the output tubes to Class AB mode. With a 6SN7 driver, the tubes work with 50mA no-signal plate current, a typical value for Class AB operation. Power output was slightly higher—55W instead of 50—and distortion a bit lower at full power (much lower at low power—around 0.15% at 5W), but the problem at 20kHz remained: lots of phase shift, poor efficiency, and high distortion.

In a final experiment, I tried the original output transformer of opus 1 in a conventional Class AB pentode output stage using two 6550 tubes. The problem at 20kHz still existed, but efficiency was better, power higher, and distortion lower. Here, the better efficiency of pentodes over triodes became evident. I could push 75W (at 3% distortion) through the 30W transformer at 20kHz without overdissipating the 6550 output pentodes. I leave further investigations of this problem as a homework assignment for the curious reader.

The poor performance of opus 1 at high frequencies is not a serious problem because I use it in a home music system in which it functions at only a small fraction of its power output capability. Even when driven to near maximum output power on signal peaks, the energy at high frequencies in normal program material is negligible. Long tube life can be expected in this application.

The foregoing discussion emphasizes the need to use care in matching an output transformer to the intended use of a power amplifier. Opus 1 works fine in a home music system, but it would almost certainly fail prematurely in an industrial application (vibration transducer, for example) where continuous full power at high frequencies might be required. As just illustrated, my output tubes overdissipate at full power and high frequencies—50,000 hour tube life would never happen in this case!

CONCLUSION

The Modafferi triode is outperformed by many classic tube amplifiers, includand voltage are not in phase, the pattern develops loops. The picture at (e) illustrates a condition in which the tube voltage and current are about 60° out of phase.

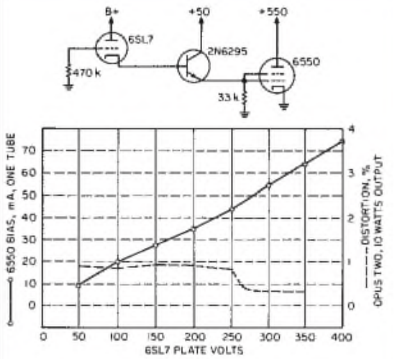


FIGURE 13: Relationship between 6SL7 driver plate voltage, output tube bias current, and distortion.

An output stage functioning under this condition will work poorly, having low efficiency and high distortion. This out-of-phase condition is the problem; the cause of which seems to be mainly due to core loss in the output transformer. Resistive losses in combination with leakage inductance and stray capacitance (high-frequency rolloff) are also contributing factors. The transformer in opus 1 simply runs out of power bandwidth at 20kHz.

MODE OF OPERATION

Class A or AB operation of Modafferi triode amplifiers is possible. The no-signal bias current in the output tube is determined by the m and plate voltage on the driver tube. *Figure 13* shows distortion and bias current for the 6550 output tubes of opus 2.

This amplifier, along with opus 3, uses Audio Research output transformers and works the best of the five amplifiers I have built so far. In Class AB mode, opus 2 achieves a low distortion level of about 0.1% at 10W and a bias current of 50mA per output tube. With 550V on the plate, this is about 28W plate dissipation—hot, but a typical operation for the 6550 in most tube amplifiers.

I provided a switch in opus 2 that puts 67 or 275V on the 6SL7 driver plates. In the 67V position, the output tubes are biased at a low 10mA and operate Class B. With 275V on the 6SL7 the amp runs in Class AB mode, having much lower distortion but running hotter. Opus 2 is the only one of the five in which I installed this "mode" switch.

Moving the switch between the two positions allows you to choose 0.1% or 0.5% distortion at 10W. Full-power distortion (120W) is about 3% in either mode. I noted no audible difference when using the switch and I did not install it in any other opus. The owner of opus 2 uses the amp in Class B mode driving two pairs of KLH electrostatics.

ing those of Marantz and McIntosh. Consider all tube amp circuits: the Class-B unity-coupled McIntosh is probably the best tube topology ever devised with regard to both reliability and performance. The only strong justification for my circuit lies in its simplicity and potential for longest output tube life of all tube amplifier output topologies.

Perhaps the best aspect of the Modafferi triode circuit lies in its "plug and play" aspect. There are no adjustments and nothing to go out of adjustment. The phase inverter is self-balancing and stays in balance regardless of tube aging, and the output stage requires no bias circuitry with its required adjustments. The moderate damping factor and relatively high distortion—with higher order harmonics—inherent in my circuit limit is usefulness. It won't sell as an audiophile product.

The nature of the distortion shown in *Fig. 9* needs to be explained. It looks like classic "crossover" distortion from an overbiased output stage, but in this case the distortion has another cause. When driven by signal, one half of the push-pull output stage is nearly cut off (almost no plate current flows) during that part of the cycle when the other half of the push-pull output stage is conducting maximum plate current. The switching of plate current back and forth between each half of the output transformer primary winding under signal conditions causes "ringing" in the transformer's leakage inductance.

This ringing appears as small but high-order distortion products in the output. The better the transformer, the lower the distortion. The McIntosh transformer in opus 1 and the Dynaco transformers of opus 4 and 5 have moderate leakage inductance and yield fair performance (distortion less than 1% at low power and around 3% at full power). The Audio Research transformers of opus 2 and 3 fared better, yielding distortion lower than 0.5% at low power and about 1% at full power.

Despite its shortcomings, I believe my circuit to be useful. The "Class-B" sound doesn't hurt my 64-year-old ears. The low damping factor and moderate power output limit opus 1's application to fairly efficient speakers having a flat impedance characteristic.

The poor efficiency at high frequencies is not a problem when used to amplify music signals in a home audio system. I cannot discern a sonic difference between opus 1 and a reference Krell KSA-150 using any of my own infinite-slope speakers. But on a Martin-Logan CLS, opus 1 failed miserably, sounding dead and unfocused while the Krell sounded open and clean.

The Modafferi triode's ordinary performance is justifiable if ultimate reliability is the goal. It could find its best application in musical instrument amplifiers where reliability is an important consideration. ❖

REFERENCES

1. "General Theory and Application of Dynamic Coupling in Power Tube Design," Charles F. Stromeyer, Proceedings of the Institute of Radio Engineers, July 1936.

2. U.S. Patent 4,918,394, "Audio Frequency Power Amplifier With Improved Circuit Topology," Richard Modafferi, April 17, 1990.

3. "Reliability Factors in Vacuum-Tube Electronics," Richard Modafferi, *Sensible Sound*, Fall 1992.

Big Mike and the Jimmy, Part 3

With the theory and design considerations behind us, it's now time to tackle construction of this mike preamp design. **By Paul J. Stamler**

I built Big Mike in a straightforward, almost classic tube manner, although not on a classic tube chassis. Instead, I housed the preamps themselves in Sescom's excellent rack-mounting boxes, which are reasonably priced and easy to work with (*Photo 1*). I mounted the bottom plate in reverse, with the anodized side facing in, to gain a bit of insurance against shorts to the chassis.

I also discovered, the hard way, that it's a bad idea to spray-lacquer the entire front or back panel (*Photo 2*). Ground connection to the rest of the box is made through the edges, and if you lacquer (or enamel) them, your shielded box doesn't shield any more. If you plan to spray, mask off the top and bottom ¼″ of each panel first.

I put the raw supply in a mini-box, because it was compact and easy to work, but there's no reason you couldn't put it into another Sescom box. Be careful of ground loops, though, if you mount the supply and preamps in a common metal rack. You may need to use insulating washers and nylon screws.

PHOTO 1: Basic single-channel-dual-output Big Mike preamp (bottom) and prototype two-channel Semi-Zen preamp (top).

Figure 15 shows the internal layout of a single-channel preamp box. I wired the audio circuits point-to-point, using old-fashioned terminal strips, but it might be easier to use terminal boards. I mounted the tubes horizontally in an aluminum panel running across half the box; the folks down at Gateway Electronics, our local parts surplus house, cut the sheet of aluminum to size and bent it on their metal brake. I used rubber grommets on its mounting screws to add mechanical isolation.

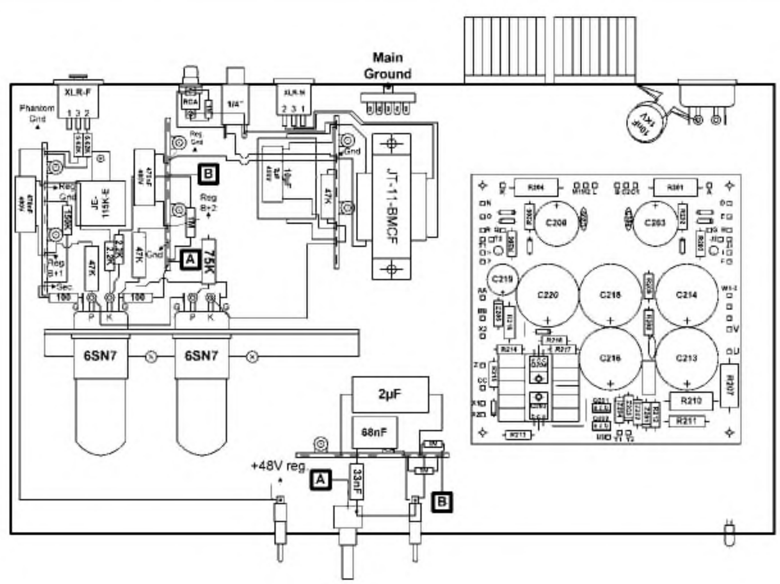


FIGURE 15: Layout for single-channel preamp. The eagle-eyed will note that the regulator board is an earlier version than the one described. For clarity, only a few of the wires are shown.

GOING TO GROUND

The grounding scheme I used is shown in *Fig. 16*. Grounding is one of those subjects that's as much voodoo as science, and everyone's preferences are different, but this worked for me—in fact, it was hum-free from the moment of turn-on.

The most salient feature is that the ground of each regulator does not go to the main grounding point in the chassis, but rather to the ground point on the load, and then to the main grounding point. An article in *Audio Amateur* suggests that this scheme generates the least residual noise, and it has worked like a charm in every project in which I've tried it. Your mileage, of course, may vary.

A MAN OF PARTS

In the parts lists, I've specified audiophile-grade components without going insane. There is no question in my mind that the good performance of this preamp comes, in part, from the high grade of pieces from which it's made. On the other hand, you eventually reach a point of diminishing returns—a point which is different for each constructor. If you want to spend the price of a Neumann microphone on capacitors, by all means do so.

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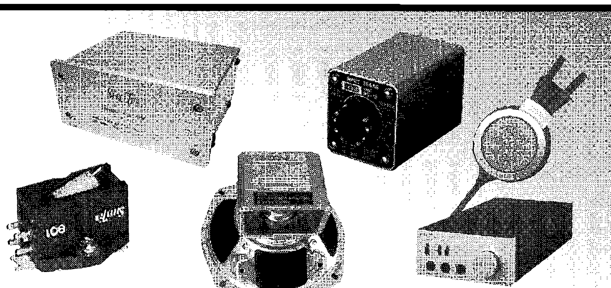

■Cartridge

Model	Price (US $)	Postage (Air Economy)	
Denon DL-102 (MONO)	150	Area I $ 18	China,Korea Hong Kong Taiwan
Denon DL-103 (STEREO)	200	Area II $ 22	Singapore Malaysia Indonesia
Denon DL-103R (STEREO)	250	Area III $ 27	North America Oceania Europe
Shelter Model 501 II (CROWN JEWEL REFERENCE)	650	Area IV $ 34	Africa South America
Shelter Model 901 (CROWN JEWEL SE)	1,300		

These Area I ~ IV are for all products except book.

■Japanease Audio Book

Postage $ 15

Title		Price(US $)
Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
The Joy of Vintage Tube Amps	(Tadaatsu Atarashi)	30
MJ Selected 300B Amps	(MJ)	30
Audio Tubes	(Hisashi Ohtsuka)	35
SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
Classic Valve	(Hisashi Ohtsuka)	40
20TH CENTURY OF AUDIO	(Stereo Sound)	30
The Remembrance of Sound Post	(Susumu Sakuma)	30
Tube Amp Craft Guide	(MJ)	30
FOSTEX CRAFT HAND BOOK	(FOSTEX)	20

■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

■Speaker

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

*Price is for a pair **Air Economy

■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

■TANGO TRANS (28 models are available now)

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

■TAMURA TRANS (All models are available)

**Air Economy

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

**Air Economy

E·I·F·L EIFL Corporation

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Card Charge 6%

TABLE 1
AUDIO CIRCUITS (ONE CHANNEL)

Item	Description	Mfg.	Part #	Source	Cat. #	Cost	Quan.	Total
C101,103,107	470nF 400V polyprop.	Panasonic	ECQ-P4474JU	DK	P3496-ND	2.03	3	6.09
C102	10nF 1000V ceramic disc	Sprague	5GAS10	AL	926-3256	0.34	1	0.34
C104	33nF 600V polystyrene	Reliable	RTX series	WE	RTX04	8.40	1	8.40
	or- 33nF 400V polypropylene	Panasonic	ECQ-P4333JU	DK	P3482-ND	0.62	1	0.62
C105	68nF 600V polystyrene	Reliable	RTX series	WE	RTX06	10.30	1	10.30
	or- 68nF 400V polypropylene	Panasonic	ECQ-P4683JU	DK	P3486-ND	0.82	1	0.82
C106	2μF 400V met. polyprop.	Reliable	PPMFX series	WE	PPMF42	12.00	1	12.00
C108,110*	5μF 400V met. polyprop.	Reliable	PPMFX series	WE	PPMF48	19.75	2	39.50
	or- 10μF 630V met. polyprop.	Solen		WE	SOL33	5.25	2	10.50
JK101	XLR, 3-pin female, gold	Neutrik	NC3FP-B-1	AL	514-1265	5.60	1	5.60
JK103,106*	RCA, panel-mount, gold	Vampire	M1F	WE	M1F	11.95	2	23.90
JK104,107*	¼″, nylon bushing	Switchcraft	N111	AL	932-9400	2.21	2	4.42
JK105***, 108* ***	XLR, 3-pin male,gold	Neutrik	NC3MP-B	AL	514-1064	4.76	2	9.52
KN101	Knob for level control	Davies	4108BJ	AL	543-4108	3.52	1	3.52
R101,102 **	5.62k ½W MF	Holco	H4	WE	H4-5.62K	0.50	10	5.00
R103	150k ½W MF	Holco	H4	WE	H4-150K	0.50	1	0.50
R104,112	100 Ω ½W MF	Vishay/Dale	V4	WE	V4-100	0.20	2	0.40
R105,113,115,117*	50k 5W wirewound	Vishay/Dale	RS5	MO	71-RS5-50K	2.32	4	9.28
R106,114	2.21k 1W MF	Holco	H2	WE	H2-2.21k	0.60	2	1.20
R107	75k 5W wirewound	Vishay/Dale	RS5	MO	71-RS5-75K	1.86	1	1.86
R108,109,111,116,118*	1M ½W MF	Holco	H4	WE	H4-1.0M	1.20	5	6.00
R119 ****	49.9k ½W MF	Holco	H4	WE	H4-49.9k	1.20	1	1.20
S101	SPDT/co min. toggle	C&K	7103SYZBE	DK	CKN1025-ND	4.26	1	4.26
S102	SPDT min. toggle	C&K	7101SYZBE	DK	CKN1021-ND	3.75	1	3.75
SO101,102	8-pin tube sockets	WPI	78S8	AL	719-0014	3.20	2	6.40
T101	1:10 input transformer	Jensen	JT-115K-E	JE	JT-115K-E	73.32	1	73.32
T102***,103* ***	1:1 output transformer	Jensen	JT-11-EMCF	JE	JT-11-EMCF	66.85	2	133.70
V101,102,103*	6SN7 tube	Sovtek	6SN7	WE	6SN7-ND	7.00	3	21.00
VR101	50k audio taper pot							
	[use dual 100k, parallel]	Noble		WE	NOB100K	24.00	1	24.00
VR202 ****	50k dual linear pot							
					Total - single, unbalanced, less-expensive capacitors:			194.85
					Total - dual, balanced, more-expensive capacitors:			414.26

* Used only in dual-output version.
** Match pair to 0.1% or better; quantity is 10 to allow selecting matched pairs.
*** Optional, for balanced output only.
**** Semi-Zen version only.

The fact is, you can make a darned good version of this preamp using polypropylene caps from Panasonic, which you can get at Digi-Key. Better is better, though, and my own balance point falls at the place specified in the price-lists. (I've included the less-expensive caps as alternate choices in *Parts List 1*.)

Many of the resistors in the audio circuits are wirewound, a type not often found in audiophile equipment these days. I was turned on to these by Martin Colloms, who reported an almost accidental finding that, in certain applications, a good wirewound beats the pants off some very fancy metal films. My experience has been that these wirewounds, at any rate, are extremely neutral-sounding, clean, and clear without ever becoming harsh. They're also stable and reliable.

The wirewounds I used aren't billed as non-inductive, but in fact their induc-

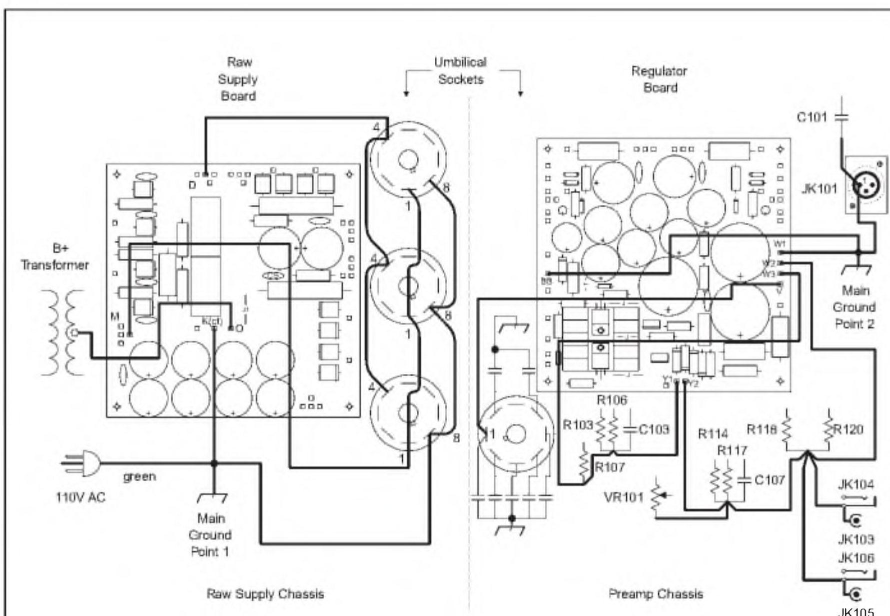


FIGURE 16: Grounding scheme; ground connections shown as heavy lines. Each channel has a single main grounding point, connected to the chassis. Note that VR101's ground connects to the second stage's input resistors, not directly to ground. R120 and JK105–106 are used only in the double-output design.

TABLE 2
REGULATORS (ONE CHASSIS)

Part(s)	Description	Mfg.	Part #	Source	Cat. #	Cost	Quan.	Total
BOX1	19x10x3.5" rack cabinet	Sescom	2RU10	SE	2RU10	46.75	1	46.75
C201,202,204, 206,207,209, 211,212,215	10nF 1000V ceramic disc	Sprague	5GAS10	AL	926-3256	0.34	9	3.06
C203,208	4700µF 35V electrolytic	Panasonic	ECO-S1VA472CA	DK	P6600-ND	3.66	2	7.32
C205,210	1µF 35V tantalum	Panasonic	ECS-F1VE105K	DK	P2059-ND	0.56	2	1.12
C213,214,216	470µF 400V electrolytic	Panasonic	ECO-S2GA471EA	DK	P6857-ND	11.47	3	34.41
C218a-c,,220a-c	100µF 450V electrolytic	Panasonic	ECO-S2WB101BA	DK	P10155-ND	4.25	6	25.50
C217	39nF 400V polypropylene	Panasonic	ECQ-P4393JU	DK	P3483-ND	0.64	1	0.64
C219	330µF 63V electrolytic	Panasonic	ECA-1JFQ331	DK	P5803-ND	1.49	1	1.49
COV201,202	TO3 transistor cover	Eagle	21PTC01032	MO	561-E003	0.97	2	1.94
D201,204	1N4148	Micro	1N4148MSCT-ND	DK	1N4148CT-ND	0.53	1	0.53
D202,203,205,206	1N4001	General Semi.	1N4001GICT-ND	DK	1N4001CT-ND	0.40	1	0.40
D207,208	1N4007	Diodes Inc	1N4007GICT-ND	DK	1N4007CT-ND	0.40	2	0.80
HS201,202	TO3, 25W	Aavid	568303B00000	DK	HS103-1.25-ND	4.73	2	9.46
HS203	TO220, 10W	Aavid	507222B00000	DK	HS114-ND	0.70	1	0.70
LED201	LED assembly, green	IDI	5111F5	DK	L10065-ND	0.84	1	0.84
LED202	LED assembly, yellow	IDI	5111F7	DK	L10067-ND	0.84	1	0.84
LED203	LED assembly, red	IDI	5111F1	DK	L10061-ND	0.84	1	0.84
MOUNT201-202	TO3 mounting kit	Keystone	4725	MO	534-4725	1.13	2	2.26
MOUNT203-204	TO220 mounting kit	Keystone	4724	MO	534-4724	1.13	2	2.26
PL201	Octal plug	WPI	86-CP8	AL	719-0024	2.62	2	5.24
Q201-204	TIP50	ON Semi.	TIP50	AL	568-2055	0.42	4	1.68
R201,204	2.4 Ω, 5W wirewound	Yageo	SQP500JB-2R4	DK	2.4W-5-ND	0.41	2	0.82
R202,205,215,216	22Ω 2W MF (pkg 5)	BC	5083NW22J12AFX	DK	BC22W-2CT-ND	1.60	1	1.60
R203,206	430 Ω 2W MF (pkg 5)	BC	5083NW430J12AFX	DK	BC430W-2CT-ND	1.60	1	1.60
R207	47 Ω, 5W wirewound	Yageo	SQP500JB-47R	DK	47W-5-ND	0.41	1	0.41
R208,209	200k 2W MF (pkg 5)	BC	5083NW200KJ12AFX	DK	BC200KW-2CT-ND	1.60	1	1.60
R210	470 Ω 5W wirewound	Yageo	SQP500JB-470R	DK	470W-5-ND	0.41	1	0.41
R211,213	20k 2W MF (pkg 5)	BC	5083NW20KJ12AFX	DK	BC20KW-2CT-ND	1.60	1	1.60
R212,214	1k 2W MF (pkg 5)	BC	5083NW1KJ12AFX	DK	BC1.0KW-2CT-ND	1.60	1	1.60
R217	110k 2W MF (pkg 5)	BC	5083NW110KJ12AFX	DK	BC110KW-2CT-ND	1.60	1	1.60
R218	220k 2W MF (pkg 5)	BC	5083NW220KJ12AFX	DK	BC220KW-2CT-ND	1.60	1	1.60
R219	47k 3W MF (pkg 5)	BC	5093NW47KJ08AFX	DK	BC47KW-3CT-ND	2.71	1	2.71
SO201	8-pin tube socket	WPI	78S8	AL	719-0014	3.20	1	3.20
VR201,202	LM340K-5.0	National Semi.	LM340K-5.0	AL	288-1298	4.78	2	9.56
Z201-204	82V 5W	ON Semi.	1N5375B	AL	568-2186	0.23	4	0.92
Z205	47V 5W	ON Semi.	1N5368B	AL	568-2179	0.27	1	0.27
							Total:	177.58

TABLE 3
RAW SUPPLY (POWERS THREE CHANNELS)

Part(s)	Description	Mfg.	Part #	Source	Cat. #	Cost	Quan.	Total
BOX2	4x7x12" minibox	LMB	TF-787 PL/UNPD	DK	L109-ND	24.68	1	24.68
-or-		Bud	CU-3011-A	AL	736-3662	19.60	1	19.60
C1-2,5,7-11,16	10nF 1000V ceramic disc	Sprague	5GAS10	AL	926-3256	0.34	6	2.04
C4,6	4700µF 35V electrolytic	Panasonic	ECO-S1VA472CA	DK	P6600-ND	3.66	2	7.32
C12a-d,13a-d	100µF 450V electrolytic	Panasonic	ECO-S2WB101BA	DK	P10155-ND	4.25	8	34.00
C14-15	470nF 400V polypropylene	Panasonic	ECQ-P4474JU	DK	P3496-ND	2.03	2	4.06
D1-8	100V 6A fast recovery (pkg 10)	Diodes Inc.	FR602-T	DK	FR602CT-ND	5.44	1	5.44
D9-12	1000V 1A fast recovery (pkg 10)	Diodes Inc.	UF1007-T	DK	UF1007DICT-ND	0.48	4	1.92
F1-4	1.5A slo-blo (pkg 5)	Littelfuse	31301.5	DK	F322-ND	3.39	1	3.39
F5	0.25A slo-blo (pkg 5)	Littelfuse	31300.25	DK	F310-ND	4.67	1	4.67
FH1-5	Fuse holder	Littelfuse	342014	DK	F006-ND	2.45	5	12.25
LC1	Line cord, 6.5', 3-cond.	Qualtek	311024-01	DK	Q112-ND	5.38	1	5.38
NE1,2	Neon lamp, snap-in	IDI	1030D1	AL	679-5050	1.90	2	3.80
R1,3	1 Ω, 10W wirewound	Yageo	SQP10AJB-1R0	DK	1.0W-10-ND	0.60	2	1.20
R2,4	2k 2W metal oxide	Yageo	RSF200JB-2K0	DK	2.0KW-2-ND	0.23	2	0.46
R5-8	100k 2W metal oxide	Yageo	RSF200JB-100K	DK	100KW-2-ND	0.23	4	0.92
R9	1 Ω, 5W wirewound	Yageo	SQP500JB-1R0	DK	1.0W-5-ND	0.41	1	0.41
R10,11	200k 2W metal oxide	Yageo	RSF200JB-200K	DK	200KW-2-ND	0.23	2	0.46
S1,2	SPST toggle switch	Carlingswitch	2FA53-73	AL	683-0048	3.39	2	6.78
SO1-3	8-pin tube socket, ring-mount	WPI	78S8	AL	719-0014	3.20	3	9.60
T1,2	12.6V 2A	Allied	6K36HF	AL	227-0046	9.18	2	18.36
T3	300-0-300V 100mA	Hammond	P-T272BX	AN	P-T272BX	44.00	1	44.00
ZNR1-3	150VAC 2500A transient absorber	Panasonic	ERZ-V10D241	DK	P7207-ND	0.42	3	1.26
							Total:	187.32

tance must be pretty low, as they measured pretty close to flat up to 700kHz or so.

For audio coupling capacitors, I've used mostly MultiCaps for several years now. I try to use film-and-foil (polystyrene when I can), but in large sizes metallized polypropylenes are all they sell. I've been warned many times that all metallized-film caps are failure-prone, but I haven't had one of these go out on me yet.

The biggest capacitor problem in this design is the output capacitor on stage 2. If you're driving a DAT recorder with an input impedance of 47k, like my Sony, you'll be okay with a 2.0µF MultiCap (the largest polypropylene-and-foil they make with a 400V rating). The bass will roll off at 1.6Hz, the same as the input stage.

If you're operating into an ADAT, however, or one of many semi-pro multitrack recording systems or sound cards, you're more likely facing an input impedance closer to 10k, and that requires at least a 10µF capacitor at the output. I use a metallized polypropylene in parallel with a 2.0µF 400V MultiCap, which has given good results.

Supply bypass capacitors are all Panasonic polys, which are easy to find and do their job reliably and well.

Tubes? Ah, there's a whole kettle of worms, which I don't propose to open. The preamp will work fine with 6SN7s from Sovtek, which are reasonably-priced and easily available. In my experience, some new-old-stock (NOS) tubes—RCA and GE—can sound a bit "beefier," but many NOS tubes are quite microphonic, which rules them out for preamp use. *You pays your money and you takes your choice*, or perhaps your chance.

Don't skimp on the mechanical parts. Professional audio gear—which this is—should keep on working for years, and be trustworthy during that time. You can't get reliability with cheap switches and pots. I used gold-plated switches and jacks, high-quality conductive-plastic level controls, and so forth. I expect them to pay for themselves through good behavior over the years.

Addendum on parts: I compiled my original parts lists a few years ago.

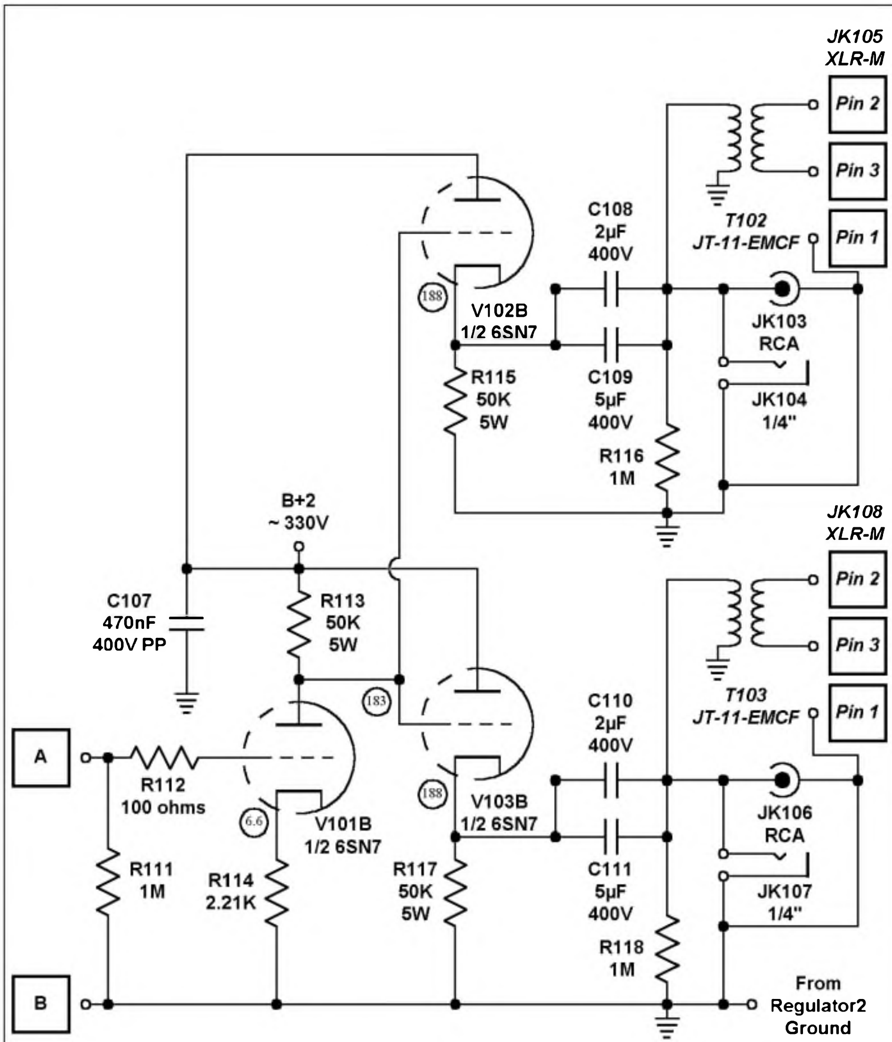


FIGURE 17: Preamp, stage 2, with dual isolated outputs. The output transformers are still optional—use one, two, or none. If you use transformers, you may omit R116 and/or R118.

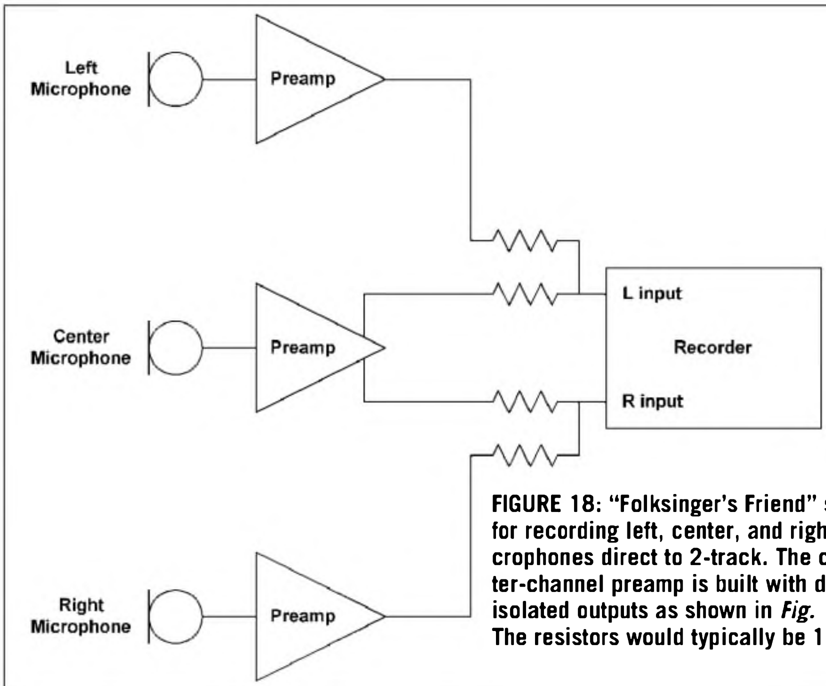


FIGURE 18: "Folksinger's Friend" setup for recording left, center, and right microphones direct to 2-track. The center-channel preamp is built with dual, isolated outputs as shown in *Fig. 17*. The resistors would typically be 15k.

When I began updating them for this article, I discovered that the parts world had shifted a good deal. The Bourns conductive-plastic 50k pots I used for the level controls, for example, do not seem to be available in small quantities any longer. (I've substituted a 100k stereo Noble pot, with the two halves wired in parallel.) Also unavailable were the General Instruments diodes I'd used for power-supply rectifiers (I suggest other high-speed diodes, still available from Digi-Key).

No doubt things will change some more by the time this article appears. Meanwhile, I've tried to keep the information as up-to-date as possible. All prices are current as of late August 2001, as verified on the sellers' websites.

BUILDING BLOCKS

The basic design of this preamp is one input and one output per box. The power supply will drive up to three preamp channels, and it's possible to double them up, but at a price.

The price is heat. The 6SN7 eats up a lot of filament power (twice that of a miniature tube such as the 12AX7 or 12AU7), and we're not exactly starving the plates for current either. Each tube (with its associated components) dissipates about 5.5W, not to mention pass transistors, zeners, and dropping resistors in the supply regulators.

I wanted to keep the preamp's box sealed to ward off potential microphonics. I decided that if the Dynaco PAS preamp could run four 12AX7s in an almost-sealed box and still remain reliable for years, I could run two, maybe three, 6SN7s in a somewhat larger box, provided I left it some breathing space above and below. It's worked for four years with no problems.

That means carrying extra suitcases to remote recording gigs, but it's a price I'm willing to pay. I also get separate power supply regulators for each preamp channel, which eliminates any measurable crosstalk.

You might decide differently; I think it'd be reasonable to put two channels of basic preamp into a single chassis, powered from a single regulator card, provided you replaced the solid top and bottom of the cabinets with perforated screens. (You'd also need to change a few resistors on the regulator board.) This might be a useful compromise if you need to travel light.

Even with ventilation, a two-channel unit would run hot—you'd need to allow a couple of rack spaces above and below for air to circulate. Heat is the enemy of reliability. (For another possible two-channels-in-one-box solution, see the section entitled "Semi-Zen".)

ISOLATIONISM

I built one preamp with a single input, but two identical outputs (*Fig. 17*). Basically, I hung a second, parallel cathode follower on the plate of the voltage amplifier in stage 2. This has little effect on the voltage amplifier; the load capacitance is doubled, but since cathode followers have no Miller effect (no amplification), the capacitance is still quite low. In practice, the bandwidth of the preamp as a whole is still determined by the input transformer.

What do I get for my extra tube? Three interesting possibilities. The first, and most pedestrian, is the possibility

ADDENDUM

Some work on another project led me to experiment with constant-current sources in the cathode followers of Big Mike, replacing the usual resistors. My results were mixed. In the input stage, the distortions (THD-400 and SMPTE-IM) were lowered by a factor of about three when I used a current source of 1.6mA, but at about +32dBu I ran into brick-wall hard clipping, rather than the softer overload of the resistor-based stage. I'd rather avoid hard clipping in a stage that may see hot signals, so I decided to stick with the resistor.

On the other hand, the current source proved a real boon in the second stage, with no obvious drawbacks. Operating into a 10k load at −10dBV nominal output, the distortions were already quite low, but current sourcing (5mA) lowered them still further— .08% SMPTE-IM at +6dBV, clipping level for an ADAT or the equivalent. Higher harmonics in the THD-400 measurement dropped to almost nothing, leaving only second and a touch of third.

The current source shone brightest, though, when I ran at 4dBu nominal output, the standard balanced professional level. At 20dBu into 10k (clipping on an ADAT), my second stage with a 5mA current source yielded .13% SMPTE-IM and .025% THD-400, the latter almost entirely second and third harmonics. (Results with a 50k resistor were 2–3× higher, with many more high harmonics in the THD-400 spectrum.)

In the unlikely event that you're driving a Studer multitrack recorder and need to push nominal 4dBu into a 5k load, a 9mA current source lets you do that with only .21% SMPTE-IM and .03% THD-400 at 20dBu, again with a remarkably non-toxic harmonic spectrum. A resistor-loaded stage simply won't work into 5k; it hard-clips at 13.5dBu. This is the equivalent of only 9.5dB of headroom on an analog recorder (pretty poor), or 6.5dB below full-scale on a digital recorder. I think a current source output is desirable if you're driving 10k loads, and mandatory if you need to drive 5k.

Figure 20 shows the constant-current source I used for my tests. Yes, it's a transistor, and a bipolar one at that. I'm not convinced that a vacuum-tube current source will perform any differently or sound better, but if you prefer (and don't mind the extra heat generated by an additional filament) by all means use a pentode instead. An EF86 might be a good, easily-available choice.

The only down side is that, for best results, you must optimize the current source for a particular load impedance. Here are the numbers:

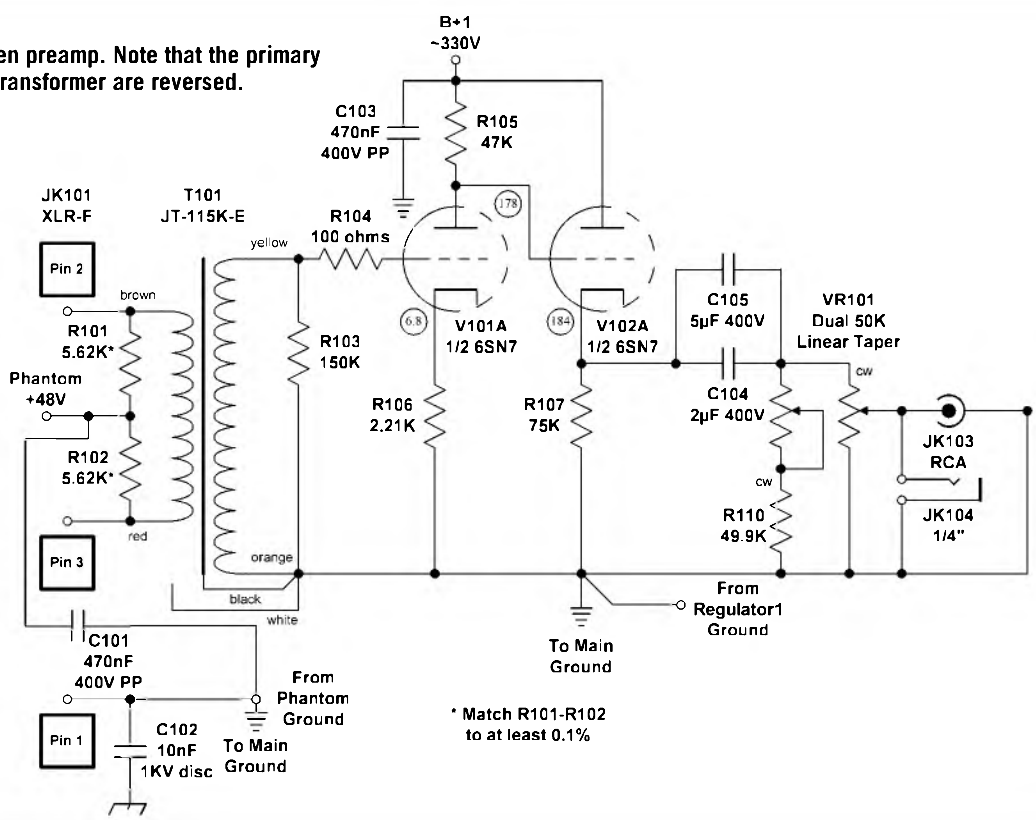
Load impedance (ohms)	Current (mA)	R401 (ohms)
5k	9	71.5
10k	5	130
15k	3.5	187
20k	3	215
33k	2.1	309
50k	1.7	383

The resistors are rounded off to nearest E96 values. If you drive a variety of loads, it should be easy enough to arrange parallel resistors and a switch.

On another topic: Since I published the original mike preamp article back in 1996, microphones have gotten hotter—in fact, there are now several on the market with higher outputs than the Neumann U-87 that was previously the champion, including Neumann's excellent (and sensibly-priced) TLM-103. As a result, I've found myself needing a pad to avoid overloading the input circuit and saturating the transformer. I'm allergic to switches in low-level circuits such as microphone inputs (a set with bad contacts nearly got me fired during a live TV fundraiser). I'd much rather carry an inline pad in the case with my preamp.

Figure 21 shows a 20dB pad that doesn't interfere with phantom power. You can build it into an XLR-female/XLR-male shell, available from Neutrik (part #NA3FM). Use good resistors, of course, and match R501–R502 to within 0.1% for best noise rejection.

FIGURE 19: Semi-Zen preamp. Note that the primary leads on the input transformer are reversed.



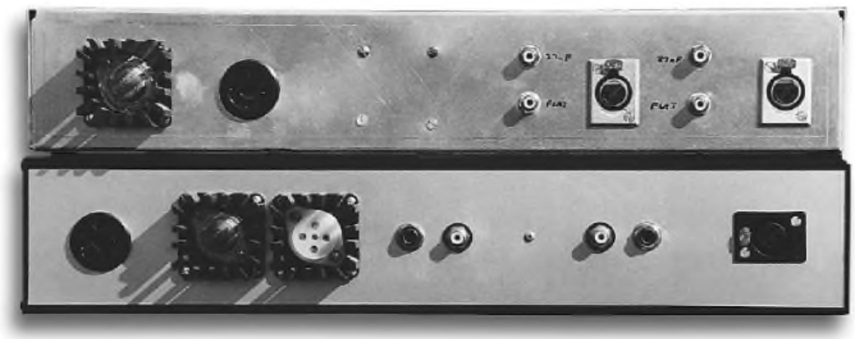


PHOTO 2: Rear views of both preamps. The prototype Semi-Zen has two sets of RCA outputs; the upper ones have additional series capacitors to provide bass rolloff.

of simultaneous unbalanced and balanced outputs, isolated from one another. They'd be the same level, though, which is only marginally useful.

The second is the chance to feed one channel into a monitor mixer while the other, isolated channel feeds into a computer sound card. This allows you to monitor the input signal without "latency," the delay caused by converting the signal from analog to digital and back. Latency is less of a problem these days than in the early years of computer audio, but it still exists, and it's nice to be able to dodge around it completely by using the extra output for monitoring.

The third possibility is illustrated in *Fig. 18*, which I call the "folksinger's friend" direct-to-2-track setup. This allows me to record a stereo pair of microphones on a musician's instrument, while using a third, centered mike on the voice. The resistor network mixes the three signals passively, while the isolated outputs of the center-channel preamp preserve stereo separation. This setup has other uses, including recording three spaced omnidirectional mikes, hung in front of an orchestra or choir.

Of course, there's a drawback: You need an extra half-tube for the additional cathode follower. More heat, more parts. The same circuit works reasonably well using 12AU7 tubes for the followers (two 12AU7s dissipate the same filament power as one 6SN7), but I must say that I think the 6SN7 sounds less colored to my ears, with better impact in the bass.

SEMI-ZEN

No, not a meditation on diesels. My first prototype of this circuit was a pair of input stages, mounted in an aluminum chassis with the usual outboard supply. I thought perhaps I could use it for two-mike stereo recordings (at the time, I was doing a lot of choirs and classical recitals), using the level control at the input of my DAT machine.

Sonically, it was a smashing success—in fact, the best-sounding preamp I'd built, and one of the best I'd heard. Ergonomically, however, it was a disaster. The problem was that microphones don't always match, and the 20-year-old pair of Neumann KM-84s that I use for

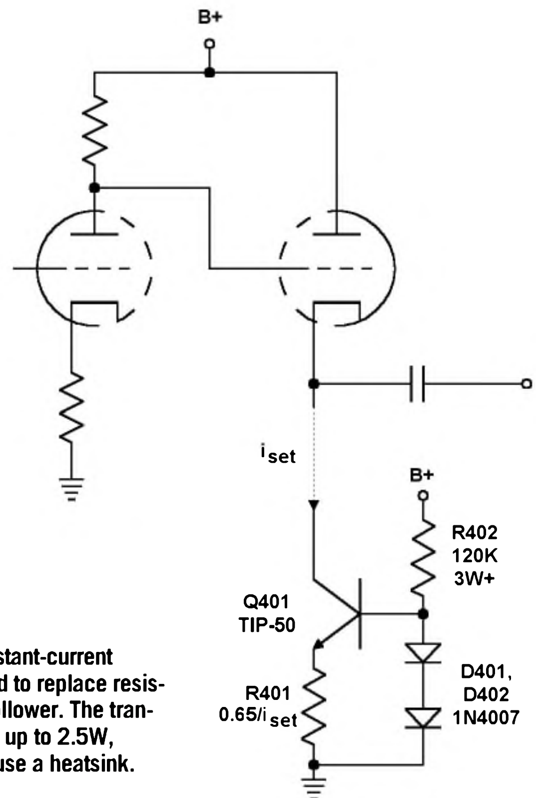


FIGURE 20: Constant-current source, designed to replace resistor in cathode follower. The transistor dissipates up to 2.5W, worse case, so use a heatsink.

classical recording were never closer than 2dB in sensitivity.

Unfortunately, my DAT machine's level controls are ganged, with a slip clutch, and adjusting balance using a setup like that is about as much fun as having a tooth pulled. So I quickly added a pair of passive controls at the output; the resulting circuit is shown in *Fig. 19*. These ganged linear pots, when used into a load impedance of 47k, maintain a fairly constant load on the tubes.

I've also found that this stage works well into an ADAT, using the −10dBV unbalanced RCA jacks. The 10k input impedance means the load is no longer constant when the level is turned up, but in practice the preamp doesn't seem to mind.

There are several constraints in this Semi-Zen approach. (True Zen, à la Nelson Pass, requires a single active device rather than a pair.) The first is gain. Between the transformer and the tubes, only about 38dB of gain is available. This is enough for most crossed pair recordings using condenser microphones (especially since the DAT recorder has some gain of its own), and it's fine for close-miked instruments, again using condensers, but I wouldn't pair it with a low-output ribbon mike. It's also a problem if you need a bass rolloff. (I suggest inserting a 10nF capacitor between the input transformer and its 150k terminating resistor, plus a switch to bypass it.)

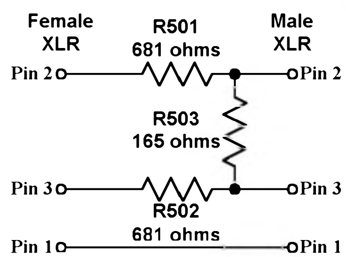


FIGURE 21: Inline 20dB pad, designed to fit inside an XLR-female/XLR-male plug.

Because there's only a single gain stage, the Semi-Zen preamp inverts polarity. I wired the transformer's input leads in reverse to correct that. The maximum output impedance of the Semi-Zen is about 12.5k. Use short cables to connect its outputs to the recorder's inputs.

With all this finickiness, why do I mention the Semi-Zen at all? Because it sounds so good. This very-short-signal-path design is the best, least-colored, most neutral preamp I've built to date. The standard two-stage preamp (with or without extra output) comes very close, though, and it's far more flexible and universally applicable.

Further parts addendum: I can no longer find current-stock dual 50k linear pots at any of my standard suppliers. If you decide to build a Semi-Zen preamp, I'm afraid you're on your own—

PARTS SOURCES

AL: Allied Electronics, Inc.
7410 Pebble Dr.
Fort Worth, TX 76118-9981
(they have multiple branches)
800-433-5700
Web: www.alliedelec.com
Minimum order: $50

AN: Antique Electronic Supply
6221 S. Maple Ave.
Tempe, AZ 85283
480-820-5411
Fax: 800-706-6789
Web: www.tubesandmore.com
Minimum order: $10

DK: Digi-Key Corp.
701 Brooks Ave. South
Thief River Falls, MN 56701-0677
800-344-4539
Fax: 218-681-3380
Web: www.digikey.com
Minimum order without surcharge: $25

JE: Jensen Transformers, Inc.
7135 Hayvenhurst Ave.
Van Nuys, CA 91406
818-374-5857
Fax: 818-374-5856
Web: www.jensen-transformers.com
No minimum order

MO: Mouser Electronics
958 N. Main St.
Mansfield, TX 76063-4827
800-346-6873
Fax: 817-483-6899
Web: www.mouser.com
No minimum order

SE: Sescom, Inc.
517 Main St., PO Box 839
Wellsville, KS 66092
785-883-3009 (orders only 800-634-3457)
Fax: 785-883-4422
Web: www.sescom.com
Minimum order: $30

WE: Welborne Labs
PO Box 260198
Littleton, CO 80126
303-470-6585
Fax: 303-791-5783
Web: www.welbornelabs.com/index.html
No minimum order

you may find good-quality units at a surplus store. You could also use the pot specified in the standard preamp design, but you'd then face a drastically-varying load when you cranked the pot wide open. Or, of course, you can use the Semi-Zen without a level control, if you're recording into a machine that has independent controls for the two inputs.

IN THE CRUCIBLE

So how does the preamp fare when put to the test—using it for recordings? Very well; it's clean, quiet, smooth without polishing away the corners of the sound, rich without being muddy. I've made some very good recordings with these boxes, and once they were up and running, none of them—knock wood—has ever let me down on a gig.

What are their limitations? The basic two-stage design will drive a professional-level (+4dBu) input with an impedance of 10k or higher, but it won't drive a 600Ω load. These days, that's not terribly important. The only gear you're likely to find that still adheres to that old telco-derived standard is a retro re-creation of a classic compressor, such as the Teletronics 1176, or perhaps a replicated Pultec equalizer. Those lovely gadgets require hefty driving currents, but heck, most of us couldn't afford them anyway.

In any case, I suggest you stick with unbalanced −10dBV inputs wherever possible. They put less stress on the electronics, and the distortion is almost always lower. (Not only with this preamp; most of the commercial designs I've tested are also cleaner at the lower "semi-pro" level.)

IT'S A WRAP

To my admittedly biased ears, this preamp design, in any of its many possible configurations, produces fully professional results. It's not a "tube-effects" box trying to mask poor-quality digital recording with mud and calling it warmth, but, rather, a low-coloration preamp. It's warm, but not in the sense of adding warmth. Instead, it's warm because it isn't cold; it doesn't add the icy tinge endemic to many solid-state preamp designs—including some very expensive ones.

If you're looking for a microphone preamp that does justice to a high-quality recording setup, digital or analog, you may want to give Big Mike a try.

THANKS

As I said last time, no one creates in a vacuum, and this preamp design has benefited from the advice and support of many people. (Any errors are all my own, however.) Thanks to my teachers Dana Sawyer, Dan Landiss, and Steve Fuller; also to Fred Forssell for a great hour of talk at the AES show, to Ed Dell for providing a forum where I could both write and learn, to Jimmy Smith for asking good questions, to Wylie Williams for technical good sense, support, and friendship, to Brad Sarno for being a good uki and lending his ears and studio to listening tests, and Rebecca Taylor for warmth and encouragement. And, most of all, thanks to all the wonderful musicians whose art makes this worth doing. ❖

A Safe Power Supply for Valves

This power-supply design ensures extended tube life and user safety, in addition to hi-fi performance. **By Mike Panymo**

When after a lifetime of experience I heard of the valve renaissance, I refreshed my memory by toying with the old familiar circuits on paper. I too had drifted into the transistor age and was glad to say goodbye and good riddance to the big, clumsy, fragile, hot, inefficient, noisy, unreliable, and expensive devils—until a friend put my Mozart sonatas on his CD and valve rig. Wow! I was off, and I lost no time in that winter of '96.

I asked myself what I could build today using my experience and ordinary off-the-shelf materials. The answer was a power supply. Quality and safety always come first. The mains-in does not need a third ground wire, but it must be neatly soldered to a twin-fuse open panel mounted on a sheet of durable plastic glued to the chassis and covered by a transparent cap. From there it goes to a DPST switch with a voltage-dependent resistor (VDR) across the passive contacts, again covered by a plastic cap.

For the B+ transformer, I chose a toroidal 220 2 × 127V 200W type with a separate mains fuse, readily and cheaply available as a USA-European appliance adaptor. For the heaters I chose a toroidal 220 2 × 12V 100W type, also easily available at low cost anywhere in the world, and also with a separate mains fuse.

The real damage I had encountered in valve amps was invariably caused by bias failure. I decided to secure the bias voltage to the basics of the amp itself—the heaters and B+ supply. I also remembered heater shorts in power valves and the noisy humbuck pots, as well as the necessity for a hum-free heater supply and a reasonable warm-up time.

I decided to make a proper job of it and came up with the design in *Fig 1*. Summing 12 + 12V gives a –32V DC that is easily stabilized at –25V for bias, as well as to supply a positive temperature coefficient (PTC)-controlled delayed relay to switch B+ (see the Philips book, *Components and Materials*, Part 11, p. D16). When put in series, the heaters simply surge-protect one another. Although fed by AC, the heaters see DC with respect to the chassis, and are entirely hum-free.

I built it, measured it, tested it, and did my best to blow up a series of worn-out power valves. In addition to a box full of fuses, I managed to blow the 24V rectifier, which promptly shut down the B+. At the 12 + 12V center tap, I included a 4,700/40V for extra filtering and to provide a –15V DC supply for a preamp.

I switched the 320V B+ through a fuse to ground to provide the power valves with an easily digestible 160V during warm-up, which I set at three minutes. For safety, I drew the line at 375V, which you can reach simply by slipping a small low-volt transformer between the center tap of the B+ transformer (don't forget the fuse). And that's it. No transistors, but a simple voltage stabilizer, LM7912 or the equivalent, for negative bias.

And for those who insist on a valve rectifier, I have the ultimate delay solution (*Fig. 2*)—believe it or not, just a single component. The buffer NTC Philips type 2322 644 90005 (or equivalent) in series with the heater takes three minutes over a steeply sloping temperature/Ω curve to drop from 15Ω to 1Ω, so that you may now safely increase the first filter cap from 50 to 200µF. The heater winding should be 1V higher than the heater volt-

(to page 69)

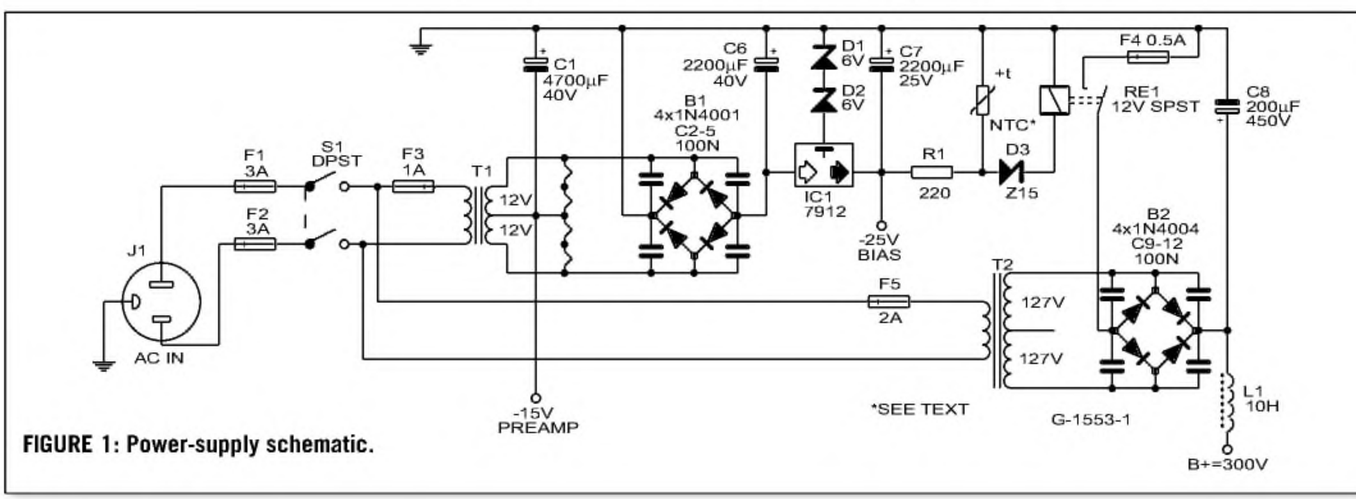


FIGURE 1: Power-supply schematic.

The Infinite Box: Constructing a Subwoofer, Part 1

Part 1 of this article covers the design and construction of a subwoofer using the infinite box (IB) concept, which was featured in the January and February 2002 issues of *aX*. **By G. R. Koonce and R. O. Wright, Jr.**

To recap, our IB concept involves the use of a driver with high-density damping material filling the rear of an open-back box. This damping material modifies the driver's parameters to yield a highly damped system with the option of frequency-response peaking available. Additionally, the IB approach yields a fairly small box when used with large drivers.

We use a 12″ DVC woofer in the IB approach to yield a moderate-sized subwoofer (*Photo 1*). Our goal was to show that the IB approach produces the same fast, taut bass with a large driver as with smaller drivers. We also discuss an IB subwoofer built using the 15″ version of this same woofer.

THE DRIVER

The driver is the American-made Parts Express #295-185 Dayton 12″ woofer with dual voice coils. *Table 1* shows the properties of this driver from the Parts Express catalog. The 15.1mm X_{MAX} means this driver has 1.2″ peak-to-peak displacement capability, making it ideal for a subwoofer. Lacking experience with high-displacement drivers, we decided to do some testing and determined that this driver showed some unexpected behavior, which we document here.

The first test was to measure the T/S parameters for our single unit. *Table 2* compares the catalog values with what we measured, showing reasonable agreement. These values are for the driver as received, before any break-in. We also ran frequency-response tests, which we cover.

BOX CHOICES

An IB was pre-ordained for this work, but it is useful to compare the computed performance of this driver in various box types. Feeding the T/S parameters to design software produced optimum box designs and predicted responses. A lossless closed box (CB) design with B2 alignment (Q_{TC} = 0.707) yields the minimum CB f_3 value. A vented-box (VB) design using a jammed B4 alignment yields the minimum f_3 value with a flat response.

Table 3 compares these results with those for an IB design. The net box volume is shown for the CB and VB. The IB volume shown is the dead-air (DA) volume, which is the volume of air between the driver and the start of the damping material.

We currently have no way to predict the f_3 value for an IB other than an upper limit. The overall (OA) depth shown is based on a fixed construction approach. The VB had a 5% volume penalty applied for loss to the port. The IB has a 5⅛″ depth allowance for the damping material and its mounting.

Clearly, the CB performance does not look great on paper. The VB shows the best response in terms of f_3, but with the biggest box. Not reflected in this table is the fast, taut bass the IB approach can produce. On paper, the CB and VB designs will have flat responses, while the IB design would show peaking. Acoustic testing would show different results for a CB and VB with this driver.

RESPONSE PEAKING

One advantage of the IB approach is that it allows building a speaker with a bass response peak while still maintaining a highly damped system. As developed in reference 1, you can use such peaking to offset the effect of diffraction spreading loss (DSL), which tends to dip the low-frequency on-axis response of any enclosure sitting out from the rear wall. Those not familiar with DSL may also want to read reference 2. The initial IB subwoofer design did have such peaking, which tends to keep the IB small.

We measured the baffle-mounted frequency response of the 12″ woofer via near-field technique (*Fig. 1*). Taking the generally flat region 200–400Hz as 0dB, this driver has a built-in response peak of about 3.6dB around 70Hz. Would this peak show up in boxes using this driver?

Figure 2 shows the T/S parameter predicted response in a 2.85ft^3 lossless CB, very flat with f_3 near 50Hz. The measured response for this driver in such a CB (*Fig. 3*) shows a peak of 5dB at 70Hz with f_3 around 26Hz. This is not a bad response and would be a reasonable way to use this woofer.

The computer determines that a 2.85ft^3 VB—close to the optimum B4 VB volume from *Table 3*—with this driver should produce the response of *Fig. 4*, flat with f_3 at 26Hz. We tested the driver

PHOTO 1: Front view of completed subwoofer.

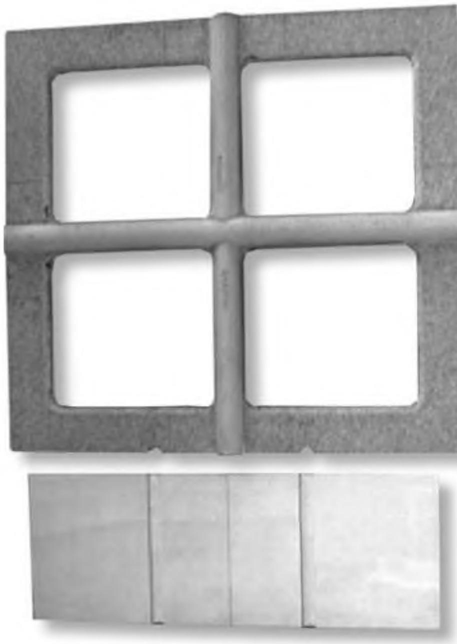


PHOTO 2: Front view of finished damping panel and bottom tunnel board.

in a 2.85ft^3 VB tuned slightly low at 22Hz. *Figure 5* shows the measured cone and port responses with the port response level corrected for the driver/port area difference. Ignoring the noise caused low-frequency wiggles; these are typical VB curves with the port radiating over a frequency range about the box's tuned frequency and the cone response suppressed in this region.

Software combined these two responses to produce the system response (*Fig. 6*). Rather than the computer-predicted flat response, you see a 6dB peak around 50Hz and f_3 at about 18Hz. This is also a reasonable way to use this woofer.

The basic result is that this driver has several dB of response peak built in. The measured response in a box is not what software predicts from the driver's T/S parameters. If you are using it only up to 100Hz or so, then you don't get the peaking and f_3 values reported here, but any attempt at using it as a woofer to around 400Hz must take this peaking into consideration. We did no testing on the 15″ version of this driver.

BASIC IB DESIGN

You design an IB by developing a CB prototype. The Q_{TC} value used in the CB-prototype design will predict the peaking in the IB system response as covered in reference 1. With peaking built into the driver, we redesigned the CB prototype with a lower Q_{TC} to limit additional peaking. Prior work has shown designing with a Q_{TC} value of 0.6 to 0.7 should add little response peaking at the sacrifice of a bigger DA volume and thus a larger box.

One problem is what driver data to use—catalog values or the test values for

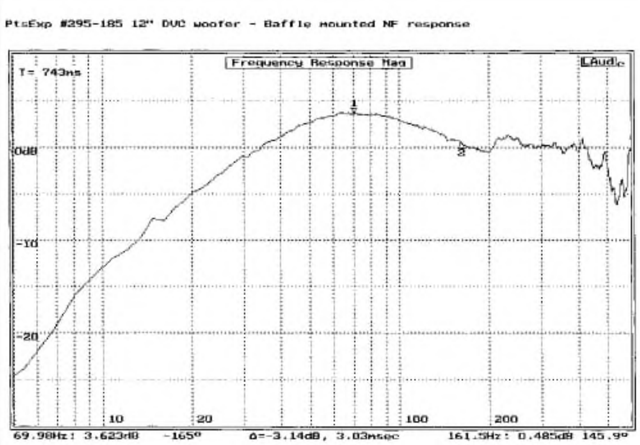


FIGURE 1: Driver frequency response in test baffle.

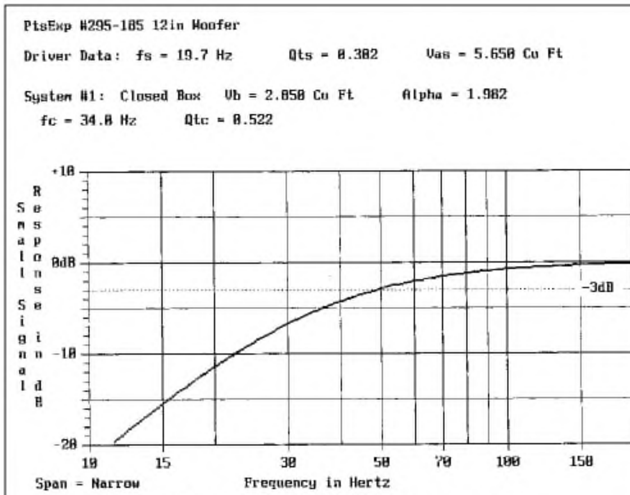


FIGURE 2: Predicted response of driver in closed box.

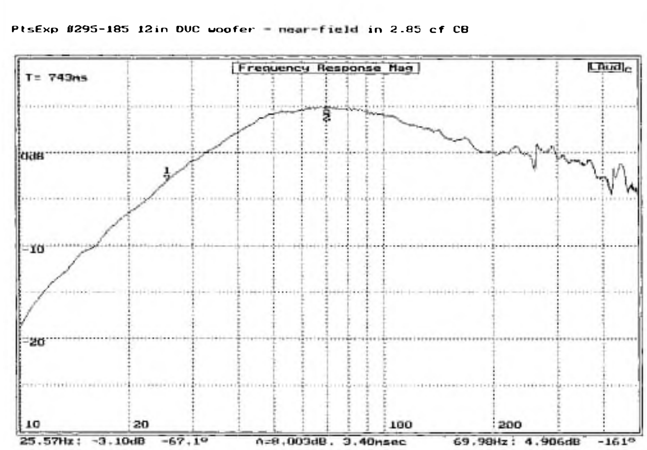


FIGURE 3: Measured response of driver in closed box.

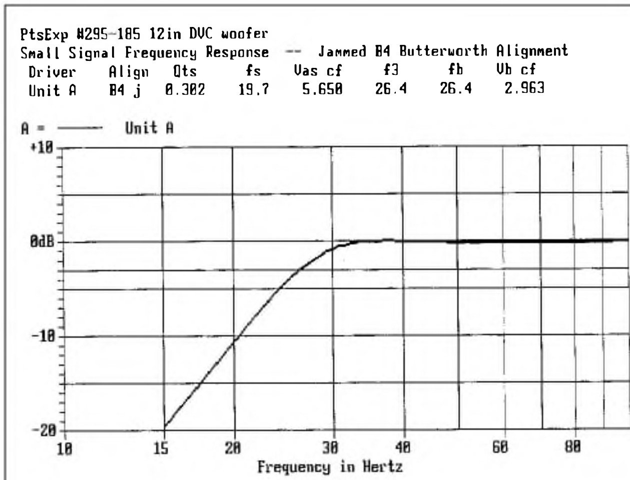


FIGURE 4: Predicted response of driver in vented box.

our single unit. We decided to use catalog values, which we considered more representative of any driver a reader might purchase. Based on catalog data, the IB should have a DA volume of at least 2,100 in^3 to keep the CB-prototype Q_{TC} below 0.7. We decided to shoot for a DA volume of about 2,100 in^3 (1.22ft^3).

With any box type there are items in the construction that affect the net volume. This is a large-volume driver, but front-mounted into the box so the mounting hole will tend to offset the volume lost. Additional volume is lost to any bracing in the box. We will see the DA volume of the IB ends at a "damping panel" that retains the inner end of the damping layers. This damping panel has large holes that add to the DA volume.

When all this was worked out it resulted in the gross DA volume between the front panel and damping panel being close to the net DA volume. We are thus looking for a gross DA volume of about 2,100 in^3.

WOOD THICKNESS AND FRONT-PANEL SIZE

We are not fans of using very thick ma-

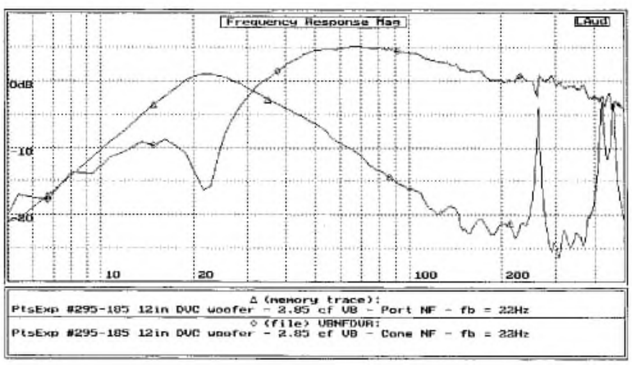


FIGURE 5: Measured cone and port responses in vented box.

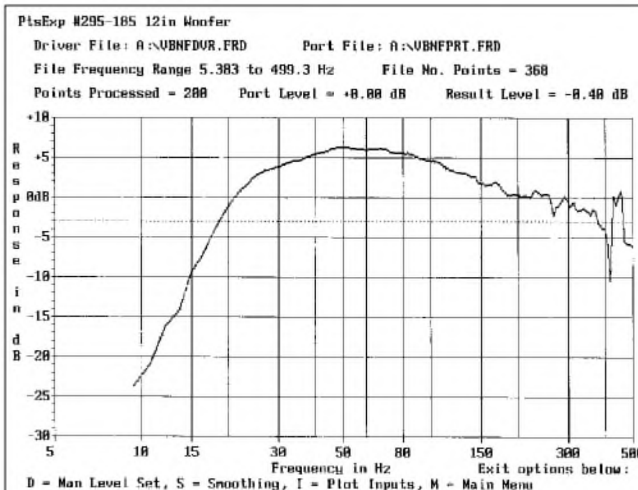


FIGURE 6: System response for driver in vented box.

PHOTO 3: Bracing blocks viewed through driver hole.

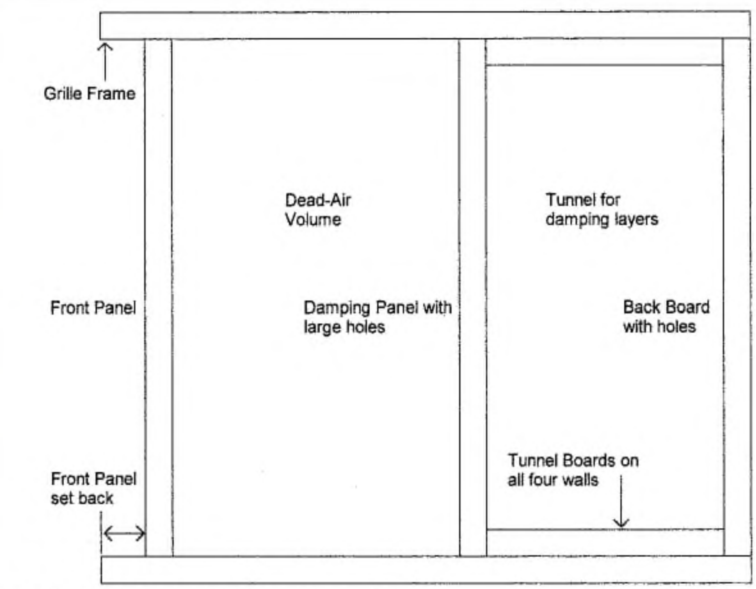


FIGURE 7: Basic layout of an infinite box.

TABLE 1
CHARACTERISTICS OF THE DAYTON #295-185 12″ WOOFER

Power capability—350W RMS per coil, 600W RMS total
Voice-coil diameter—2″
Voice-coil inductance—1.81mH
Nominal impedance—8Ω/coil, 4Ω coils in parallel
DC resistance—2.69Ω, coils in parallel
Frequency range—20 to 450Hz
Magnet weight—112 oz
SPL—87.4dB/W/m, 90.4dB/2.83V/m, coils in parallel
X_{MAX}—15.1mm
Weight—18 lbs

terial for speaker box construction. The thicker the material, the higher its mass, and thus the more energy it can store and the slower it will release it. Thus we believe that a very thick material requires even more bracing than a more reasonable thickness to keep the wall resonant frequency sufficiently high. As developed in reference 3, materials such as MDF and particleboard

TABLE 2
T/S PARAMETERS FOR DAYTON #295-185 12″ WOOFER

ITEM	CATALOG	MEASURED	UNIT
f_S	21.7	19.7	Hz
Q_{ES}	0.38	0.31	
Q_{MS}	20.37	12.33	
Q_{TS}	0.37	0.30	
V_{AS}	4.25	5.65	Cubic feet

tend to optimize in the ¾″ to 1″ thickness range.

We decided to build this subwoofer out of ¾″ thick industrial-quality particleboard with bracing where needed. You can go with thicker material if desired by simply correcting the box depth and outside dimensions; material up to 1″ or 1⅛″ thick should be reasonable. Switching to MDF material in the same thickness range should not modify performance. One way to have thicker walls would be to build the unit as

TABLE 3

COMPARISON OF THEORETICAL BOX DESIGNS

BOX TYPE	V_B OR DA[1]	f_3	OA DEPTH[2]
Closed Box	1.26	46.1	11.4
Vented Box	2.96	26.4	27.8
Infinite Box	1.08	<46	14.9

Notes: 1—V_B in cubic feet or for IB dead-air (DA) volume in cubic feet.
2—Overall depth in inches for fixed construction, see text.

shown and then finish it by carefully gluing ½″ thick dress material on the top, sides, and bottom.

We built this enclosure with a radial-arm saw that has a cutting limit of 15¼″ for material ¾″ thick. To minimize the amount of hand-cutting necessary, we designed the box to have all sides, except the top, limited to 15¼″ in one dimension. The construction approach is relatively insensitive to cutting accuracy.

You can build this box by purchasing one 4′ × 8′ sheet of ¾″ particleboard and having the lumberyard cut off two 8′ strips 15¼″ wide. Use the wider strip remaining to make the top board. As long as the two strips are both cut to the same consistent width, the box will fit together.

Figure 7 shows the basic layout of an IB. The front panel has been recessed by 1¼″, allowing the sides of the box to form the grille frame. If you don't like this approach, you can modify the dimensions to remove this grille frame. The 1¼″ recess value keeps the grille cloth far from the cone of this high-displacement driver. This driver sticks out ¾″ from the front panel, so the minimum grille frame depth is probably about 1″.

As discussed, the region from the front panel to the damping panel defines the DA volume. We used bracing in this region to stiffen the front panel and side walls. Behind the damping panel all four walls of the box are lined with nominal 1″ thick wood to form a "tunnel" for the damping material. The depth of this tunnel is set by the total thickness of damping material.

Testing[1] showed that the total thickness of damping material is not a critical design factor. To allow use of nominal 1 × 6 wood for the tunnel boards, we decided to use five 1″ thick layers of Owens-Corning #705 damping material, making the tunnel for our material 5⅞″ deep. This tunnel depth should be modified to fit the measured thickness of your damping material. It is only nominally 1″ thick. The back board of the box screwed onto the edges of the tunnel boards should compress the damping layers by about ⅞″.

To minimize rear leakage at high frequency, the open area at the rear of the box should be limited to a value of about 110% to 120% of the driver-piston area. The piston area for this driver is about 75 in^2, so large holes should be made in the back board totaling about 83-90 in^2.

ACTUAL IB ENCLOSURE

Before building this subwoofer, you might want to wait and read the section entitled "What Would We Do Different-

TABLE 4

SIZES OF ¾″ THICK PARTICLE-BOARD PIECES TO BUILD BOX

PIECE	DIMENSIONS IN INCHES
Top	16¾+ × 18⅞—See text
Sides (2)	15¼ × 18⅞
Bottom	15¼ × 18⅞
Front panel	15¼ × 14½
Damping panel	15¼ × 14½
Back board	15¼ × 14½

ly" in Part 2, which lists some things to watch for and possible improvements.

Figure 8 shows a front view of the IB enclosure staying with a 15¼" limit in one dimension for all boards except the top board. This results in a front panel that is 15¼" wide by 14½" high. *Figure 9* shows the side view of the box.

The distance between the front panel and the damping panel is 9½", giving a gross DA volume of 2,100 in^3. This results in an overall box depth of 18⅛", width of 16¾", and height of 16". The bottom board sits right on the floor. If you don't like this you could add feet or spikes to the bottom of the box.

Table 4 shows the dimensions of the ¾" particleboard pieces used to build the box. All but the top has one dimension of 15¼", so you can easily cut the pieces from the 15¼" strips of particleboard. The approach used for the top board was to cut it about ⅛" too wide, so after installation it could be routed with a flush-cutting bit to make a good fit to the side walls. The other long "joints" are on the box's bottom.

If the lumberyard cut your particleboard strips to a width other than 15¼", you should correct the 14½" dimension of all three vertical panels. The proper dimension is the width of your strips minus the thickness of your particleboard.

FRONT PANEL

The front panel requires only the large hole for the driver and holes for the eight mounting screws. Cut an 11" diameter hole centered on the front panel for the driver. No chamfer or other contouring is needed on the back of this hole with a ¾" thick front panel. Placement of the mounting screw holes is important because of the front-panel bracing.

Orient the driver so that the screw holes are on both the vertical centerline and horizontal centerline of the driver hole (*Fig. 8*). Because of the weight of this driver (18 lbs) and the power it will handle, attach the driver with 1" long 10-24 machine screws and T-nuts on the inside.

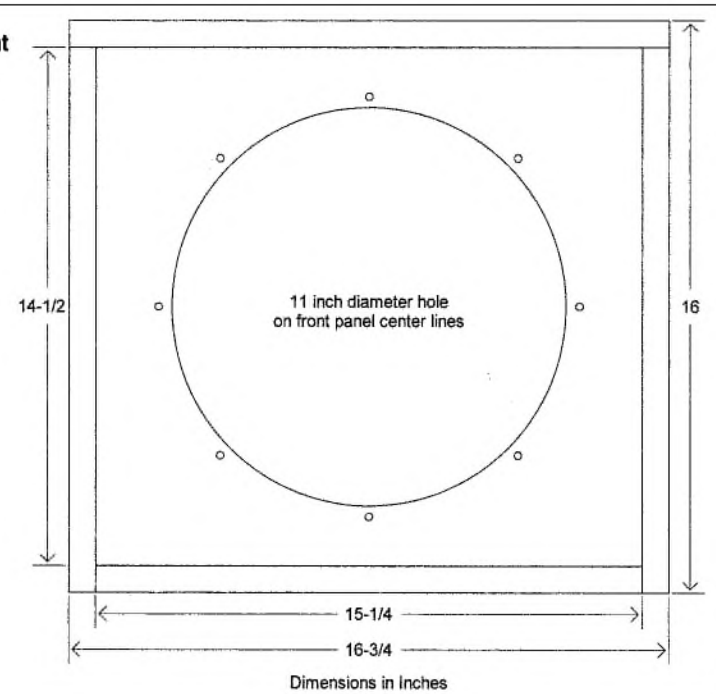
DAMPING PANEL

The damping panel is a ¾" thick panel the same size as the front panel. It will

TABLE 5
MATERIALS LIST FOR ONE ENCLOSURE

QTY.	ITEM
1	4′ × 8′ sheet of top quality ¾″ thick particleboard cut into two 8′ strips each 15¼″ wide plus the remaining strip. Industrial quality particleboard sheets are actually 49″ × 97″.
1	Top quality 1 × 6 board 6′ long
1	Box ELCO 1⅛″ long hardened steel panel nails
1	Tube of silicon rubber sealant
8	1″ long 10-24 machine screws
8	10-24 T-nuts—Do not need one side cut flat for this box
18	1¼″ long #8 flat-head wood screws to mount back
	Few feet of #16 Zip cord or equivalent wire for each voice coil
2	2′ × 4′ sheets of 1″ thick unfaced Owens-Corning #705 Fiberglas® insulation
	Good wood glue. We used Titebond® Original Wood Glue.
1	Dayton #285-185 12″ DVC woofer from Parts Express
1	Piece of acoustically transparent grille cloth about 13″ × 13¼″ for back
1	Piece of acoustically transparent grille cloth about 16″ × 16¾″ for front
1	Optional—Driver mounting kit Parts Express ships with drivers
2	Optional—2-terminal barrier strips to terminate driver wires
	Optional—Round-head wood screws to mount barrier strips
	Optional—Stick-on vinyl to cover box
	Optional—Strip material to dress around grille frame
	Optional—feet or spikes for bottom of the box
1	Optional—Bottle of rubber cement
	Optional—Nominal 2″ thick fiberglass to line DA volume walls

FIGURE 8: Front view of actual enclosure.



contain holes as large as possible while still retaining the damping material with a ¾″ lip and 1″ wide ribs across the center (*Fig. 10*). These holes should have a radius cut on the edges on all of the rear side, which is against the damping material, and a portion of the front (driver) side. We used a router with ⅛″ rounding bit.

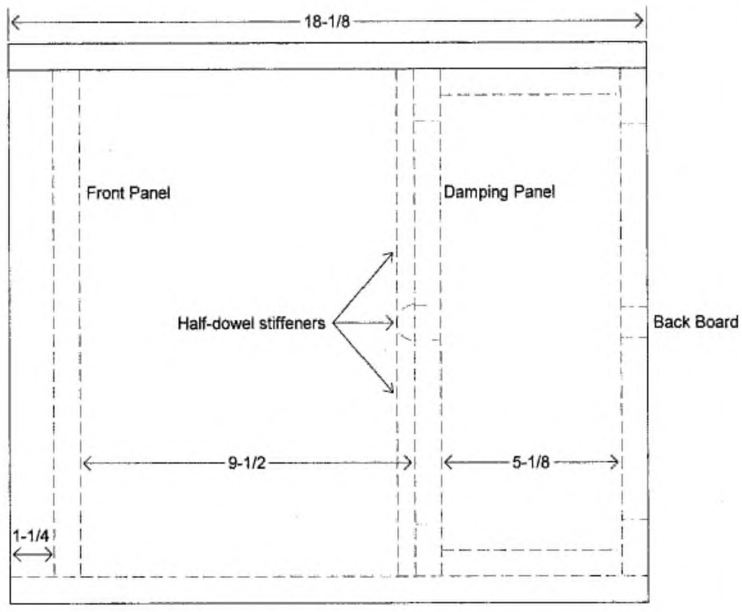
Do not radius the edges of the ribs across the center of the damping panel on the front side. Particleboard is not very stiff or strong when you cut it down to 1″ width.

To stiffen the center ribs in the damping panel, reinforce them on the front side as follows. Cut a section of 1″ diameter dowel about 16″ long down the center lengthwise to produce two semi-circular strips. Glue one of these on as a continuous piece to stiffen the damping panel horizontal rib. (GRK has never been able to split a dowel exactly on center, so we used the thicker "half" to brace the horizontal rib!)

Stiffen the vertical rib by hand-fitting the other half-dowel as two pieces—a necessary compromise. Cut the stiffening strips flush with the edges of the damping panel so they will glue to the sides of the box.

Photo 2 shows the completed damping panel from the front side. Note the two notches for the driver wires on the bottom edge of the panel. A good approach is to clamp the damping panel and back board together and cut these notches into both boards at the same time. We used two pieces of #16 zip cord.

FIGURE 9: Side view of actual enclosure.



BRACING

The front panel and walls of the DA volume should be very stiff for this high-power driver. Note that the stiffening blocks are the last thing you install in the box. If you install them before the damping panel, you will never get the rear of the box assembled properly. Cut eight blocks from scrap ¾″ particleboard and install them between the front panel and the damping panel (*Fig. 11*). Individually cut each block to the proper length to fit firmly between the front and damping panels.

Glue these eight blocks to the front panel very close to the T-nuts mounting the driver (*Fig. 12*). If your blocks cover any portion of the T-nut, file a clearance in that area. *Photos 3* and *4* show the installed blocks. The four corner blocks stiffen the front panel but are of little help for the side walls, which are already "stiff as stone" at the corners.

The four center-of-side blocks mount close to the T-nuts on the front panel, just touching the half-dowel stiffeners on the damping panel. These blocks stiffen both the front panel and the side walls. Their effect is magnified by the fact the damping panel ties opposite box walls together. The box walls will vibrate, but the box will not "balloon" and store excess energy.

The short wall segments behind the damping panel are 1½″ thick but cannot be braced without interfering with the damping material. This has not been a problem, because they are in direct contact with the damping material, are not exposed to the full box pressure, and are stiffened by the damping panel and back board.

TUNNEL FOR DAMPING MATERIAL

To make the tunnel for the damping material, take 1 × 6 (measured ¾″ by 5½″) wood and cut it down to the proper width for five layers of your damping material. Make the tunnel depth so that the back compresses the damping layers about ⅛″ when installed. Our depth was 5⅞″ for our material.

Cut four pieces of 1 × 6 to fit the rear of the box (*Photo 4*). Make the side pieces full height (about 14½″) and cut the top and bottom pieces to fit between them (about 13¾″).

Provision is made on the bottom side of the bottom 1 × 6 to pass the two driver wires (*Photo 2*). This is accomplished by cutting notches to the depth of the wire that line up with the notches in the damping panel and back board. The notches shown in *Photos 2* and *4* will pass a #16 zip cord with ease.

BACK BOARD

The back board is a ¾″ particleboard panel the same size as the front panel. It requires four holes for the open rear area (*Fig. 13*). Both sides of these holes should have about a ⅜″ radius on them. Also note the two notches for the speaker wires.

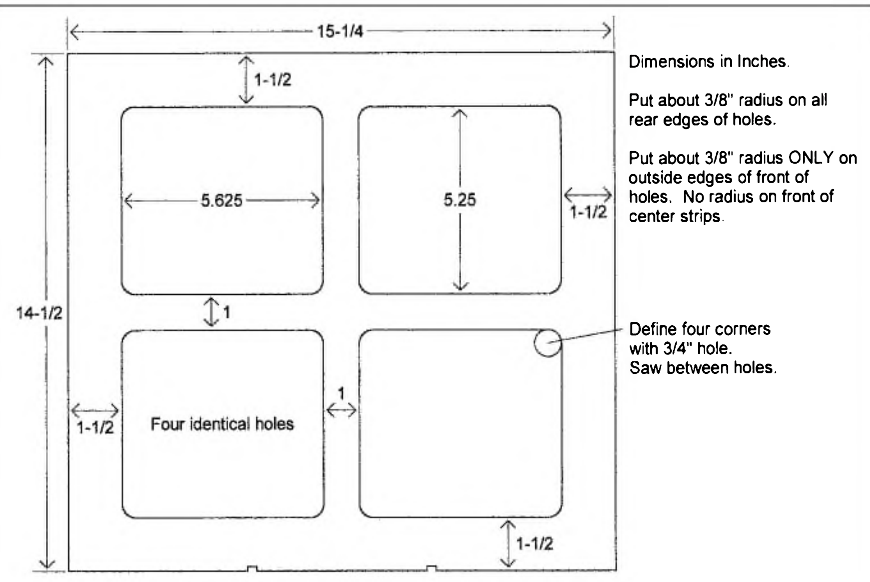


FIGURE 10: Damping panel for IB enclosure.

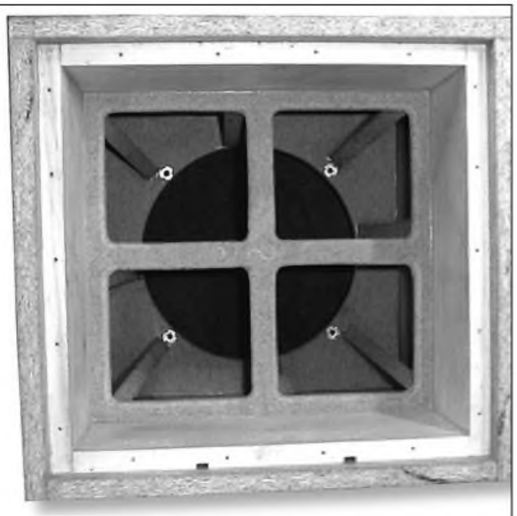


PHOTO 4: Rear view of empty box.

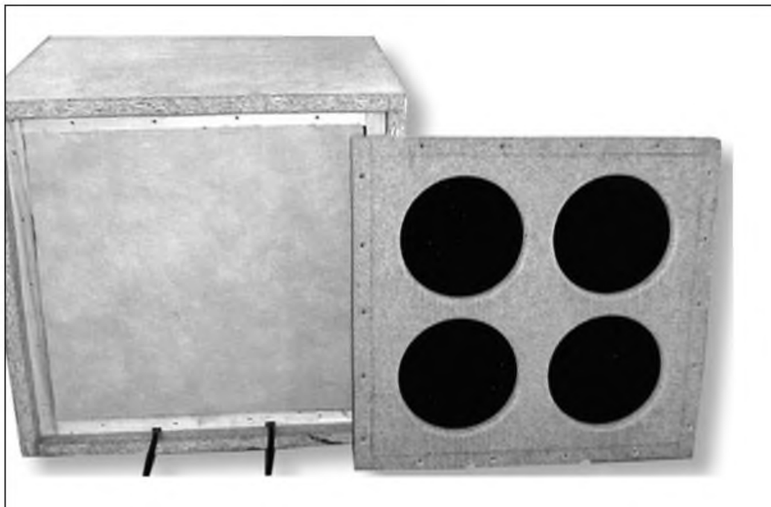


PHOTO 5: Back board and open rear of box showing last damping layer.

The back also has holes for 1¼″ long #8 flat-head wood screws for attachment. Space these screws about every 3″, since the back will tend to move because of the pressure on the damping layers. To keep all of the fiberglass-based damping material inside the box, install grille cloth on the inside of the back board. This should cover only the area pressing against the damping material. *Photo 5* shows the back board and rear of the open box showing the last damping layer.

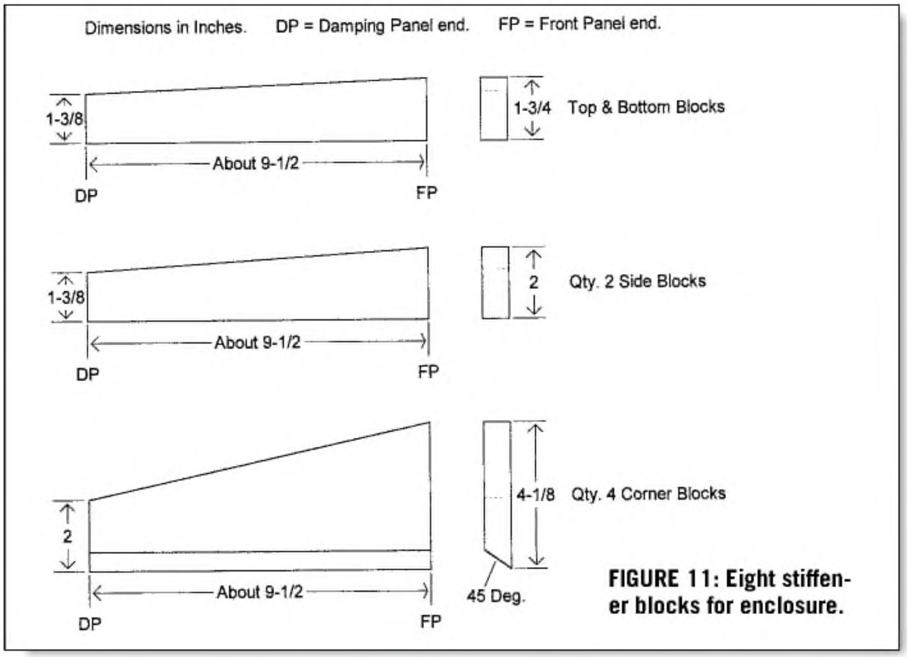


FIGURE 11: Eight stiffener blocks for enclosure.

ASSEMBLY ORDER

Table 5 is a list of materials used in the construction of this subwoofer. The recommended assembly order for the box is as follows:

1. Make all the needed ¾″ particleboard panels as previously defined.
2. Mark the bottom for nails to attach the front panel. Avoid putting a nail on the front-panel centerline where it might hit the T-nut. We also recommend that you try to keep nails back at least 1⅛″ from the end of a panel. Mark the side boards for nails to attach them to the bottom and front panel, again avoiding the driver centerline. Drill nail holes with a #51 drill (0.067″) in these panels. Use no nails in mounting the damping panel.
3. Attach the front panel to the bottom, being sure the front panel stands vertical and is set back a constant 1¼″. Use lots of glue and 1⅛″ hardened steel nails (*Table 5*). Let the glue dry on this assembly before continuing.
4. Attach the sides to the front-panel/bottom assembly, being sure the

internal box height stays constant. Wipe the glue off the inside of the joints so the damping panel and tunnel boards will fit properly. All nails in the sides should be "set" below flush. Again, let the glue dry.

5. Mark the top for nails into the front panel and sides and drill holes. Attach the top and again wipe off excess glue on the inside. If you made the top extra wide, you should later flush-cut it to fit the box. Again, set the nails in the top below the surface and allow the glue to dry before continuing.

6. Cut the four 1 × 6 tunnel boards to fit the box and remember the wire notches on the bottom of the bottom board.

7. Mark the box for the back edge of the tunnel boards in ¾″ from the rear of the box. It does not matter whether your tunnel boards are slightly different in width from our 5⅞″ value, because slight variation in DA volume is OK.

8. Slide the damping panel well into the box and install the four tunnel boards with lots of glue and clamp them into position. Now glue the front edges of the tunnel boards and the box walls in this region and slide the damping panel up against the tunnel boards and clamp it there until the glue dries. You should clear the wire notches of any glue so you can later install the wires.

9. Cut to fit and install the eight stiffener blocks. With proper fit, simply gluing them in place is sufficient.

10. To assure an airtight box and no rattles, fillet all joints. We find white/yellow glue ideal for this, because it "sucks" into any air gaps. Continue this process until the glue stays on the surface to form a fillet. This is a multi-day process, which you should not skip.

11. Test-fit the back board and drill the necessary holes for the screws to mount the back to the rear edges of the tunnel boards. Remember the grille cloth and wire notches on the back board.

12. Now fill the nail holes on the top and sides and sand and finish the box as you desire. This construction approach allows you to simply paint the box or wrap it in stick-on vinyl. You will later staple the grille cloth directly to the front lip of the box and then apply thin trim strips around the outside of the grille area. These strips can be made from half-round or similar material and painted or covered with the same stick-on vinyl. We skipped putting a finish on the box at this time.

13. Now install the two speaker wires using silicon rubber to seal them at the damping panel notch. This driver has press-and-insert terminals, so you only need to strip and tin with solder the driver end of each wire. Keep track of polarity. If you drive the two voice coils out-of-phase, you will get absolutely no sound out of the box.

We unfortunately confirmed this result once while working on the 15″ unit! We covered the wires with silicon rubber over most of the tunnel board depth making them likely "permanent." We thus ohmed each wire to assure that it was good and properly marked for polarity before installation.

14. This is a good time to put fiberglass on the walls of the dead-air section if you are going to do so. Read the section entitled "System Frequency Response" in Part 2 for help on a decision here and on details of the fiberglass batts used. You can add the batts later as we did.

FIGURE 12: Placement of eight stiffener blocks.

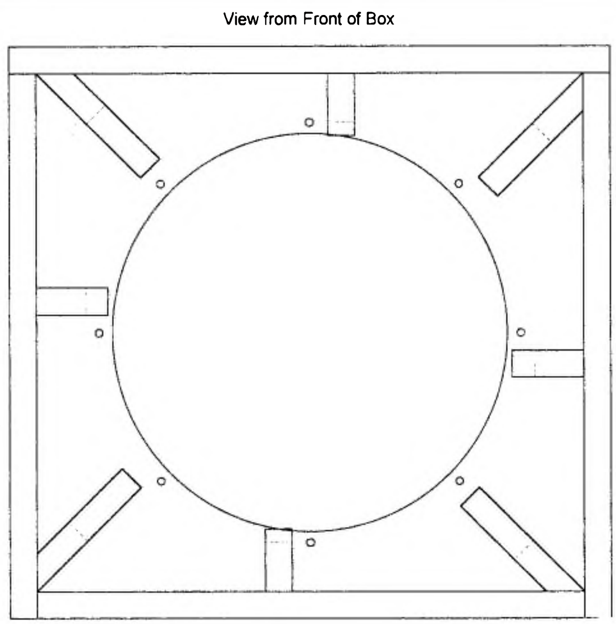
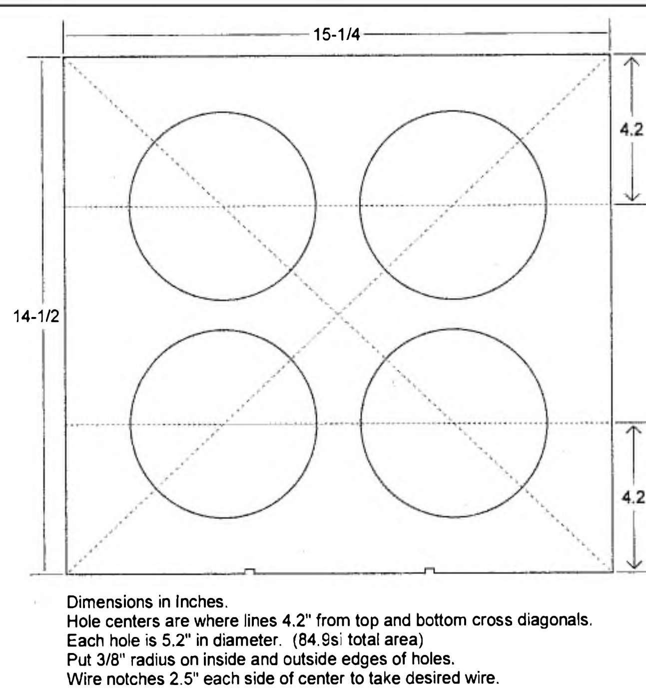


FIGURE 13: Back board layout.



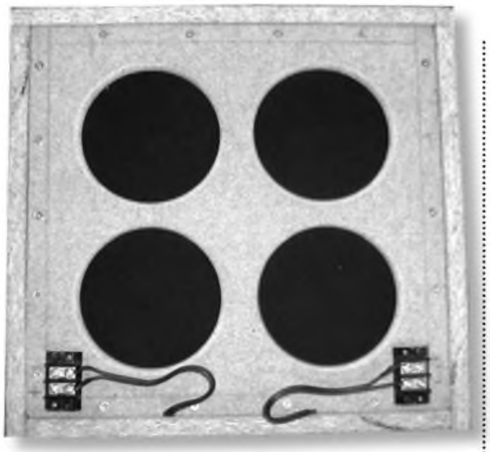


PHOTO 6: Rear view of completed subwoofer.

15. Now cut the five damping layers (about 13″ by 13¾″) and slide them carefully into the tunnel and install the back board. It will take two 2′ × 4′ sheets of 1″ thick Owens-Corning #705 damping material to cut out five layers this size. You can mark this material with a dull #2 pencil and cut it easily with a serrated bread knife.

Make the layers fit snugly, but any attempt to force in an oversized piece will result in the layer breaking up around the edges. We normally face the rougher surface of the layers toward the driver, but have done no testing to detect direction sensitivity. Remember this is a fiberglass-based product, so take the appropriate safety precautions when working with it. *Photo 5* shows the rear of the installed damping layers.

16. Install the back board to compress the damping layers. The back is installed "dry" without any sealing material. If desired, add barrier strips on the back to terminate the four speaker wires. Again, keep track of the polarity for each voice coil. *Photo 6* shows the rear view of the completed subwoofer.

17. Install the driver and you are ready to play. To maintain the maximum stiffness at the driver/box interface, we installed the driver "dry." Pushing the cone quickly showed a greatly increased stiffness, meaning a good seal. If you don't get a good seal dry, you may need to use the gasket or rope putty that Parts Express ships with the driver. You want this driver bolted down as solidly as possible.

18. When you are happy with the subwoofer, you can add the grille cloth and front trim strips. This driver moves lots of air, so use very acoustically transparent grille cloth.

This concludes construction of the subwoofer. Now it's time to play it. In Part 2 we provide test and listening results, application information, and discussion of the 15″ driver version. ❖

SOURCES

Parts Express
725 Pleasant Valley Dr.
Springboro, OH 45066-1158
937-743-3000
FAX: 937-743-1677
Website: www.partsexpress.com

Owens-Corning
800-328-7617

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Load-Invariant Power Amplifiers

Part 2

This noted author continues his groundbreaking work in the area of distortion reduction. **By Douglas Self**

You can reduce the extra nonlinearity engendered by sub-8Ω loads by using two or more output devices in parallel, even though this is quite unnecessary for handling the power output required. If two output devices are connected in parallel, the collector current divides in two between them, and beta-droop due to increasing collector current is much reduced.

From the SPICE evidence in Part 1, I predicted that paralleling devices ought to reduce large-signal nonlinearity (LSN), and in real life, indeed it does. You must never neglect this sort of reality-check when you are using simulators.

Figure 6 compares 4Ω THD at 60W for single and double output devices, showing that doubling reduces the distortion by about 1.9 times, which is well worthwhile. The output transistors were contemporary power devices, in this case Motorola MJ15024/15025.

The historical 2N3055/2955 complementary pair gives a similar halving of LSN on being doubled, though the initial distortion is three times higher into 4Ω. 2N3055s with an H suffix (for hometaxial?) are markedly worse than those without such a mark. No current-sharing precautions were taken when doubling the devices. This omission seemed to have no effect on LSN reduction, and there was no evidence of current-hogging.

Multiplying the power devices naturally increases the power output capability, though if this is exploited, LSN will tend to rise again, and you will be back where you started. You will also need to uprate the power supply, heatsink, and so on.

The essence of this technique is to use parallel devices to reduce distortion rather than to increase power handling. You could argue that multiplying output transistors is an expensive way to solve a linearity problem. To give this perspective, in a typical stereo power amplifier, with the cost of heatsink, metal work, and mains transformer included, doubling the output devices will increase the total by only about 5%.

BETTER OUTPUT DEVICES

Knowing that LSN is a device-dependent effect, you can look around for transistor types that perform well in this respect. The 2SC3281, 2SA13 02 TO3P complementary pair has a reputation in the hi-fi world for being "more linear" than the run of transistors. This is the sort of vague claim that arouses the deepest of suspicions, not least because the Vbe/Ic relationship of a bipolar transistor is about as negotiable as the value of pi. Also bear in mind the many assertions of superior linearity in power FETs, which is the exact opposite of reality.[8]

In this case, however, the kernel of truth is that the 2SC3281 and 2SA1302 show not only higher beta but also much less beta-droop than average power transistors. Toshiba introduced these devices; Motorola versions are numbered MJL3281A, MJL1302A, and

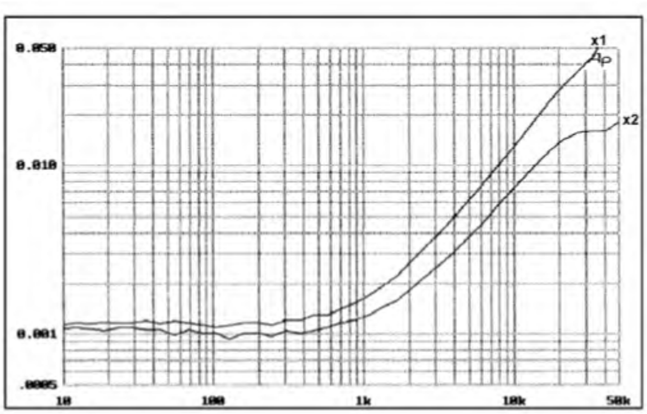

FIGURE 6: 4Ω distortion is reduced by 1.9× upon doubling standard (MJ15024/15025) output transistors. 30W/8Ω.

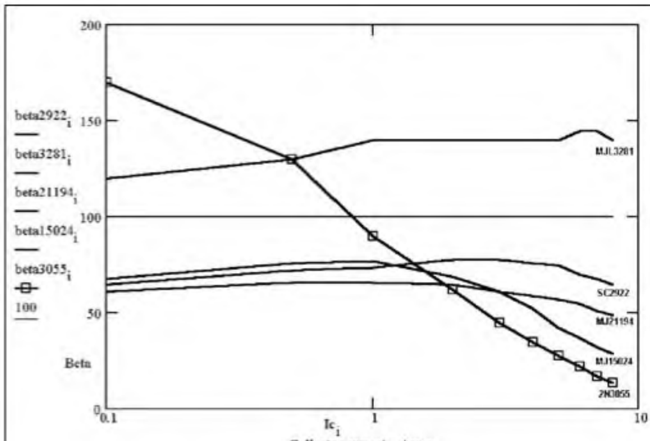

FIGURE 7: Power transistor beta falls as collector current increases. Data points shown on 3055 curve (from manufacturers' data sheets).

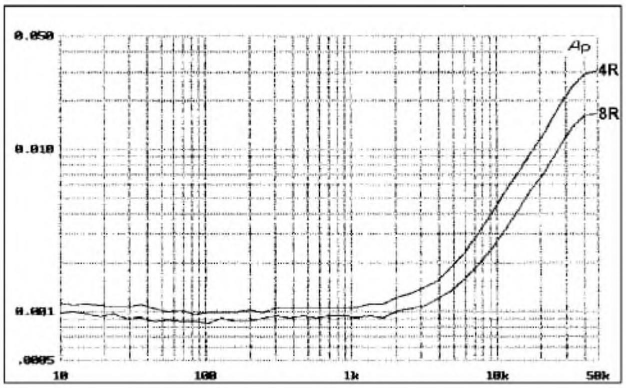

FIGURE 8: THD at 40W/8Ω and 80W/4Ω with single 3281/1302 devices.

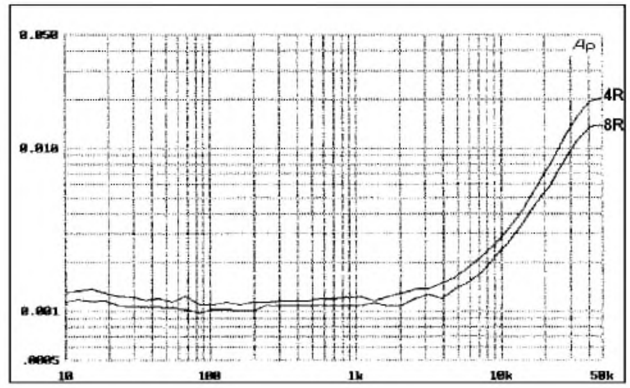

FIGURE 9: THD at 40W/8Ω and 80W/4Ω with doubled 3281/1302 output transistors. 4Ω THD has been halved compared with *Fig. 8*.

also come in a TO3P package. *Figure 7* shows beta-droop, for the various devices discussed here, and looking back at the THD plots, it is clear that more beta-droop means more LSN. There seems to be no special name for this class of BJTs, so I have called them "sustained-beta" devices.

The THD into 4 and 8Ω for single 3281/1302 devices is shown in *Fig. 8*. Distortion is reduced by about 1.4 times compared with standard devices. (See Part 1.) Several pairs of 3281/1302 have been tested and the 4Ω improvement is consistent and repeatable.

The obvious next step is to combine the two techniques by using multiple sustained-beta devices. Doubled 3281/1302 device results are shown in *Fig. 9*, where the distortion at 80W/4Ω (15kHz) is reduced from 0.009% in *Fig. 8* to 0.0045%; in other words, we are using better transistors, but doubling the number in parallel still halves the distortion. The 8 and 4Ω traces are now very close, the 4Ω THD being only 1.2 times higher than the 8Ω case. This indicates that the intrusion of LSN has been much reduced.

Figure 7 shows that the 2SC3281 and 2SA1302 pair seems to be in a class of its own. The 2N3055 starts off with high beta, but it collapses completely at high Ics. There are also devices that show beta-maintenance intermediate between the "super" 3281/1302 and "ordinary" MJ15024/25, such as MJ21193, MJ21194 (TO3), and MJL21193, MJL21194 (TO3P), also from Motorola. Other examples are the 2SC2922/2SA1216 from Sanken in an MT200 package. It seemed likely that they would give less LSN than ordinary power devices, but more than the 3281/1302. This prediction was happily fulfilled.

LOADS BELOW 4Ω

So far we have concentrated on 4Ω loads as opposed to 8Ω. Loudspeaker impedances can and do sink a good deal lower than this, so I pursued the matter down to 3Ω. One pair of 3281/1302 devices will give 50W into 3Ω for a THD of 0.006% (10kHz, *Fig. 10*). Two pairs of 3281/1302 once more halve this to 0.003% (10kHz, *Fig. 11*). This is a very good result for such simple circuitry, and may be something of a record for 3Ω linearity.

At this point it seems that whatever the device type, doubling the outputs halves the THD percentage for 4Ω loading. The multiple-device principle can be extended down to 2Ω operation, but triple devices will be required for sustained operation into this impedance. Resistive losses in the circuitry can be serious, and often 2Ω power output may not be much greater than that into 4Ω.

BETTER 8Ω PERFORMANCE

It was wholly unexpected that the sustained-beta devices would also show lower crossover distortion at 8Ω, but they do, and the effect is again repeatable. I don't have an explanation. It could be that whatever improves the beta characteristics also subtly alters the turn-on law so crossover distortion

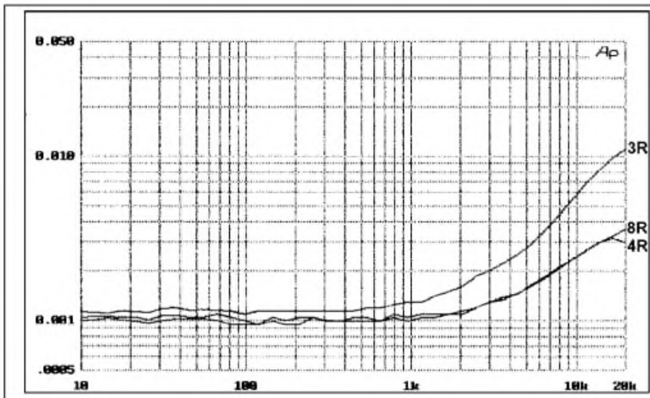

FIGURE 10: Distortion for 3, 4, and 8Ω loads, single 3281/1302 devices. 20W/8Ω, 40W/4Ω, and 60W/3Ω.

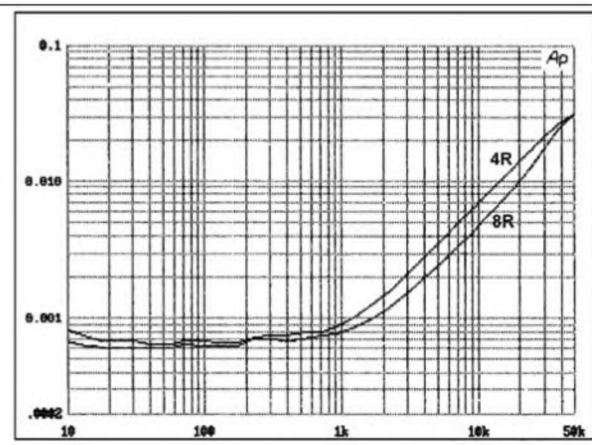

FIGURE 13: Measured THD for a triple-EF output stage with N=1 50W/8Ω.

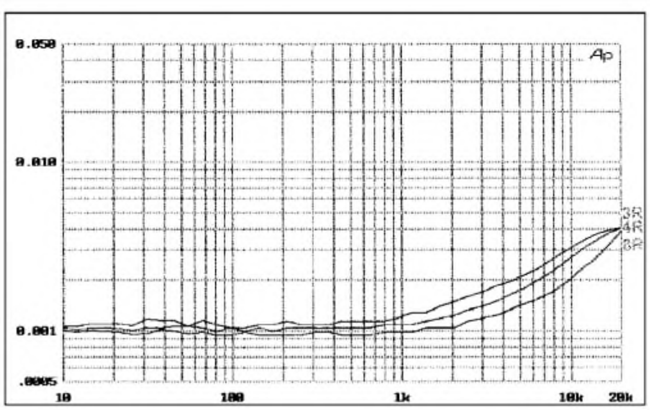

FIGURE 11: Distortion for 3, 4, and 8Ω load, double 3281/1302 devices. Power as *Fig. 10*.

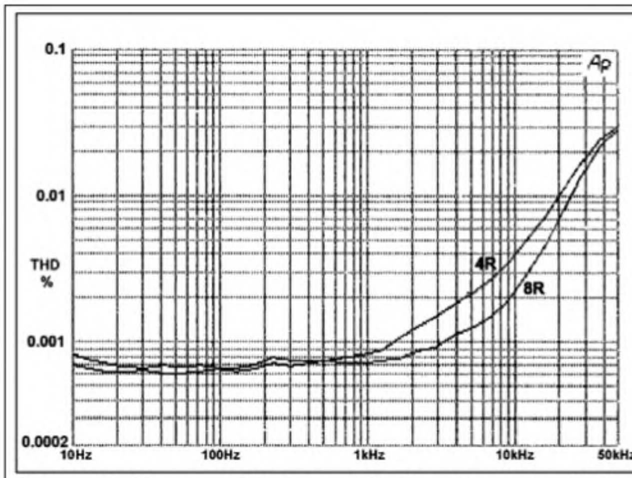

FIGURE 14: Measured THD for a triple-EF output stage with N=2.

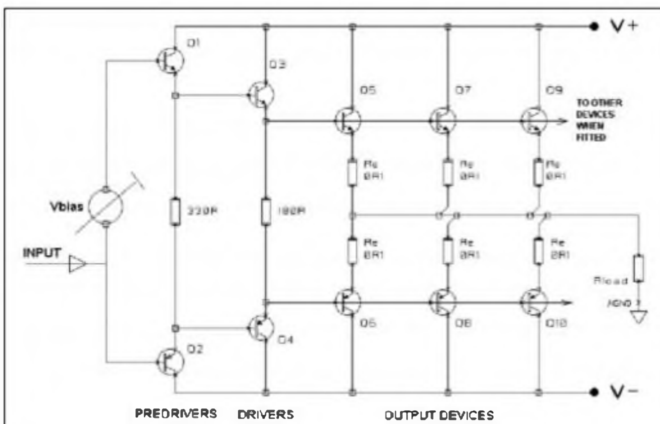

FIGURE 12: The triple-EF output stage, using multiple output devices with individual emitter resistors. Shown for N=3; i.e., three output devices.

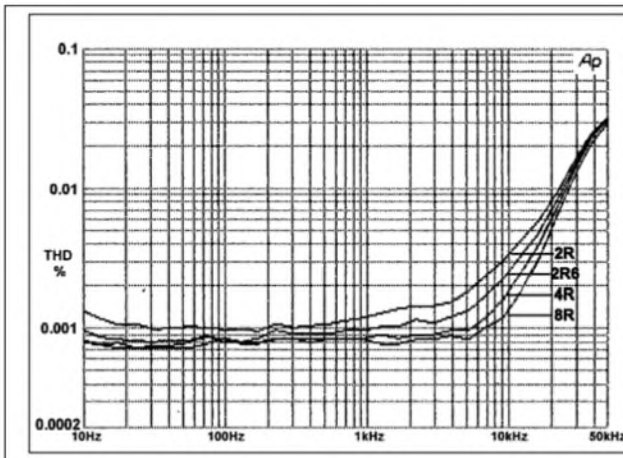

FIGURE 15: Measured THD for a triple-EF output stage with N=7.

is reduced; alternatively, traces of LSN, not visible in the THD residual, may have been eliminated.

Figure 11 shows the improvement over the MJ15024/ 25 pair; the 8Ω THD at 10kHz is reduced from .003% to .002%, and with correct bias adjustment, crossover artifacts are simply not visible on the 1kHz THD residual. In fact, crossover artifacts are only just visible in 4Ω testing, and to get a feel for the distortion being produced, and to set the bias optimally, it is necessary to test at 5kHz into 4Ω. There is less negative feedback at this frequency, and so there is enough distortion to measure with reasonable accuracy.

TROUBLE WITH TRIPLES

Electronics sometimes offers a choice between applying brawn (i.e., multiple power devices) or brains to solve a given problem. The "brains" option here would be represented by a clever circuit configuration that gave the same results without replication of expensive power silicon. The use of local negative feedback is an obvious possibility; however, the CFP configuration examined previously has just that in place, with feedback wrapped around the output device, and it certainly doesn't eliminate the problem.

Another possibility is the use of one of the various output-triple topologies that have occasionally been used. Note that the term "output-triples" here refers to pre-driver, driver, and output device all in a local negative-feedback (NFB) loop, rather than three identical output devices in parallel, which I would call "tripled outputs." Keeping the terms straight is a bit of a problem here.

In simulation, output-triple configurations do indeed reduce the gain-droop that causes LSN. There are many different ways to configure output-triples, and they vary greatly in their general linearity and effectiveness at minimizing LSN. The difficulty with this approach is that three transistors in a local loop are very prone to parasitics and local oscillations. This is worsened by sub-8Ω load impedances, presumably because the higher collector currents lead to increased device transconductance.

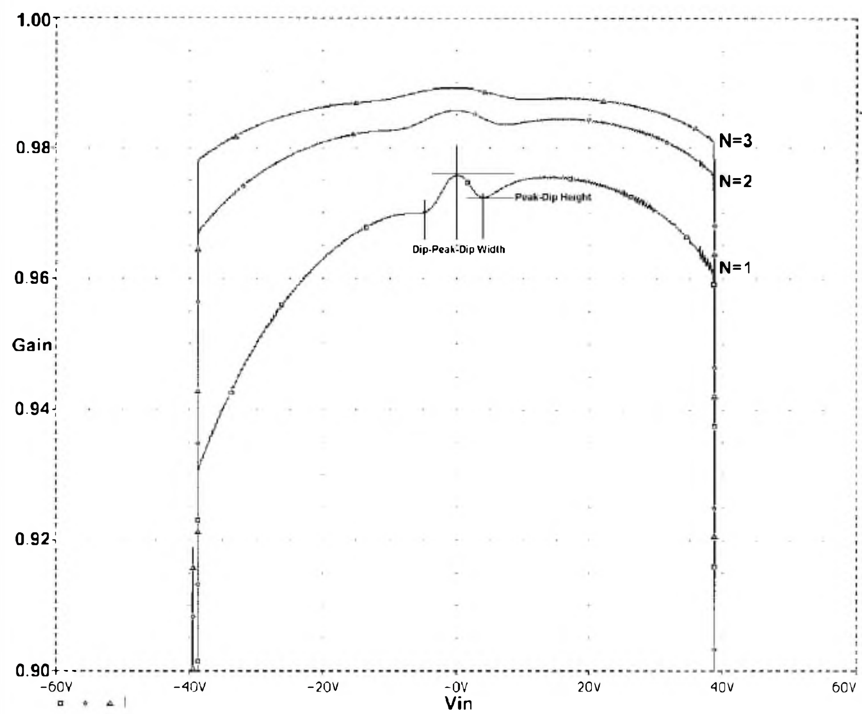

FIGURE 16: Simulation showing how as more devices are used the central gain-wobble that causes crossover distortion becomes flatter and spread over a wider range of output voltage.

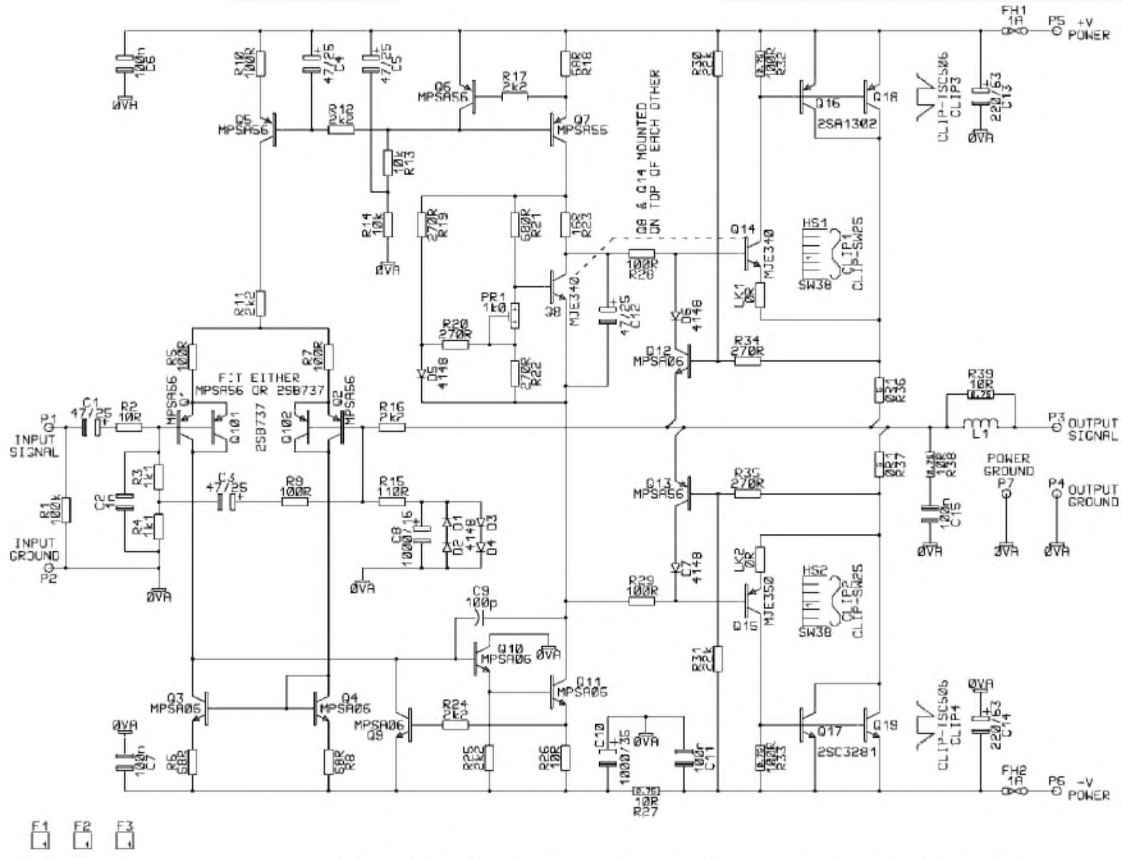

FIGURE 17: Circuit diagram of a practical load-invariant power amplifier.

This sort of local HF instability can be very hard to deal with, and in some configurations appears to be virtually insoluble.

One configuration that can be stabilized is the triple-emitter-follower (EF, *Fig. 12*). More on this later.

THE LATEST FINDINGS

I have recently experimented further with multiple 2SC2922/2SA1612 devices, using 3, 4, 5, and 6 in parallel. In this case the circuit I used was somewhat different (*Fig. 12*). With a greater number of devices, I was now more concerned about proper current sharing, and so each device has its own emitter resistor. This makes it look much more like a conventional paralleled output stage, which essentially it is. This time I tried both the double and triple-EF output configurations, as I wished to prove:

1) that LSN-theory worked for both of the common configurations EF and CFP. It does.
2) that LSN-theory worked for both double and triple versions of the EF output stage. It does.

For reasons of space, I show only the triple-EF results here.

Figure 13 shows the measured THD results for one complementary pair of output devices in the triple-EF circuit of *Fig. 12*. Distortion is slightly higher, and the noise floor relatively lower, than in the standard result (Fig. 2 in Part 1) because of the higher output power of 50W/8Ω.

Figure 14 shows the same except there are now two pairs of output devices. Note that THD has halved at both 8 and 4Ω loads; this is probably due to the larger currents taken by 8Ω loads at this higher power. *Figure 15* shows the result for six devices; 8Ω distortion has almost been abolished, and the 4Ω result is almost as good. It is necessary to go down to a 2Ω load to get the THD clear of the noise so it can be measured accurately. With six outputs, driving a substantial amount of power into this load is not a problem.

On a practical note, the more output devices you have, the harder it may be to purge the amplifier of parasitic oscillations in the output stage. This is presumably due to the extra raw transconductance available, and can be a problem even with the triple-EF circuit, which has no local NFB loops. I don't pretend to be able to give a detailed explanation of this effect at the moment.

Having demonstrated that sustained-beta output devices not only reduce LSN but also unexpectedly reduce crossover distortion, it seemed worth checking to see whether using multiple output devices would give a similar reduction at light loading. I was rather surprised to find they did.

Adding more output devices in parallel, while driving an 8Ω load, results in a steady reduction in distortion. *Figure 15* shows how this works in reality. The SPICE simulations in *Fig. 16* reveal that increasing the number of output devices (N) not only flattens the crossover gain wobble, but also spreads it out over a greater width. This spreading effect is good because it means that lower-order harmonics are generated, and at lower frequencies there will be more negative feedback to linearize them. (Bear in mind here that a triple-EF output has an inherently wider gain wobble than the double-EF.)

Taking the gain wobble width as the voltage between the bottoms of the two dips, this appears to be proportional to N. The amount of gain wobble—measured from the top of the peak to the bottom of the dips—appears to be proportional to 1/N.

This makes sense. You know from Fig. 1 (Part 1) that crossover distortion increases with heavier loading; i.e., with greater currents flowing in the output devices, but under the same voltage conditions. It is therefore not surprising that reducing the device currents by using multiple devices has the same effect as reducing loading. If there are two output devices in parallel, each sees half the current variations, and crossover nonlinearity is reduced. The voltage conditions are the same in each half and so are unchanged.

This insight offers the interesting possibility that crossover distortion—which has hitherto appeared inescapable—can be reduced to an arbitrary level simply by paralleling enough output transistors. To the best of my knowledge, this is a new insight.

A PRACTICAL LOAD-INVARIANT DESIGN

Figure 17 shows the circuit of a practical load-invariant amplifier intended for 8Ω nominal loads with 4Ω impedance dips. Its distortion performance is shown in *Figs. 6–11*, depending on the output devices fitted. A PCB has been prepared which allows the immediate easy installation of either single or double TO3P devices. Alternatively, wires may be taken off to TO3 devices or further TO3Ps. The supply voltage can be from ±20 to ±40V; I leave checking power capability for a given output device fit to the constructor.

Apart from load-invariance, the design also incorporates two new techniques of mine. The first one greatly reduces time-lag in the thermal compensation. With a CFP output stage, the bias generator aims to shadow driver junction temperature rather than the outputs. A much faster response to power dissipation changes is obtained by mounting the bias generator transistor TR8 on top of driver TR14, rather than on the other side of the driver heatsink. The driver heatsink mass is thus largely decoupled from the thermal compensation system, speeding up the response by two orders of magnitude.[9]

The second new technique is the use of a bias generator with an increased temperature coefficient, to reduce the static errors introduced by thermal losses between the driver and the sensor. The tempco is increased to –4.0mV/°C.[10] D5 also compensates for the effect of ambient temperature changes.

The design is not described in detail because much of it closely follows the Blameless Class-B amp described in references 12 and 13. Some features are derived from the Trimodal amplifier,[9] notably the low-noise feedback network (with its requirement for input bootstrapping if a 10k input impedance is required). Single-slope VI limiting is incorporated for overload protection (see TR12, 13). As usual in my Blameless amplifiers, the global NFB factor is a modest 30dB at 20kHz.

The input stage (current-source Q5 and differential pair Q1, 2) is heavily degenerated by R5 and R73 to delay the onset of third-harmonic Distortion 1. The contribution of transistor internal regarding variation is minimized by using the unusually high tail current of 6mA. Q3, 4 form a degenerated current-mirror that enforces accurate balance of the Q1, 2 collector currents, preventing the input stage from generating second-harmonic distortion—something that should never happen.

The input resistance (R3+R4) and feedback resistance R16 are made equal and kept as low as possible consistent with a reasonable value, so that base current mismatch caused by beta variations will give a minimal DC offset; this does not affect Q1-Q2 Vbe mismatches, which appear directly at the output, but these are much smaller than the effects of Ib. Even if Q1, 2 are high-voltage types with low beta, the output offset should be within ±50mV, which I think is quite adequate. This eliminates balance presets and DC servos. A low value for R8 means a low value for R15 to maintain gain, and this improves noise performance.

However, the low value of R3+R4 at 2k2 makes an input impedance that is inconveniently low. You want a low DC resistance, but a high AC resistance here; in other words, it's time to use the fine old practice of bootstrapping. The signal at Q2 base is almost exactly the same as the input, so the mid-point of R3 and R4 is driven by C3. I initially had grave doubts about the HF stability of this arrangement, and added isolating resistor R9, which limits the bootstrap factor, but in any event I have had no trouble, and no reports of any from the other constructors of this design. With R9 = 100R, the AC input resistance is raised to 13k.

The value of C8 shown (1000µF) gives a LF rolloff with R15 that is –3dB at 1.4Hz. The aim is not unreasonably extended sub-bass, but the prevention of an LF rise in distortion due to capacitor nonlinearity. 100µF used here degraded the 10Hz THD from <0.0006% to 0.0011%, and I judge this unacceptable aesthetically if not audibly. LF bandlimiting should be done earlier, with non-electrolytic capacitors. Protection

REFERENCES

Please note: The following references were originally to articles I published in the British journal *Electronics World* (*Wireless World*). Since USA readers are likely to have difficulty accessing back copies of this, I have changed them to refer to my books *Audio Power Amplifier Design Handbook* (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, custserv@audioXpress.com) and *Self on Audio*, which gather together all these articles and many more besides.

8. Self, D., "FETs vs BJTs—The Linearity Competition," *Self on Audio*, Newnes, ISBN 0-7506-4765-5, p. 116. The collected works from 25 years of *Wireless/Electronics World*. Amplifier classification, preamps, power amps Class-A, and so forth. The poor linearity of power FETs.

9. Self, D., "Trimodal Audio Power: Part I," *Self on Audio*, Newnes, p. 263. Reduction of THD in Class-AB by low res.

10. Self, D., "Thermal Dynamics in Audio Power: Part II," *Audio Power Amplifier Design Handbook* 2nd Edition, Newnes ISBN 0-7506-2788-3, p. 292. Reduction of sensor delay by mounting on driver.

11. Self, D., "Thermal Dynamics in Audio Power: Part III," *Audio Power Amplifier Design Handbook* 2nd Edition, Newnes, p. 316. Bias generator with increased tempco and ambient compensation.

12. Self, D., "Distortion in Power Amplifiers: Part 6," *Self on Audio*, Newnes, p. 202. The Blameless amplifier concept.

13. Self, D., "Distortion in Audio Power Amps, Parts 1, 2, and 3," *Audio Electronics* 2, 3, and 4/99.

14. Self, D., "Distortion in Power Amplifiers: Part 8," *Self on Audio*, Newnes, p. 231. A Class-A power amplifier.

diodes D1–4 prevent damage to C2 if the amplifier suffers a fault that makes it saturate in either direction; it looks like a funny place to put diodes but adds no detectable distortion.

The VAS stage Q11 is enhanced by an emitter-follower Q10 inside the Miller-compensation loop, so that the local NFB that linearizes the VAS is increased by augmenting total VAS beta. This extra local NFB effectively eliminates Distortion 2 (VAS nonlinearity). Further study has shown that thus increasing VAS beta gives a much lower collector impedance than a cascode stage, due to the greater local feedback, and so a VAS-buffer to eliminate Distortion 4 (loading of VAS collector by the nonlinear input impedance of the output stage) appears unnecessary.

Cdom is relatively high at 100pF, to swamp transistor internal capacitances and circuit strays, and make the design predictable. The slew rate calculates as 40V/μs. The VAS collector-load Q7 is a standard current source, to avoid the uncertainties of bootstrapping.

Since almost all the THD from a Blameless amplifier is crossover, keeping the quiescent conditions optimal is essential. Quiescent stability requires the bias generator to cancel out the Vbe variations of four junctions in series—those of two drivers and two output devices. Bias generator Q8 is the standard Vbe-multiplier and should be in contact with the top of one of the output devices, not the heatsink, as this is the fastest and least attenuated source for thermal feedback.

The output stage is a standard complementary-feedback pair (CFP). C15, R38 are the Zobel network, while L1, damped by R39, isolates the amplifier from load capacitance.

A PCB for this amplifier has been made available by The Signal Transfer Company (www.signaltransfer.freeuk.com). I can vouch that it works properly.

A POINT OF DEPARTURE

The improvements described here fit neatly into the philosophy of Blameless power amplifiers. The fundamental principle of the Blameless concept is that Distortion 3 should be the only significant distortion remaining, because Numbers 1, 2, and 4 to 8 can all be reduced to negligible levels in straightforward ways. For 8Ω operation, the main nonlinearity left is crossover distortion.

As hoped, the concept of a Blameless power amplifier is proving extremely useful as a defined point of departure for new amplifier techniques. Starting from the standard Blameless Class-B amplifier, I have derived:

1. A pure Class-A power amplifier.[14]
2. A Trimodal Class-A/AB/B amplifier.[9]
3. The load-invariant Class-B amplifier described here.
4. An efficient and low-THD Class-G amplifier (so far unpublished).

Note that 2 and 3 are simple add-ons to the basic Blameless Class-B configuration. The former adds a Class-A biasing subsystem, while the latter grafts on extra (or better) output devices. Number 4 uses a rather different output stage, but the rest of the circuitry is unchanged.

I hope to publish details of some of these designs here in the future.

CONCLUSIONS

My initial thoughts were that an amplifier could be considered as load-invariant if the rise in THD from 8Ω to 4 was less than some given ratio. For normal amplifiers the THD increase factor is from two to three times. The actual figure attained by the amplifier presented here is 1.2 times, and I for one am prepared to classify this as load-invariant. The ratio could probably be made even closer to unity by tripling the outputs.

Remember, this amplifier is designed for 8Ω nominal loads and their accompanying impedance dips; it is not intended for speakers that start out at 4Ω nominal and plummet from there. To deal with that you would need twice as many output devices. Nonetheless, I hope that this article has provided some progress towards load-invariance, and that power amplifier design might have taken another small step forward. ❖

A Trapezoidal Closed 3-Way

The author premiered this unusual three-way design at the Moscow high-end show and received more than rave reviews for his work.

By S. Batti

The annual "Hi End of Russia" show takes place in Moscow and is organized by the Moscow Technical University of Communication and Informatics. Most of the show's participants are small companies from all over Russia and former USSR republics. Hi-end enthusiasts and amateurs also have an opportunity to demonstrate their devices at this show. However, they are not allowed to demonstrate the sound of their systems on the main floor; instead, each participant is allotted a special time to demonstrate in a professionally designed listening room.

I personally participated in this show in 1999, 2000, and 2001. This article describes the loudspeaker system that I designed for demonstration at the 1999 show.

DRIVER SELECTION

On the one side of the coin, I required a loudspeaker with a powerful bass for the 500ft^2 exhibition's listening room; on the other side I wanted to use it in my home 180ft^2 listening room (*Photo 1*), so I didn't want to produce a huge box. Moreover, I prefer the sound of a sealed box with an f_3 of 0Hz compared to the sound of a vented one with an f_3 of 30Hz. After spending much time looking at the available compromises, I finally decided to design and make a 3-way system with a 10″ woofer in a sealed cabinet. But the 10″ bass driver for a small cabinet was the main problem: I couldn't find the right one in Russia. My friend Artem Blagodarsky, who worked in the Russian-American joint venture, helped me. He agreed to buy the drivers in the US and provided me with the Madisound catalog to choose from.

I chose the Madisound 10204 DVC woofer, which solved my problem. Preliminary calculations showed f_3 = 50Hz in a sealed 30 ltr box. The cone excursion 6mm provides 100dB SPL at 1m

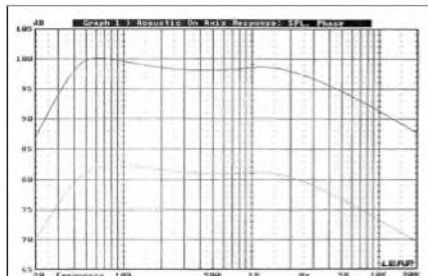

FIGURE 1: Low-frequency SPL simulation of the 10204 DVC in a 30 ltr sealed box at 1 and 40W.

PHOTO 1: The loudspeakers in author's listening room.

distances in low-frequency range. These figures confirm the satisfactory bass parameters for my project.

Figures 1 and *2* show the results of preliminary calculations. After that I selected the Vifa P13 00 08 midrange and the Morel MDT-29 tweeter. I believed that these relatively inexpensive drivers were suitable for a good-sounding loudspeaker.

CABINETS

I thought that decreasing the square of the front panel around the drivers would improve the loudspeaker's dispersion for good soundstage. I chose the trapezoidal shape for the front panel (*Photo 2*). I decided to make two cabinets for each loudspeaker (right and left channels) to decrease the woofer vibration influence upon the mid and tweeter. The loudspeaker's cabinet consists of two parts: the bass cabinet and mid-high cabinet. *Figure 3* shows the construction plan of the bass cabinet. *Figure 4* shows the construction plan of the mid-high cabinet. *Photo 3* shows the rear of the completed unit.

I used 18mm thick plywood to make the cabinet. The internal surfaces of both cabinets are lined with 4mm-thick bituminous felt. The bass cabinets are filled 100% with Dacron low-density sheet wadding. The mid-high cabinets are filled 80% with fiberglass insulation.

DRIVER MEASUREMENTS

With the cabinets assembled, I installed the drivers and began to measure the drivers' parameters, which were necessary for crossover design (*Photo 4*). I used LMS to measure all necessary data. *Figure 5* shows the drivers' impedance curves measured with constant current. *Figure 6* shows the drivers' SPL response. The woofer curve is a scaled –6dB ground plane measurement. The midrange curve at 400Hz is a combined scaled –6dB ground plane measurement and gated measurement. The tweeter curve is a gated measurement.

PHOTO 2: Front view of loudspeaker without grille.

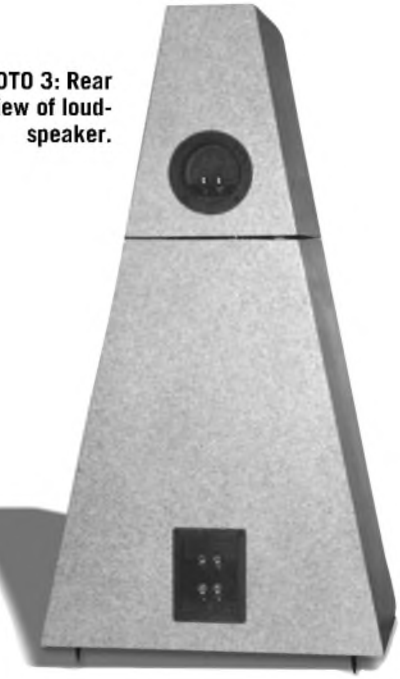

PHOTO 3: Rear view of loudspeaker.

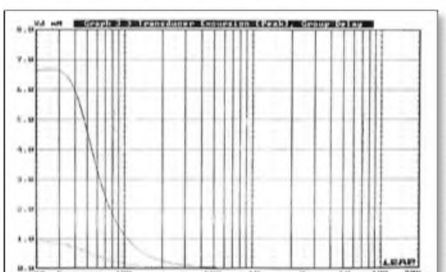

FIGURE 2: Low-frequency cone excursion simulation of the 10204 DVC in a 30 ltr sealed box at 1 and 40W.

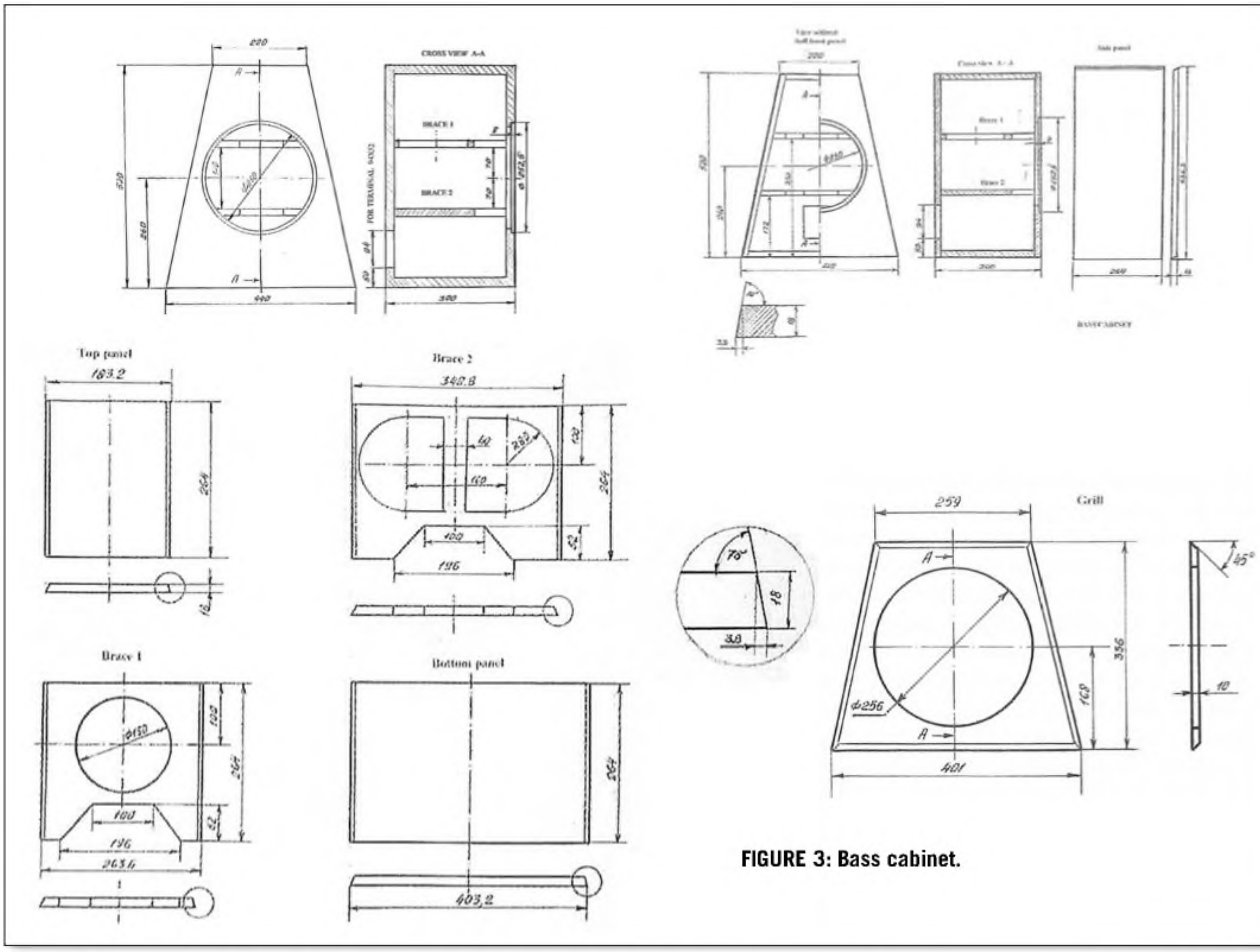

FIGURE 3: Bass cabinet.

PHOTO 4: Ground plane and gated measurement of the drivers in cabinets.

PHOTO 5: The loudspeakers with STORM glass amplifier at the exhibition.

CROSSOVER

I used LEAP 4.6 for crossover design. *Figure 7* shows the LEAP simulation of the crossover section transfer functions. *Figure 8* shows the LEAP simulation of drivers with crossover section SPL response. *Figure 9* shows the circuit diagram of the crossover network, which consists of two parts: bass circuit and mid-high circuit. Crossover elements are mounted at two boards. This design makes it easy to use the mid-high box as a satellite and biamp system if it is necessary.

When I made and installed the crossover boards into the cabinets, I completely measured the loudspeaker's parameters. *Figure 10* shows composite on-axis frequency response. This is a combined curve: scaled –6dB ground plane and gated measurements. *Figure 11* shows system impedance over full audio range measured with constant current.

EPILOG

I demonstrated this loudspeaker system at the "Hi End of Russia" 1999 show and called it "The White Pyramid" (*Photo 5*). I received many positive responses from both participants and guests who listened to the system at the show.

One of the participants was the firm

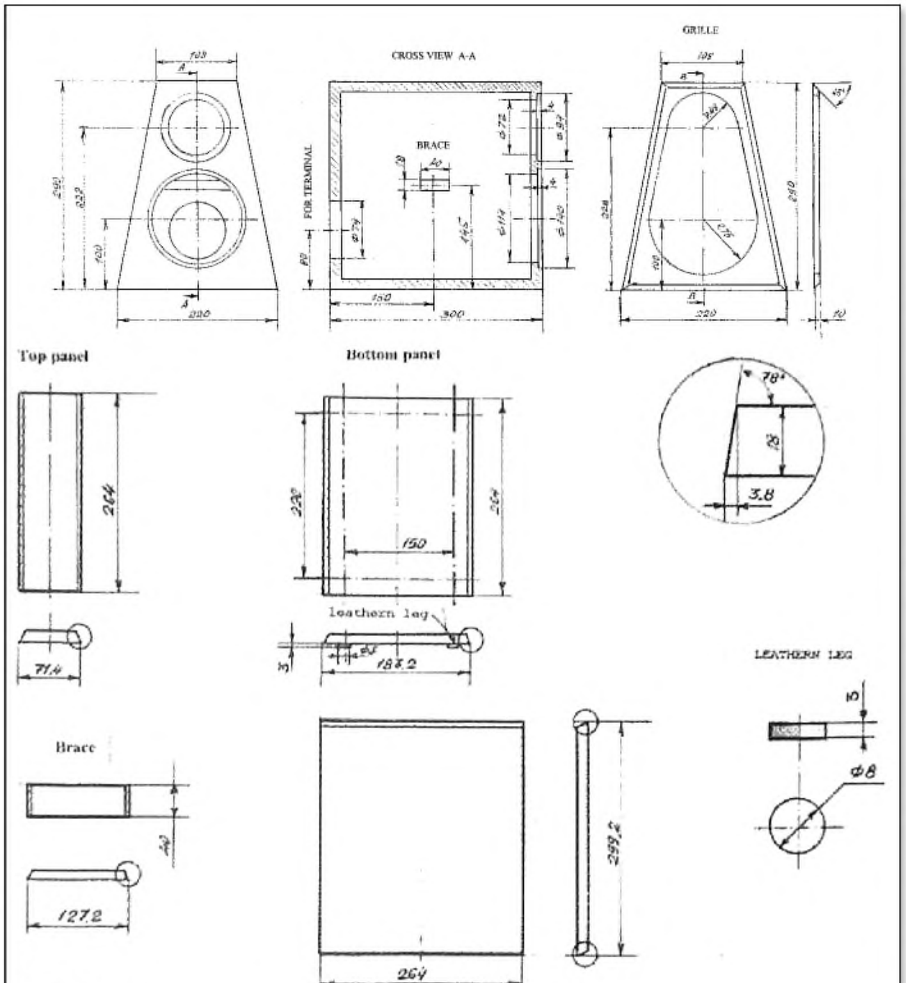

FIGURE 4: Mid-high cabinet.

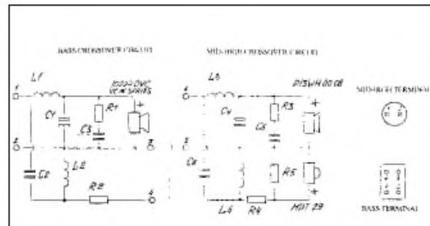

FIGURE 9: Circuit diagram of crossover network.

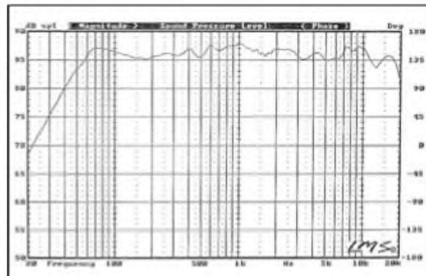

FIGURE 10: System measured SPL response.

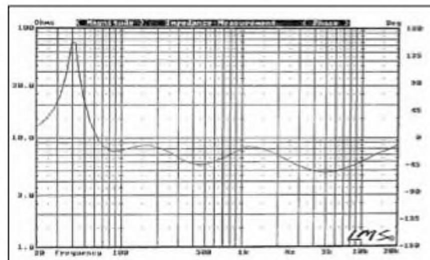

FIGURE 11: System impedance measured with constant current.

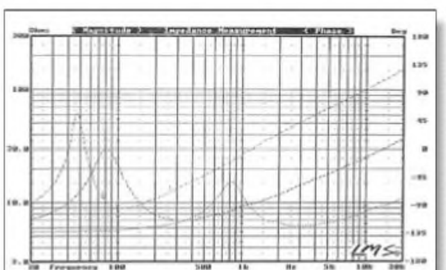

FIGURE 5: Woofer, midrange, and tweeter impedances measured with constant current.

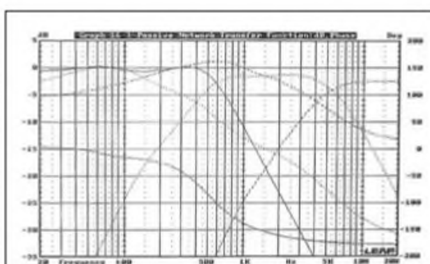

FIGURE 7: Simulated crossover section transfer functions.

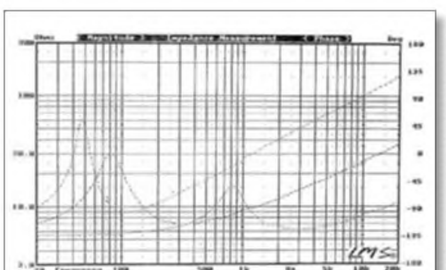

FIGURE 6: The drivers' measured SPL response.

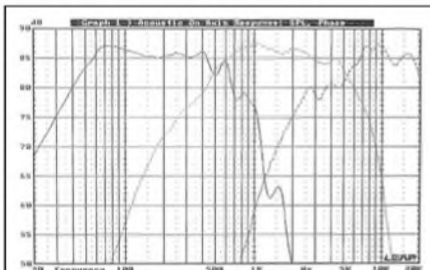

FIGURE 8: Drivers with crossover section simulated SPL response.

TABLE 1
CROSSOVER PARTS LIST

C1, C2	30.0μF polypropylene
C3	47.0μF polypropylene
C4	4.0μF polypropylene
C5	10.0μF mylar
C6	3.9μF polypropylene
L1	3.0mH 1.0Ω
L2	4.5mH 1.3Ω
L3	0.3mH 0.3Ω
L4	0.5mH 0.35Ω
R1	10.0Ω 15W
R2	2.2Ω 10W
R3	8.0Ω 10W
R4	3.3Ω 10W
R5	20.0Ω 10W

ARKADA from St. Petersburg, which is one of the Peerless distributors in Russia. The manager of this company, Roman Shmidt, was interested in my devices, and as a result I signed a contract with this company and currently design loudspeakers for ARKADA. Some of my works are presented at their website, www.arkada.com. ❖

Product Review

Perpetual Technologies P-1A/P-3A

Reviewed by Gary Galo

Perpetual Technologies P-1A Digital Correction Engine and P-3A Digital-to-Analog Converter. Perpetual Technologies, L.L.C., 368 South McCaslin Blvd. #189, Louisville, CO 80027, 303-543-7500, FAX 303-543-7200, www.perpetualtechnologies.com. P-1A—$1,099, Loudspeaker Correction Software for P-1A—$399, Room Correction Software for P-1A—$699 (availability unknown), P-3A—$799.

Perpetual Technologies is a fairly new audio company operated by Mark Shifter, a name familiar to long-time readers of this magazine's predecessors, *Audio Amateur* and *Audio Electronics*. Mark's previous company, Audio Alchemy, made a number of high-value products discussed in those pages, including the DAC-In-The-Box and the Digital Transmission Interface. The Digital Transmission Interface was one of the first outboard jitter suppressors, and made many audiophiles—myself included—aware of the importance of reducing clock jitter.

The P-3A digital-to-analog converter is based on Cirrus Logic's CS4397 D/A converter, a 24-bit chip that will operate at sampling frequencies as high as 192kHz. The CS4397 is a combination chip that includes a digital interpolation filter, an 8× oversampling multi-bit delta-sigma modulator, and digital-to-analog converter. The P-3A is the first outboard DAC I've seen that uses a combination filter/DAC chip—these are usually found in CD and DVD players. However, the multi-bit architecture of the CS4397 offers lower out-of-band noise and reduced sensitivity to clock jitter when compared to 1-bit designs more typically found in stand-alone CD and DVD players.

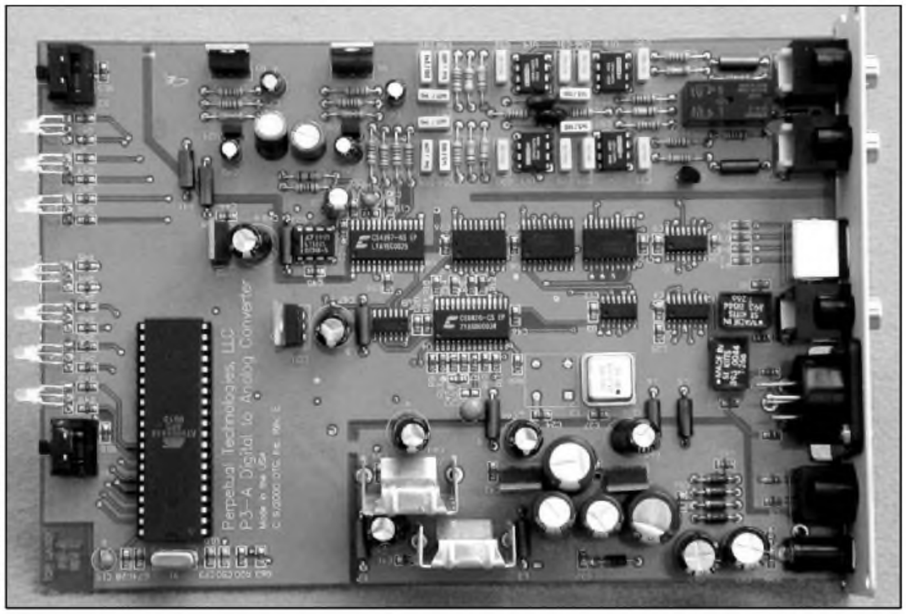

PHOTO 1: PC board for the P-3A D/A converter. This product also contains a CS8420 sampling rate converter, along with Cirrus Logic's CS4397 combination digital filter and D/A converter chip. Analog circuitry consists of four Burr-Brown OPA134 op-amps.

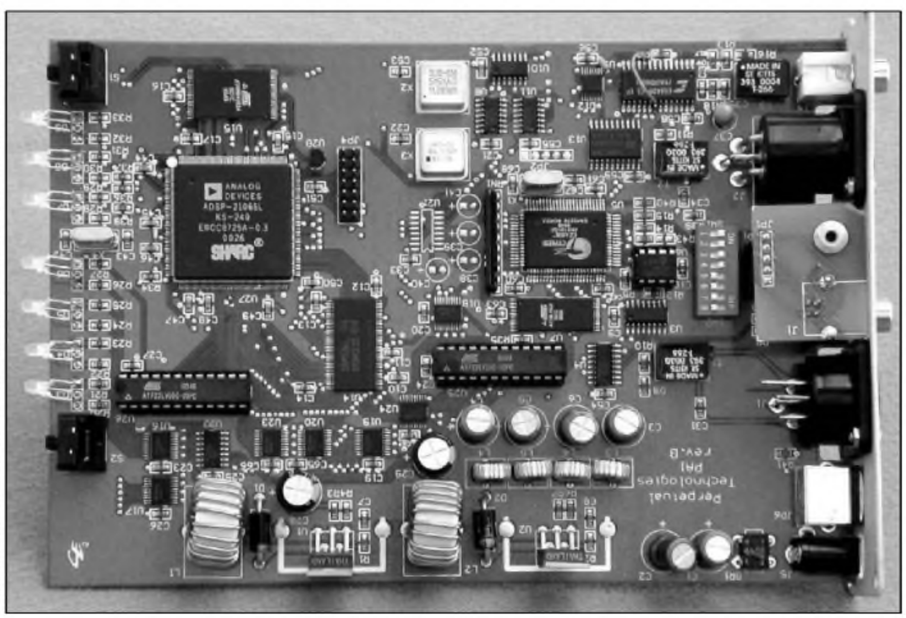

PHOTO 2: PC board for the P-1A. Analog Devices' ADSP-21056L (upper left) is the heart of the digital signal processing used for loudspeaker correction. Cirrus Logic's CS8420 sampling rate converter (upper right) upsamples all digital inputs to 96kHz.

Like most combination filter/DAC chips, the CS4397 has only voltage outputs—the current-to-voltage conversion is done internally. This can be a performance disadvantage since the internal I/V converters usually offer inferior performance when compared to the ultra high-speed current feedback op-amps used as I/V converter chips in many high-end designs. The CS4397 is capable of performing de-emphasis in the digital domain, which is typical of modern digital filters (more on this later).

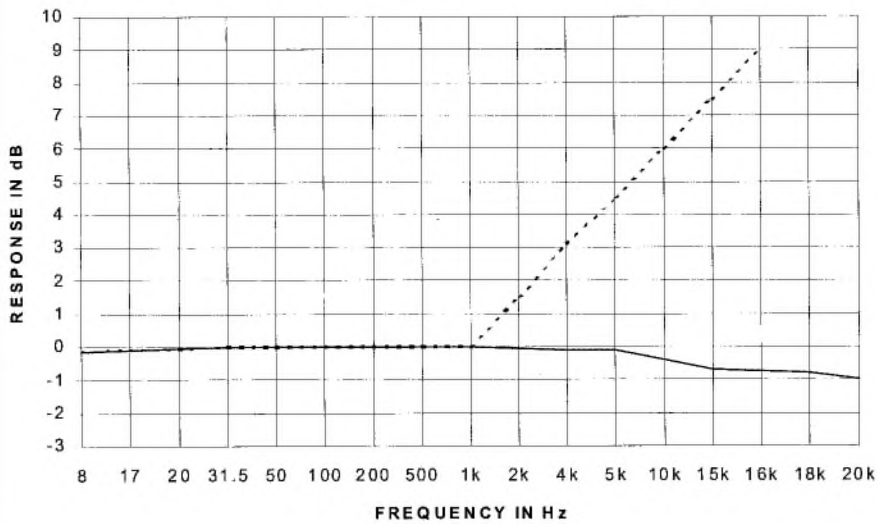

FIGURE 1: Solid line is the frequency response of the P-3A. The dashed line is the de-emphasis error. Strangely, the P-3A does not provide de-emphasis to CDs recorded with pre-emphasis.

The P-3A also incorporates Cirrus Logic's CS8420 asynchronous sampling rate converter chip, and automatically upsamples all digital inputs to 96kHz (*Photo 1*). The CS8420 incorporates an AES3-type input receiver, and also supports serial digital audio inputs.

One of the features of the CS8420 is excellent rejection of clock jitter. Perpetual Technologies appears to have selected the external PLL filter components for slow lock times, which yields the best jitter suppression. The –3dB point with this scheme is 1kHz. The asynchronous nature of the CS8420 also contributes to the device's inherently low clock jitter (as Analog Devices' ubiquitous AD1890-series devices did back in the 44.1kHz days).

The P-3A is supplied with an external AC power transformer, which supplies 9V AC at 2.5A. The AC supply is an "in-line" type rather than a "wall-wart." Internally the P-3A has eight separate voltage regulators, all 3-terminal 7800-series types (two of these are actively buffered). Each regulated supply is inductively decoupled, which reduces high-frequency noise and minimizes high-frequency crosstalk between the digital and analog ICs.

High-performance Burr-Brown OPA134 op-amps are used in the analog circuitry, two per channel. The analog circuitry is driven differentially by the CS4397's analog outputs. A 5-pole analog filter, incorporating a combination of active and passive techniques, is used for the post-DAC output filtering.

The P-3A includes the three most common digital inputs—S/PDIF coax (via an RCA connector), AES/EBU (an XLR connector), and Toslink optical. Perpetual Technologies has also included an I^2S input, following the specification used in the Audio Alchemy products (which is essentially the Philips spec), using a mini-DIN connector. The P-3A will accept input sampling frequencies up to 108kHz, with word lengths as high as 24-bit. As an option, the P-3A can be equipped to accept sampling frequencies as high as 192kHz via the I^2S input—contact the manufacturer for details.

Normally, the P-3A sends everything through the CS8420 sampling rate converter. Front-panel settings also allow the device to be used in direct mode when the I^2S input is used. In this case, the serial digital audio input (along with the clock and data recovery) of the CS8420 is used, but the sampling rate conversion is internally bypassed; the CS8420 supports internal bypassing of the SRC.

P-1A DETAILS

The P-1A Digital Correction Engine is intended as a companion to the P-3A. On the surface, the P-1A appears to be an outboard jitter suppression and upsampling device, based on the same CS8420 sampling rate converter chip used in the P-3A (see sidebar "Upsampling Demystified"). Perpetual Technologies refers to the upsampling function of the P-1A as "Resolution Enhancement." They claim to have developed a sophisticated digital interpolation algorithm that provides superior performance when compared to the CS8420 used as a stand-alone upsampling device.

According to the Cirrus Logic datasheet, the CS8420 supports an I^2C/SPI-compatible microcontroller interface allowing "full block processing of channel status and user data via block reads from the incoming AES3 data stream and block writes to the outgoing AES3 datastream." Although no schematics were provided with my review samples, it would appear that Perpetual Technologies is controlling the digital interpolation via the CS8420's microprocessor interfacing.

In addition to the effective clock jitter suppression and upsampling, the P-1A also uses DSP (Digital Signal Processing) technology for loudspeaker and loudspeaker-plus-room correction. At the heart of the P-1A's loudspeaker and loudspeaker/room correction is Analog Devices' ADSP-21065L SHARC (Super Harvard ARchitecture Computer) Digital Signal Processor (*Photo 2*). The loudspeaker and loudspeaker/room corrections are available as incremental options for the P-1A.

As of this writing, a relatively small number of loudspeakers are supported—41 models from 22 manufacturers. The P-1A is equipped with a USB computer interface connector, allowing licensed users to download upgrades to their loudspeaker correction program. The loudspeaker algorithms include correction for frequency response, phase response, and cumulative spectral decay.

Perpetual Technologies had listed the combined loudspeaker/room correction option at $699, but the price has recently been removed from their website. There appears to be a delay in getting this option to the market. My review sample was not equipped with the loudspeaker and room correction capabilities, so my evaluation of the P-1A was strictly as an upsampler and jitter suppressor. The ADSP-21065L was chosen with future possibilities in mind. The manufacturer claims that the programmable DSP architecture supports future upgrades, including multi-channel loudspeaker and room correction, Dolby Digital®, DTS®, and HDCD® decoding, but none of those options are available as of this writing.

The P-1A is supplied with an in-line outboard power transformer/rectifier that supplies +12V DC at 670mA. The P-1A does not use the usual 7805 3-terminal regulators. Instead, a pair of Linear Technology LT1074 Step-Down Switching Regulators is used to regulate the DC supply rails, one at +5V and the other at +3.25. I was not familiar with the LT1074 series prior to preparing this review, and I have not seen this chip used in other audio equipment (that certainly doesn't mean that it hasn't been).

The LT1074 contains a 5A onboard switch operating at a frequency of 100kHz. Linear Technology claims "greatly improved dynamic behavior" over conventional regulators. The regu-

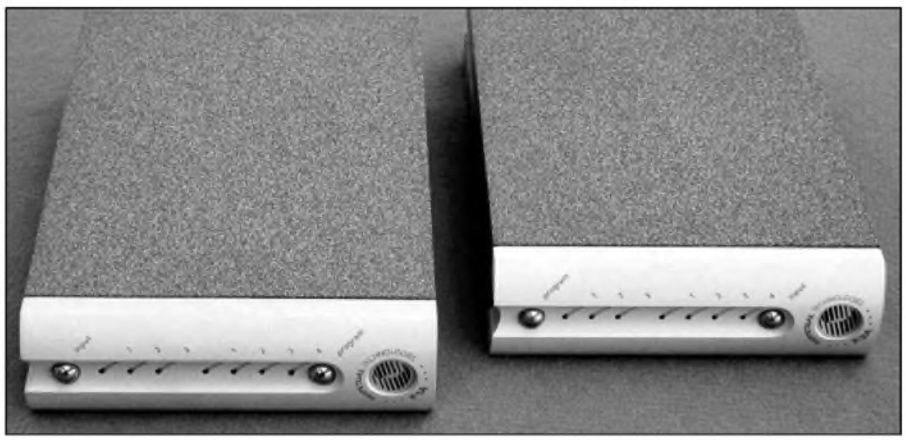

PHOTO 3: Perpetual Technologies' P-1A Digital Correction Engine (left) and P-3A Digital-to-Analog converter (right). Both products can also be oriented vertically with the stands supplied by the manufacturer.

PHOTO 4: Rear views of the P-1A (left) and P-3A. The P-1A includes S/PDIF, AES/EBU, and Audio Alchemy-style I^2S inputs and outputs. The P-3A includes the same three inputs, plus a Toslink optical input, along with stereo audio outputs.

lator used in the P-1A is the "CT" version, packaged in a 5-pin TO-220 case (physically identical to Linear Technology's well-known LT1010 buffer).

OPERATION

The P-1A and P-3A (*Photo 3*) are generally simple to install and operate, with a few exceptions. Both units require connecting the power supplies to the jack on the rear panel. If you purchase both, this can lead to confusion because the power connectors used on the two supplies—one at +12V DC and the other at 9V DC—are identical. To make matters even more confusing, there are no markings on the rear panels to indicate which type of supply should be connected (*Photo 4*).

Perpetual Technologies should, at the very least, label the rear panel, and should also consider changing the power connectors on one of these products so the supplies can't be accidentally interchanged. There are no power switches on the P-1A and P-3A—the units are intended to be left on, or switched with a power line filter (the power line filters from Adcom and Parasound, among others, provide sequenced power switching of audio components).

If you purchase both the P-1A and the P-3A, you should connect them using the I^2S interface (an I^2S cable is supplied with the P-3A). The I^2S interface makes a notable improvement in performance, since an unnecessary S/PDIF encode/decode process is eliminated. The P-1A manual notes that I^2S inputs and outputs are handled a bit differently from the other inputs and outputs. I^2S signals are fed through the resolution enhancement (DSP) circuitry, but not through the sampling rate conversion. This has been done so that when 192kHz data becomes available, it will not be hampered by the limitations of the 96kHz limit of the CS8420 sampling rate converter.

What the manual doesn't tell you is that this applies only when the I^2S input and output are used. The manual points out that if you use the I^2S input along with the S/PDIF or AES/EBU output, the sampling rate converter remains in the signal path. But the manual does not mention that the converse is also true: If you use the S/PDIF or AES/EBU input and the I^2S output, the sampling rate converter is also in the signal path. This is confirmed by the fact that when a CD player is connected to the P-1A via S/PDIF, and the P-1A is connected to the P-3A via I^2S, the P-3A indicates that a 96kHz signal is being received (assuming that you have selected an output sampling frequency of 96kHz on the P-1A).

Each unit is equipped with two rows of LEDs—seven total—that are used to program the units for various modes of operation. This can be a little cryptic, and will probably require repeated referral to the Operating Instructions until you get the hang of it. The P-1A has three LEDs indicating the input in use. The input button is toggled until the desired input is selected. Another row of four LEDs indicates the selected program, either Bypass, CD Resolution Enhancement, Correction (if available in your unit), and Resolution Enhancement plus Correction.

Since my review sample was not equipped with the correction software, my choices were limited to Bypass and Resolution Enhancement. Again, you simply toggle the program button until the appropriate LED is lit. The program and input buttons are also used to select the output sampling rate (44.1, 48, or 96kHz; 192kHz when it becomes available), and the output bit rate (16, 18, 20, or 24-bit). When you use the P-1A with the P-3A, you will normally set the output at 96kHz/24-bit.

The P-3A has four input LEDs, corresponding to Toslink, AES/EBU, S/PDIF, and I^2S, which you select by toggling the input button. You can use the program button to invert absolute polarity, and the input button to select the I^2S-direct mode of operation. The I^2S-direct mode bypasses the sampling rate conversion inside the CS8420 chip. This is the desired mode of operation when a 96kHz signal from the P-1A is fed to the P-3A.

Program LED #2 is used to indicate the input sampling rate frequency—red, orange, or green illumination for 44.1, 48, and 96kHz. Strangely, this LED gives a correct indication only when fed from the P-1A via the I^2S interface, and with the P-3A set to the I^2S-direct mode.

Connecting a CD or DVD transport, or any other S/PDIF coax source, gives a green indication, regardless of the source sampling rate. The indicator is also incorrect when the P-3A is fed, via I^2S, from Parts Connexion D2D-1 Sampling Rate Converter. The LED is green no matter what output sampling rate has been selected for the D2D-1.

NO DE-EMPHASIS

I checked the frequency response and the de-emphasis error of the P-3A using the *Hi-Fi News and Record Review* Test CD, V.2. I was quite surprised to find that the P-3A does not provide de-emphasis to CDs that have been recorded with pre-emphasis. The de-emphasis error is shown in *Fig. 1*. Relative to 1kHz, the P-3A's output on CDs with pre-emphasis rises to +3.1dB at 4kHz, and +9dB at 16kHz. CDs with pre-emphasis will sound too bright when played on the P-3A.

For some audiophiles, this may not be a concern, since only a relatively small number of CDs have been recorded with pre-emphasis. For some of us, however, it is a problem, precluding the purchase of the P-3A for our systems.

I own about 3,500 CDs, and have a substantial number with pre-emphasis, including the entire Eliahu Inbal Mahler cycle on Denon. At first I thought that perhaps the CS8420 Sampling Rate Converter was not designed to pass the de-emphasis flag. This proved not to be the case. I connected the P-1A to another D/A converter—a Monarchy 22C—and found that the P-1A is totally transparent. The Monarchy D/A converter provided correct de-emphasis with the P-1A ahead of it.

At digital 0dB, the P-3A's analog voltage output is 2.35V RMS per channel. When you use the P-1A ahead of the P-3A, the P-3A's output drops to 1.17V, a 6dB reduction. The P-1A's DSP circuit probably re-writes the digital data to ensure that the maximum boost available from the loudspeaker/room correction will not drive the P-3A into digital clipping. THD at 1kHz, using track 49 of my Denon Audio Technical CD, measured 0.0087%, consisting of mostly low-order harmonics plus high-frequency noise.

THE SOUND

I conducted my sonic evaluations using the P-3A as a stand-alone unit, and with the P-1A and P-3A in combination. Used as a stand-alone D/A converter, the P-3A is a musical performer, offering a delicate, detailed sound from my reference CDs. The soundstage width is a bit narrow, never reaching the outside edges of my loudspeakers. Soundstage depth is moderate, with reasonably precise localization.

PHOTO 5: Outboard power supplies for the P-1A (left) and P-3A. The P-1A supply outputs 9V AC at 2.5A. The P-3A supply has an internal rectifier, and outputs +12V DC at 670mA. The DC power connectors are the same for both supplies.

I continually noticed a bit of congestion on full orchestral passages, and a slight "spit" in the treble region when the unit is pushed. I suspect that the internal I/V converters in the CS4397 are a limitation (in my experience, the high-speed current feedback op-amps used in today's best DACs tend to improve clarity and smoothness in the treble region). The tonal balance is on the lean side, with the bass region lacking the weight and impact of a reference-quality DAC, though bass definition is quite good. Here, I believe that the outboard power supply (*Photo 5*) is the primary limitation.

Adding the P-1A to the signal path, with the I^2S interface, and the P-3A set in the I^2S-direct mode, gives a greater sense of depth, with improved localization. The treble region is smoother, with greater air and space around the instruments. The low strings in track 2 of the Reiner *Pictures* are gutsier and warmer. The front-to-back positioning, as well as the relative height information, of the two rows of trumpets in the last movement of the Reiner *Scheherazade* are rendered with greater precision.

I also tried Classic Records' 96kHz/24-bit DVD reissue of Rachmaninoff's *Symphonic Dances*, using my Pioneer DV-525 DVD player, with its 96kHz/24-bit digital output, feeding the P-1A. Here, the P-1A offers the best sonics when it is set to the Bypass mode (the input is already at 96kHz; the Resolution Enhancement mode will simply rewrite the data, since upsampling is impossible with a 96kHz input). The reproduction of this reference-quality recording was quite good, though the weight and impact of this recording's spectacular bass were rendered a bit thin. Soundstage width makes it to the middle of the loudspeakers without actually reaching the outside edge.

The P-1A/P-3A combination did a good job of reproducing individual instrumental timbres, which are amazingly palpable on this recording. Removing the P-1A from the signal path resulted in a more aggressive treble, less pre-

UPSAMPLING DEMYSTIFIED

There has been much discussion and speculation in the audio press on the issue of "upsampling." I'm afraid that a great deal of misinformation has been perpetuated, some of it by manufacturers. In the Operating Instructions for the P-1A, Perpetual Technologies notes that the resolution enhancement engine "upgrades your standard CD library to spectacular 24-bit, 96kHz" performance. The implication is that a CD, which is a 44.1kHz/16-bit medium, can be transformed to 96kHz/24-bit levels of performance, equaling the sonic performance of an original 96kHz/24-bit recording.

This is simply untrue. Upsampling does not—indeed, cannot—add information that was not recorded in the first place. Upsampled CDs still have the same 20kHz bandwidth restriction, with the same group delay characteristics, and the same 16-bit resolution. So, why can CDs upsampled in playback sound better than they do when used with conventional CD playback equipment?

Simple: Upsampling the data to 96kHz/24-bit allows the new digital filters and D/A converters to offer their highest levels of performance. An 8× oversampling, 96kHz digital filter will oversample a 44.1kHz CD to a frequency of 352.8kHz. The same digital filter will oversample a 96kHz input to 768kHz, allowing that filter, and its associated DAC chip, to perform to their full potential. Similarly, when a 16-bit signal has been re-mapped to 24-bit, the excellent low-level linearity of a 24-bit digital filter and DAC can be fully realized.

Your CDs sound better "upsampled" because upsampling minimizes limitations in the playback D/A conversion system. You are now one step closer to retrieving all of the information recorded on your CDs in as accurate a fashion as current technology permits. You'll undoubtedly find that your CDs contain more information than you thought—your old playback system was simply incapable of retrieving all of that information.

cise soundstaging, and a drier sonic presentation. Even though the upsampling feature of the P-1A is unnecessary with this recording, the reduction of clock jitter still makes the P-1A a worthwhile addition.

I also tried the P-3A D/A converter with Parts Connexion D2D-1 Sampling Rate Converter, again using the I^2S interface (the D2D-1 has an Audio Alchemy-style I^2S output), with the D2D-1's output sampling rate set to 96kHz. Strangely, the D2D-1 will only work with the P-3A if the P-3A is set to the non-direct mode. Setting the P-3A to I^2S-direct causes total audio mute, even though the input LED indicates a digital lock.

Even with this limitation, the D2D-1 outperformed the P-1A. The D2D-1 is gutsier and more dynamic, with improved bass extension, weight, and impact. The sonic presentation is more spacious, with greater ambience retrieval. The positioning of the trumpets in *Scheherazade* is even more precise, and the low strings in *Pictures* have a richer harmonic palette. Similar improvements were audible with the DVD of Rachmaninoff's *Symphonic Dances*.

POWER-SUPPLY LIMITATIONS

Early in my evaluations I suspected that dynamics and bass performance could be improved by more robust outboard power supplies (and better regulation, in the case of the P-1A's +12V DC supply). Recently, Perpetual Technologies posted an endorsement for a third-party outboard supply on their website: the P3 Perpetual Power Plant, manufactured in California by Monolithic Sound (www.monolithicsound.com). This product contains both +12V DC and 9V AC supplies, each rated at 2.5A, and replaces both of the stock outboard supplies that come with the P-1A and P-3A.

You can find a photo of the inside of this supply on the website www.diestaudio.com, but it reveals nothing exotic. The supply contains an IEC-type AC line filter and two dual-bobbin power transformers (for further rejection of power line noise). The +12V supply has four high-speed rectifiers and what appears to be a pair of 3-terminal regulators. The P3 Perpetual Power Plant retails for $350. I have not tried it, but based on past experience with such matters, I would expect it to make a worthwhile improvement in the P-1A and P-3A.

Adding the cost of the Perpetual Power Plant to the P-1A/P-3A combo brings the total cost to $2,248, within shouting distance of Parts Connexion DAC 3.1 plus their outboard D2D-1 Sampling Rate Converter. The basic assembled DAC 3.1 retails for $1,699, and the D2D-1 is another $699.99 (for several months, Parts Connexion has had the D2D-1 on sale for $399).

I am currently using my DAC 3.0 (reviewed in *AE* 6/00) with the D2D-1 and find the combination to be very close to *ne plus ultra* (a report on the D2D-1 is in progress). The DAC 3.0 as a standalone unit outperforms the P-1A/P-3A combo in every respect. I doubt quite seriously that the Perpetual Power Plant will raise the performance of the P-1A/P-3A to the level of the DAC 3.0 or 3.1, especially if the D2D-1 is added to either of these Parts Connexion DACs (the DAC 3.1 is the successor to the 3.0, incorporating a Pacific Microsonics PMD200 digital filter that offers both HDCD decoding and true 96kHz/24-bit performance).

CONCLUSIONS

Perpetual Technologies' P-1A and P-3A are fine products faced with stiff competition. The P-1A offers good performance as an outboard jitter suppressor and upsampler. But, the price tag is difficult to justify for these functions alone, since Parts Connexion D2D-1 outperforms it for far less money.

The performance of the P-1A as a loudspeaker and loudspeaker/room corrector will ultimately determine whether the asking price is justified (I hope that one of *aX's* loudspeaker specialists will have a look at the P-1A's loudspeaker and room correction capabilities at some future time). One obvious competitor to the P-3A is Parts Connexion DAC 2.7, an upgraded DAC 2.6 incorporating the PMD200 digital filter (the DAC 2.6 was also reviewed in *AE* 6/00).

I no longer have the review sample of the DAC 2.6 on hand, but my recollections and notes indicate a preference for it over the P-3A. The DAC 2.6 and DAC 2.7 have the performance advantage of separate digital filter and DAC chips, and high speed, current-feedback I/V conversion, along with more robust power supplies. The DAC 2.7 sells for $749, less than the P-3A. Adding a D2D-1 to the DAC 2.7 puts the combined cost at $1,448, substantially less than the P-1A/P-3A combo.

In fairness, I must add that the D2D-1 fails to pass the de-emphasis flag to the D/A converter. But, you are not left out in the cold, since you can unplug your digital cable from the D2D-1, and connect your CD transport directly to the DAC 2.7 or 3.1 to properly decode discs with pre-emphasis.

If the loudspeaker and/or room correction possibilities of the Perpetual Technologies products are of interest to you, then these products may prove to be worth the asking price. If you do not desire the correction capabilities, then the products from Parts Connexion will be a better bet.

Perpetual Technologies might consider dropping the price of the P-3A to bring it below that of Parts Connexion DAC 2.7. They might also consider a version of the P-1A without the DSP circuitry, just offering jitter suppression and sampling rate conversion for those who aren't interested in, or can't afford, the loudspeaker and room correction. This would bring these products closer to the killer-value performance in the old Audio Alchemy tradition. ❖

RECORDINGS USED FOR THIS REVIEW

Ravel: *Alborada del Gracioso*. L'Orchestre de la Suisse Romande conducted by Ernest Ansermet. London 433 717-2.

Rimsky-Korsakov: *Scheherazade*, Op. 35. Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68568-2 (UV22-Encoded Limited Edition Gold CD version of 68168-2).

Mussorgsky/Ravel: *Pictures at an Exhibition*, especially track 2, "Gnomus." Chicago Symphony Orchestra conducted by Fritz Reiner. RCA Victor Living Stereo 68571-1 (UV22-Encoded Limited Edition Gold CD version of 61958-2).

Schoenberg: *Five Pieces for Orchestra*, Op. 16. London Symphony Orchestra conducted by Antal Dorati. Mercury Living Presence 432 006-2.

Wagner: *Der Ring des Nibelungen*, especially "Siegfried's Death and Funeral March" from *Götterdämmerung* (CD 4, Tr. 10-11), and the "Forging Scene" from *Siegfried* (CD 2, Tr. 3-5). Birgit Nilsson, Wolfgang Windgassen, et al. Vienna Philharmonic Orchestra conducted by Georg Solti. Decca 455 555-2.

Rachmaninoff: *Symphonic Dances*, Op. 45. Dallas Symphony Orchestra conducted by Donald Johanos. Classic Records 96kHz/24-bit DVD (played on a Pioneer DV-525 DVD player with a 96kHz/24-bit S/PDIF digital output).

Book Review

Audiocraft

Reviewed by Scott Frankland

By the time *Audiocraft* magazine was launched in late 1955, it was already apparent that big things were happening in the hi-fi world. AR had introduced the acoustic suspension loudspeaker at the 1954 New York Audio Show, and RCA was making stereo tape recordings. In 1955, Dynaco released a blockbuster new product guaranteed to drive the power-hungry AR—the Mk. II 50W amplifier. Its price was astonishing even for those times—less than $1.40/W in kit form.

DIY AS CRAFT

This was a time of almost constant innovation—exhilarating and restorative after the uncertainty of war. Sons were emulating fathers, and kits of all descriptions dotted the marketplace. There was so much going on that several different magazines were needed to keep track of it all.

Besides *Audiocraft*, there was *Radio-Electronics*, *Electronics*, *Audio Engineering*, *Radio and TV News*, and *High Fidelity* (to name only the better-known American magazines). Among these, *High Fidelity* excelled in its direct appeal to audiophiles. Its focus was squarely on equipment reviews and LPs.

In the whole of the golden era, however, there was no other magazine in America quite like *Audiocraft*. Compet-

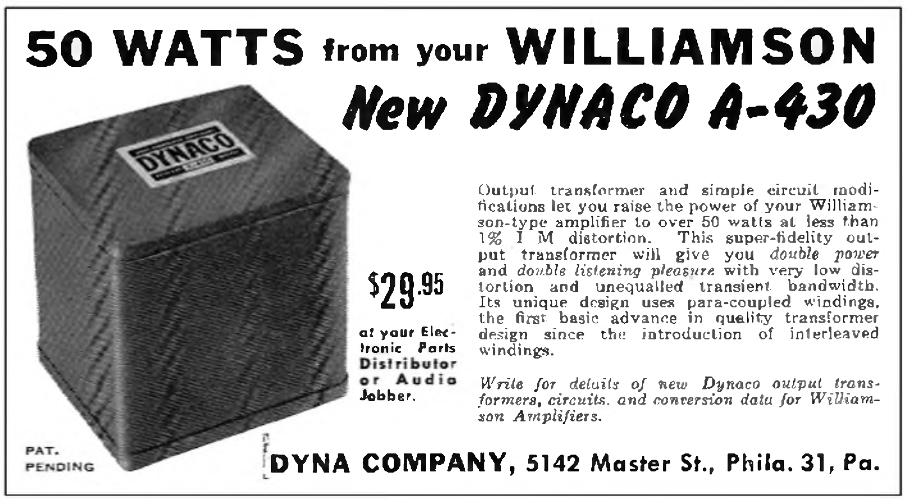

PHOTO 1: Dynaco transformer ad (*Audiocraft*, January 1956, p. 40).

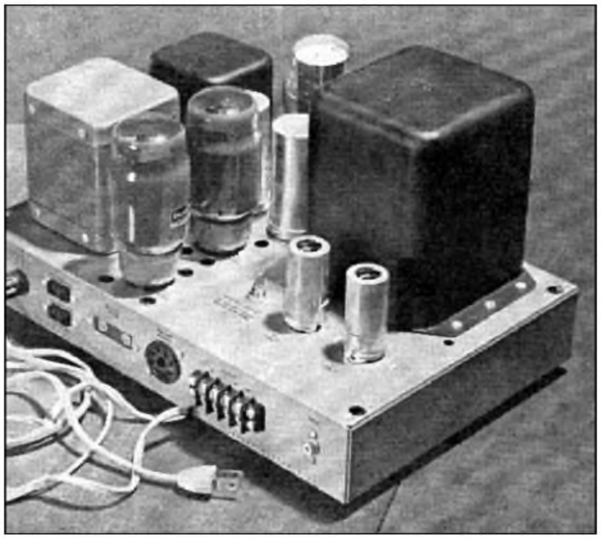

PHOTO 2: Heath W-5M amplifier (*Audiocraft*, November 1955, p. 26).

ing magazines published construction articles, but they were not nearly so focused in their approach to DIY. *Audiocraft* treated DIY as an ongoing craft, and it set about to educate its readers, if not to train them.

Audiocraft approached the home constructor in a manner that can only be described as comprehensive: it reads more like a mail-order instruction course than a magazine. From it, you learn how to set up a workshop and how to build, test, and maintain high-fidelity components. Furthermore, you learn the theory behind the designs, right down to the bare fundamentals. You are further instructed in how to listen for those unique attributes found only in high-fidelity components.

Audiocraft began as a publishing experiment by *High Fidelity* magazine. *High Fidelity's* audio editor, Roy Allison, thought that reader interest in DIY had by 1955 become sufficiently widespread that a specialized publication should be produced to cater to it. *High Fidelity's* management agreed, and placed Allison in charge of the project.

Allison recruited J. Gordon Holt, an avid recordist, to write a monthly column on tape news and recording equipment.* Joseph Marshall, a leading experimenter with amplifier circuits, was hired to report on significant new trends. Irving Fried's column explained how to test and repair just about anything in the audio chain.

In addition to the regular staff writers, there were several long-term contributors who each wrote a series of articles on a single theme. For example, Norman Crowhurst explains—over the course of many issues—how to design amplifiers. Likewise, George Augspurger explains loudspeaker principles, including enclosures and horns. R.D. Darrell covers LPs and tapes (but only those worthy of the appellation "high fidelity"). E.B. Mullings explains the ins and outs of test equipment. Editor Roy Allison provides a thorough grounding in the fundamentals of electronics theory. Joseph Marshall explains how to design and construct all manner of power supplies. And George Bowe provides a forest of woodworking advice.

CLASSIC DIY

Besides these regular columns by experienced audio writers, there are articles by talented amateurs on virtually every aspect of audio construction—including woodworking, chassis layout, soldering, testing, and troubleshooting. Everything from tonearms to tape recorders is covered.

To round things out, there are articles by leading manufacturers—such as David Hafler, Paul Klipsch, and Edgar Villchur. In the January 1956 issue, Hafler shows how to convert the Williamson amplifier from triode to ultralinear. Modifying the Williamson seems to have been a particular pastime of golden-era audiophiles.

The Hafler-modified Williamson produced twice the power output of the stock Williamson. The trend toward 50W+ amplifiers can be laid squarely at the feet of David Hafler, who often claimed in his articles that more powerful amplifiers sound better. Hafler had good reasons for saying so, and he enumerates them in his *Audiocraft* article.

In any case, Edgar Villchur's acoustic suspension loudspeaker served to cement Hafler's claim. These compact sealed-box loudspeakers, first produced by AR, demanded lots of power. At the time of its introduction, Julian Hirsch declared the AR to be the most inefficient loudspeaker ever tested—except in the low bass.

Hirsch also said that the AR established a new industry standard for low-distortion bass. That big, full sound coming from such a small footprint (relative to the competing horn speakers) made the acoustic suspension loud-

BIRTH OF THE WILLIAMSON

Several authorities have cited the long-tailed pair phase splitter as a Mullard innovation. If so, the Mullard engineers must have devised it prior to 1938, because that is the year in which it appears in the *Journal of Scientific Instruments*[1]. W.T. Cocking, in his extensive survey of phase splitters[2], also cites the *JSI* as the source of this circuit.

As early as 1939, Cocking expresses admiration for the long-tailed pair[3]. This splitter later appeared in the Leak "Point One" series of amplifiers, launched in 1948[4]. The long-tailed pair was used most famously by Harman-Kardon and Marantz in America.

Williamson, however, chose to use the split-load or "cathodyne" phase splitter[5]. The cathodyne phase splitter dates back to 1934, at least, and is apparently a Cocking innovation[6,7].

The Williamson amplifier stems from a long-term project by the *Wireless World* editors to design a "quality amplifier." This project was begun by Cocking in 1934[6] and continued intermittently until at least 1946[8,9].

In their articles[5,6], Cocking and Williamson both called for higher standards of amplifier performance, and each exerted a tremendous influence on his peers in their day[10]. What they contributed to audio can be summed up as *hi-fi orthodoxy* (some would call it heresy).

In America, the Cocking/Williamson orthodoxy was carried over and perpetuated throughout the Golden Era by Heath and Hafler.—**SF** speaker a favorite among audiophiles for years to come.

THE CALL TO DIY

Besides the Williamson, several other British amplifier designs had a major impact on American designers. Among these should be mentioned the Leak "Point One" series of amplifiers and the Quad II. First among these, however, was the Williamson.

The rush to DIY began when Williamson heralded the importance of transient response in his *Wireless World* construction article of 1947 (see "Birth of the Williamson" sidebar). In his introduction to the circuit, Williamson emphasized that wide bandwidth and linear phase response were the keys to good transient response; but he warned that these would be pointless unless some means could be found to control high-frequency distortion. His amplifier was meant to provide the necessary evidence, and the audio world seemed hell-bent to prove him right.

The full-blown Williamson circuit was perhaps most famously used in the Heath W-5M, a kit-based amplifier of exceptional quality featuring a Peerless potted output transformer. In the very first issue of *Audiocraft*, you find an article entitled "You Can Build a Power Amplifier." Associate editor Frank Wright takes you through the steps involved in hand-wiring a Heath W-5M, imparting his evident constructional experience along the way.

The September 1956 issue included an article explaining how to convert the Heath W-3 (a classic Williamson circuit) into a Dyna Mk. II. In comparing the Williamson to the Dynaco circuit, you discern the ghost of the Williamson. In the Dynaco amp, the Williamson has been stripped down to its bare essentials. Of course, important refinements were incorporated by Hafler et al.

The original Marantz amplifier is reviewed in the same issue by Joseph Marshall, who does a typically thorough job of explaining the circuit and its many features. One of the key aspects of this amplifier is the so-called "Mullard circuit" (see "Williamson" sidebar). This is a "long-tail" quasi-balanced phase splitter that Marshall hailed as superior to the split-load type used by Williamson. A large number of golden-era manufacturers apparently agreed.

VINTAGE ARTICLES

The October 1956 issue introduces us—again by Marshall—to a working prototype of the Quad ESL-57. The pros and cons are weighed vis-à-vis the conventional dynamic loudspeaker, including the possible effect on future amplifier designs. Marshall speculated that one such effect might be a direct-drive high-voltage OTL.

To quote the mysterious Mr. Marshall: "I see in the coming of the electrostatic speaker not only a serious challenge to the cone speaker, but also a stimulus to development of a high-fidelity art along unfamiliar roads." Also in this issue, Norman Crowhurst discusses the ins and outs of multiple feedback loops.

Among my favorite *Audiocraft* articles is one by Philip Geraci in the November 1956 issue on rebuilding your tape

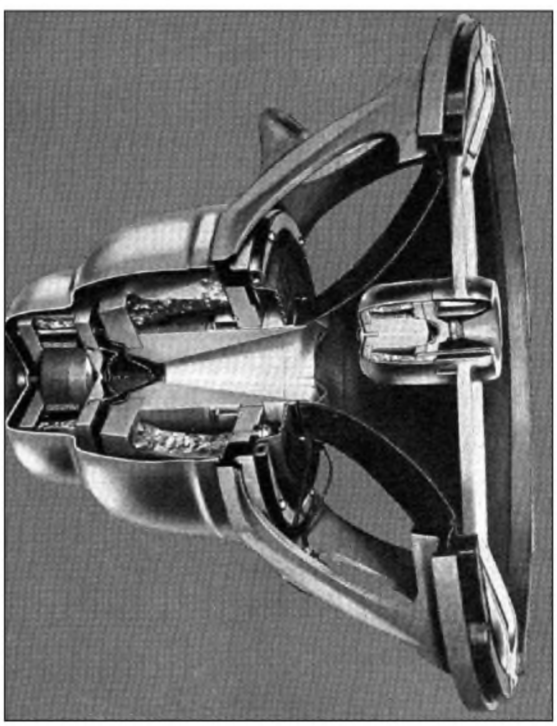

PHOTO 3: Jensen triaxial speaker (*Audiocraft*, March 1956, p. 2).

WHY DID *AUDIOCRAFT* CEASE PUBLICATION?

By Edward T. Dell, Jr.

I hesitate to disagree publicly with so respected an authority as the editor of *Audiocraft*, as well as a man who has some of the highest credentials in the audio industry—particularly where speakers are concerned. After leaving the Audiocom group, he became production manager of AR and much later founded Allison Loudspeakers. He is also one of the few authorities on speaker technology whose name is attached to an acoustic phenomenon: The Allison Effect.

However, my subject here is magazine technology, not audio or loudspeakers. As an editor/publisher with about 40 years experience with magazines, I feel on fairly firm ground in what I am about to say. I ought to preface this by warning that my ideas are speculation, albeit with some data and general knowledge to back up my suppositions.

I sat down one day and counted the ad space in *Audiocraft*. The results showed nearly 50% in every issue—even for the third and last year. This is an industry standard for magazine survival, so I am convinced there were other reasons. It is possible that the company was undercharging for its ad space, but that is highly unlikely.

The cessation notice was a small, 3″ one-column box in the November issue of 1958, page 36. It said that all the features appearing in *Audiocraft* would be transferred to *High Fidelity*'s pages. What was transferred was the "Tested in the Home" feature—very little else.

The *Audiocraft* logo persisted on *High Fidelity*'s cover for three issues and then for a year or so on the contents page. In April 1959 Roy Allison's name disappeared from the masthead, and J. Gordon Holt's is gone in the May issue—although some of his equipment reports continue to appear. Julian Hirsch begins his Hirsch-Houck Labs reports on equipment in May, replacing most of the staff-written equipment reports from then on.

The publishing company Audiocom, Inc., was sold just then by Charles Fowler, one of the co-founders of the company, to Billboard Inc. (The notice first appears in *High Fidelity*'s December 1958 issue.) Charles was a highly experienced magazine man who knew that it is wisest to start a magazine and then sell it when it becomes successful. So in its eighth year, he did.

At that time *Audiocraft* had a much smaller circulation than *High Fidelity*, so its advertising was less expensive than the larger periodical. It appears to me that Billboard just decided to kill off *High Fidelity*'s low-cost advertising "competition," and add the *Audiocraft* subscribers to the *High Fidelity* circulation list. Smart business. Most of the advertisers went along with a higher circulation in one periodical reaching the same audience.

Staff people were told that *Audiocraft* had been losing money "big time." If it had been, with 50% ad levels, something in the economics did not add up.

Billboard's management did not ask its subscribers about the change or whether they preferred refunds or a switch of unexpired copies to *HF*.

If you look at the editorial content of *HF* after the demise of *Audiocraft*, you see very little about the craft aspect of audio, but much of the advertising is for do-it-yourself products. So, Dr. Watson, with apologies to Editor Allison, I think the clues add up to a "smart business decision" which trashed a fine publication.

recorder. Virtually every aspect of this arcane subject is covered. I also enjoyed reading Roy Allison's series on electronics fundamentals. I have rarely encountered a more clear explanation of the basic mathematical principles involved.

Norman Crowhurst's amplifier series was another clear favorite, as was Joseph Marshall's series on "Listening for Quality." I also much enjoyed Gordon Holt's article in *Audiocraft's* first issue, in which he surveyed the available microphones of the day. Both photos and specs are included.

Aside from the vintage articles, there is another source of constant delight, and that is the period ads. There are numerous ads displaying vintage gear—from AR, Amperex, Dynaco, Electro-Voice, Heath, JBL, Jensen, Marantz, McIntosh, and many more. Most cases include prices, specs, and features.

IN RETRO

The release of *Audiocraft* in book form† brings the current American DIY movement full circle to its roots in the golden era. Thanks are due to Ed Dell for making available this complete compendium of articles from *Audiocraft* Volume 1 (14 issues). Volumes 2 and 3 are also expected to see print again.

I can recommend this first installment to anyone with an interest in DIY audio. Even those who are DIY-challenged will discover hidden treasure among its pages. The range of topics is sufficiently broad so that anyone with an interest in the roots of high-fidelity will find himself transported back to the '50s, rubbing elbows with the amazed and the curious at the regional hi-fi shows.

More than any other magazine, *Audiocraft* was the inspiration for Ed Dell's DIY quarterly, *Audio Amateur*, which remained in print for more than a quarter century. For years, *Audio Amateur* stood alone as a benchmark of DIY audio, until the '90s brought a renaissance of renewed interest in the craft of audio. *audioXpress*, *Positive Feedback*, *Tube CAD Journal*, *Vacuum Tube Valley*, and *Valve* magazine, among others, live on in the venerable spirit of *Audiocraft*. ❖

* Not long after *Audiocraft* folded, Holt founded *Stereophile*. The short life of *Audiocraft* was due to an insufficiency of advertisers, according to a recent telephone interview with Roy Allison. Allison believes that *Audiocraft* appeared too soon in the hi-fi boom.

† Published by Audio Amateur Press, Peterborough, NH, 2002 (available from Old Colony Sound Lab at 1-888-924-9465 or from audioXpress.com).

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Xpress Mail

POWER TO THE SPEAKER

This concerns Mr. McChesney's question regarding the power rating of resistors in Zobel networks, and Mr. Koonce's comprehensive response ("Ask *aX*," Aug. '01, p. 89).

It appears that the input power distribution to the Zobel and to the speaker has not been correctly estimated. A specific example will make this clear: consider a Focal 416 (8″) woofer, for which a 2kHz crossover is contemplated. At this frequency, the woofer has an impedance of 17.5Ω and a phase of impedance of $45°$.

A Zobel consisting of a 6.5Ω resistor and a $40\mu F$ capacitor will exhibit a system impedance of 5.52Ω at a phase angle of $-.83°$. The amplifier will present the same voltage to the Zobel and to the woofer; let this voltage be 10.

The calculation of power distribution follows. Current to the speaker will be $10/17.5 = .571A$. The resistive component of the speaker's impedance is its impedance times the cosine of the phase of impedance: $17.5\text{Cos.} \times 45 = 12.37\Omega$. Power is current squared times resistance: 4.03W. Current to the Zobel will be $10/5.52 = 1.811A$.

The resistive component of the Zobel's impedance is its impedance times the cosine of the phase of impedance: $5.52\text{Cos.}{-.83} = 5.519\Omega$. Power is current squared times resistance: 18.1W. This shows that at the crossover frequency the Zobel consumes about 4.5 times more power than the speaker consumes! Suppose that the Zobel is dissipating 50W. At this level the speaker is receiving only about 11W.

Because these power numbers differ so much from the levels that Mr. McChesney submitted, a corresponding response from Mr. Koonce would seem to be in order.

David J. Meraner
Scotia, N.Y.

G. R. Koonce responds:

The question Mr. Meraner raises with his detailed example is, "Why does he compute a Zobel power dissipation of several times the power to the woofer at 2kHz?" The answer is that this is the expected result. At high frequency the Zobel dissipation should be several times the driver dissipation, but this does not violate the guidelines in my original response.

When discussing power delivered to a driver, you get into the problem of whether you talk about true power or about "reference" power. True power is the actual number of watts dissipated by the driver. Reference power is what would be delivered to a driver if you replaced it with a fixed resistor. Normally, the resistance value used for reference power calculations is the driver's voice-coil resistance.

In his letter, Mr. McChesney cleverly avoided this confusion by discussing the Zobel dissipation in terms of the electrical power rating of the driver. I stayed with this strategy, which resulted in a predicted worst-case Zobel dissipation matching the driver's rating; i.e., the power Mr. McChesney planned to deliver to the driver at low frequency. This is not the same as the true power delivered to the driver at higher frequencies.

Clearly, with constant-voltage input, a woofer could be driven to its rated electrical power limit only at low frequency. As Mr. Meraner's example shows, at higher frequencies a woofer will dissipate less power as its input impedance rises. With rising frequency, as the true woofer dissipation falls, the Zobel dissipation rises, which is exactly why the Zobel is there: to maintain a nearly constant load on the crossover.

The intent is that the woofer-Zobel combination looks like a fixed resistance. Thus together they draw from the crossover a power approximating the reference power at all frequencies. This is the desired result as the woofer's on-axis acoustic SPL output is directly proportional to the voltage across the woofer and not the power into the woofer.

In Mr. Meraner's example the Zobel is a series 6.5Ω resistor and $40\mu F$ capacitor. This would indicate a nominal 8Ω woofer with a voice-coil resistance about matching the Zobel resistor, or about 6.5Ω. With 10V out of the crossover, at low frequency, the maximum possible driver power would be about 100/6.5, or about 15.4W. At 2kHz *and with 10V input, the Zobel current would not be 1.81A, as Mr. Meraner computes, because the resistor limits it to 1.54A maximum. This means the Zobel resistor can see no more than about 15.4W, the same as the low-frequency driver power.*

As my generalization predicts, near the upper end of the low-pass crossover, the Zobel power dissipation can approach the power you plan to deliver to the woofer at low frequency; i.e., up to the driver's electrical power rating if desired. As I pointed out, this is a worst case that normally does not occur when playing music.

If you have detailed information about the driver's input impedance and the Zobel, the best approach is to compute the actual Zobel dissipation over frequency to properly size the resistor. Lacking this information, I believe the guidelines given in my original response are valid. I hope this clarifies why Mr. Meraner's example shows a power to the Zobel resistor that is several times the true power to the driver at 2kHz. It is the desired result to get a flat on-axis response from the driver.

PREAMP PERFORMANCE

I found Eric Barbour's planar triode preamp design article (Nov. 2001, p. 18) very interesting. Eric's innovative and practical down-to-earth articles are always a delight. However, missing in this one are any performance figures.

At the minimum, an input sensitivity for a standard output level or an overall gain figure would have been nice, as would bandwidth and a minimum input impedance for the following stage required to maintain this.

Perhaps Eric might like to respond.

Bart Shepherd
Sydney, Australia

Eric Barbour responds:

Well, if you insist...voltage gain at 500Hz is about 41dB. Frequency response is well below 10Hz, and rolls off fast above 25kHz. I strongly suggest not loading down the output of the phono stage with less than 100kΩ.

I would like to build this project and experiment with the 6S17K-V and maybe with the 7077 in another design. However, I can't find any on-line shops that sell those particular devices. Can you suggest where I can find those unusual ceramic-metal tubes?

Robert Gomes
Montreal, Canada

Eric Barbour responds:

Due to the loss of a couple of major Russian surplus tube dealers, the 6S17K-V triode can be difficult to find. Bill Perry has discovered a surplus dealer in Lithuania who has them, along with lots of other Russian surplus tubes. His name is Gintaras Sakenas, and he has a website: kwtubes.s5.com. His address is PO Box 1367, Vilnius LT-2056, Lithuania, e-mail: ly2kw@kwtbes.com, FAX (370) 23-32839.

CD TRANSFERS

In the October 2001 issue, Victor Staggs did a nice job of discussing LP transfers to CD using editing software on a computer-based setup ("LP Transfer to CD," p. 30). The alternative path is using a CD recorder, which was not even mentioned. You can obtain excellent results using a pro unit such as the Tascam CDRW 2000 or Sony model 66, and clean copies of LPs. An inside look at transfers using a pro CD recorder would be worthwhile.

John Stein
Phoenix, Ariz.

Victor Staggs responds:

The Tascam CD-RW2000 is a professional, studio-quality CDRW with an analog input for direct recording, and it costs $1125 according to Teac. This costs as much as most PC computers. It would do the job nicely, except that I have yet to find a completely clean LP. All of mine have flaws of some sort.

Direct recording bypasses any computer software that might be used to clean up an imperfect LP before committing it to a CD. With a computer, you can use Roxio's Toast 5 Titanium (for Macs) or Easy CD Creator (for PCs) to record directly from an LP to a CD. This software will remove ticks and add CD access points between tracks automatically. I suspect that it will not be as clever at removing ticks as a human being using a waveform editor, but this would be the fastest way to get an LP onto a CD.

The Sony CDR-W66, at $1275, has 24-bit A/D-D/A converters, and could certainly do a better job of producing a 16-bit 44.1kHz CD than could a computer with a 16-bit converter. Teac does not specify a converter bit depth.

Since these are professional, studio-grade products, I would not expect most amateurs to use them. If there are any studio professionals among the audioXpress *readership, perhaps they would like to comment on the use of the Tascam and Sony recorders with LPs.*

HORN DIMENSIONS

I am confused about the dimensions of Bill Fitzmaurice's DR8 horn in the drawing on page 42 (Nov. 2001 *audioXpress*, "The DR8 Horn"). The top and bottom individual dimensions do not add up to the overall width of the enclosure as shown. I think the problem may be that the outside radius of 4″ PVC pipe is actually 2¼″ instead of the 1¼″ shown in the drawing. I am ready to cut these pieces and would appreciate any help to clarify this so everything will be correct when I finish.

Ronald K. Wasson
RKWASSON@prodigy.net

Bill Fitzmaurice responds:

Ouch! The dimensions in the figure are correct, but the text is not. The PVC pipe is actually 3″, not 4″. Three-inch I.D. PVC has an approximate outer radius of 1¾″. I hope you didn't end up with a piece of 4″ PVC that you can't use, but you may want to hold onto it—you'll need it to build the upcoming DR10a.

TAMING TREBLE PEAK

I decided to subscribe to the new *audioXpress* at its outset, looking forward to returning to audio electronics as I enter retirement. I was a radar tech in the USAF in the '50s/'60s, and trained on tubes and then early transistors, always maintaining good stereo systems, building Dynaco kits, and so on. So far, my experience with the magazine is very enjoyable, and the content appears to continually improve. Keep it up, especially simpler do-it-yourself projects and kits in audio, including surround and the use of PCs as test instruments or processors.

I wonder whether readers could point me to articles—or suggest techniques—to tame a nasty treble peak in my audio system. It appears to be a fairly narrow peak in the upper range of classical soprano singers, violins, trumpet, and so on. My Realistic Sound Level Meter shows a 6dB peak at 4kHz, but I suspect the peak is higher than that and possibly at a somewhat different center frequency.

I have tried various test CDs with an Audio Control 10-III equalizer, and damping with blankets around the speakers, with little effect. I'm hoping for suggestions on how to locate and to tame that nasty peak without undue expense. Maybe with a passive notch filter?

Any suggestions will be greatly appreciated. Again, keep up the good work with the magazine.

Jim Keenan
jaybee85@msn.com

AMP UPGRADES

Nice article on the amplifier project ("A Great First Amplifier Project," Dec. 2001, p. 18). I like to build amps that use "unusual" tubes. I have a couple of suggestions for upgrades, though.

First, the Hammond 1650F output is less than $20 more than the 1608, and it is much better sounding. Believe me, I have run the 1608s, which lack the punch of the bigger units. Second, it is usually advisable to add a "snubber" to the output winding; usually, a 1.0μF cap in series with a 16Ω 2W resistor at the 8Ω tap to ground works well.

Of course, a really simple amp would involve a 6Y6 operating single-ended into a One Electron UBT-3 OPT. The 6Y6 provides 6W at just 200V plate voltage; a pentode or high-mu triode driver would work nicely on this amp. Parts count for this would be less than half that of the push-pull amp, and it would definitely have a sweet tube sound!

David Wolze
San Jose, Calif.

FILTER FORMULAS

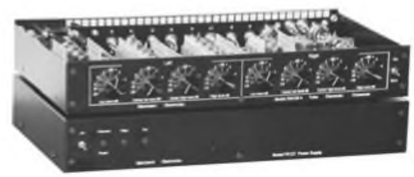
In the September 2001 *audioXpress*, Figure 1 ("Build an Automotive Sub-Satellite Speaker System," p. 10) illustrates an active low/high-pass filter. I'd like to use different cutoff frequencies, but I haven't had much luck finding formulas for determining the new component values. Where can I find these formulas?

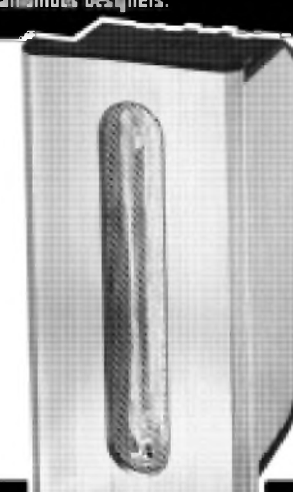
Al Moreno
Boca Raton, Fla.

See Speaker Builder, *Iss. 3, 1985, Robert M. Bullock's Part III, "Passive Crossover Networks, Active Realizations of Two-Way Designs."—Ed.*

CORRECTION

In "Testing the PSB Stratus Silver^{i}" (Nov. '01 *audioXpress*, p. 65), the contact address published for PSB Speakers was incorrect. If you wish to contact PSB Speakers, please call 1-888-772-0000.





A NEAT AMP

The Vintage Glass schematic in the Oct. 2001 issue is intriguing, to say the least. Lance Cochrane wondered about the interstage transformer feeding the 6SN7 cathode followers. I think it is more involved than that.

Note that the drive signal for the 6SN7 drivers comes, as is customary, from the anodes of the phase inverter V7 6SN7 through coupling caps C46 and C47. The positive bias for the driver cathodes comes through the secondary of T2, through R70. What is unusual is that the primary of T2 provides feedback to the phase inverter V7 anodes through R84, R83. This bootstraps the phase inverter so that it receives signal voltage from the T2 primary in series with its anode voltage from the primary's center tap, enabling it to provide a much higher drive voltage for the driver stage than otherwise!

Since the driver stage is a cathode follower, the phase inverter must, for practical purposes, provide the full drive swing for the output stage. Very neat!

I also marvelled at the switching arrangement in the lower left corner of the diagram. There are separate equalization curves for foreign, popular, classical, and long-play records, as well as for either AM or FM radio signals. I haven't worked out the turnover points yet, but it should give some insight into how the designers rated the various sources in terms of frequency response and/or noise performance.

I wish I owned one of these amps!

Jan Didden
Belgium

MYSTERY HORN

I have a 50-year-old Altec horn that I am trying to identify (*Photo 1*). It is a 2 cell by 4 cell, 2 piece, cast aluminum horn. It has a 1″ (25mm) dia. throat and is about 18″ (450mm) wide at the mouth. I would guess that it is good down to 800Hz. The gasket on the driver has NOV 51 stamped on it.

I saw a cabinet to which it supposedly was originally mounted. As I remember it (25 years ago), it had two 15″ (380mm) dia. woofers, a slotted port at the bottom, and some short front horn loading.

PHOTO 1: Mystery horn.

If anyone can identify this horn with a model number, I would be most grateful. I would like to find a matching horn if possible but cannot even look without a model number.

George Morgan
gmorgan@gecinc.com

I^2S FOLLOW-UP

Thank you for printing my letter (I^2S Outputs) in November's issue (p. 81). I wrote that many months ago and have revised what I think the circuit should be.

Even though the DF 1704 will accept an I^2S signal directly, I believe it would be better to convert the I^2S signal to the Sony/Burr-Brown format before the filter. This is just an inverter and a D-type flip-flop presented by Kal Rubinson in *Audio Electronics* 1/97 ("Get on the Bus, Part 2," p. 24).

Why? For a couple of reasons. For one, the Camelot Dragon cannot interpolate to 24 bits (20 bits only). Also, the 20-bit setting on my DTI Pro32 sounds better on my rig than the 24-bit setting. Only 20 bits are sent to the DACs anyway—the filter truncates the word length.

Also, if you need to change the input word length options in the DF 1704 (16, 20, 24 bits), you must manually switch this on the board.

Figure 1 is a drawing of the conceptual change in case some digital board

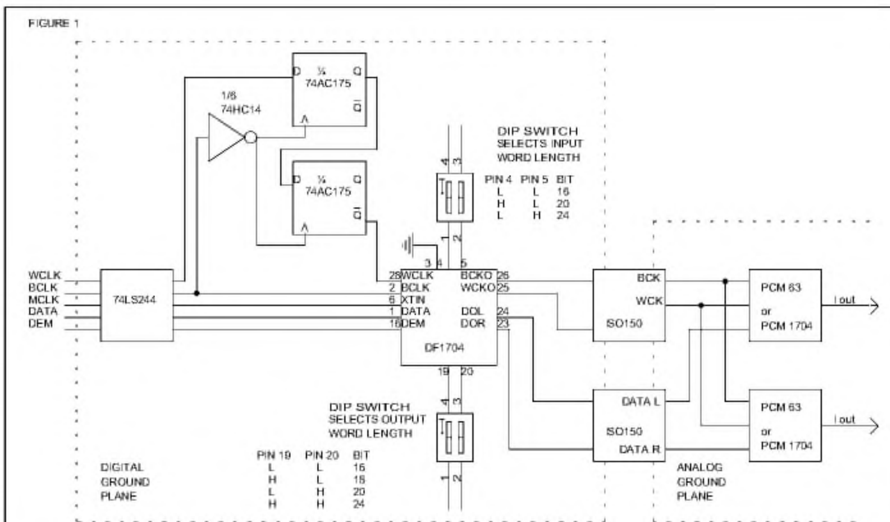


FIGURE 1: Revised I^2S circuit.

wizard actually makes such a thing. Too bad Burr-Brown doesn't make the splendid DF 1704 in a larger surface-mount package with a 20-bit I^2S option.

Rick Bergman
Missoula, Mont.

MOTORBOAT FIX

I would like to avoid the "motorboating" problem on the 150V supply you described ("A Switching Power Supply for a Mike Preamp," Nov. 2001 *audioXpress*, p. 32). Can you provide a source for ESD1 diodes?

Bill Perry
Sandpoint, Idaho

Stuart Rubin responds:

I'm very excited to see that you're interested in my power-supply project. The ES1D diodes are available from several catalog and internet sources. A current list of General Semiconductor's distributors is available on-line at their website: http://www.gensemi.com/sales/na/nadist/distilocations.htm. I checked some of the places I've used in the past (Newark, Mouser, and Future), and they all have the parts in stock.

I have discovered, though, that changing the oscillation frequency higher or lower is the easiest solution to motorboating. I would suggest changing R19 (Rt in the National Semiconductor data sheet of the LM2524D) to about 10k. This will raise the clock frequency and hopefully eliminate the motorboat problem. Good luck!

ELECTRIC BASS

I think that your new format makes a good all-round publication for the audio enthusiast. The negative comments, mostly from the *Speaker Builder* folks, it seems, are misplaced. I am interested in tube audio, yet I am content with the new magazine in spite of the ratio of speaker to tube projects.

Bill Fitzmaurice, in his DR8 project (Nov. 2001), states that the preferred driver is the Carvin PS8. The current model Carvin sells may not be the same animal that Bill thinks. Carvin has, at the least, changed the impedance to 16Ω.

Mr. Stonjek states in his article ("The Ultra Fidelity Computer Sound System, Part 3," Nov. 2001, p. 74) that "Rock music rarely contains any really deep bass, the lowest note on the bass guitar being 70Hz." Having been a bass player for about 40 years, I can tell you this is incorrect. The lowest note on a four-string bass is 41Hz (E). On a five- and six-string bass it's around 30Hz (B). Some have an extra low string tuned to F#. Trust me, there is a lot of low bass in rock music!

I really like the recent articles on switching power supplies for tubes. What we really need is a lightweight switching supply for a 600W tube amp!

Paul Whiteman
Toronto, Canada

Bill Fitzmaurice responds:

I went to the Carvin site and found that they have indeed gone to a 16Ω impedance with their PS-8. I assume the reason for this is that they use a pair of PS-8s in their three-way T/S system and wanted to impedance-match that pair with the single 8Ω woofer. You can still use the PS-8 in the DR8, but with a 3dB sensitivity loss. Output capacity is the same, but it will take twice the power to get it. I'd recommend the Eminence Beta-8, which has appropriate specs and is widely available from the usual sources.

R.K. Stonjek responds:

Mr. Whiteman is absolutely right. My apologies for getting wires crossed. The lowest note on a regular guitar is 83Hz; regular electric bass is 41Hz. I note that most mid-priced commercial PA bass amp-speaker units roll off a lot higher than 41Hz—typically at 60–70Hz. This is obviously done to get the maximum power handling and efficiency required for stage performances.

A teacher of sound engineering told me that PA amplifiers usually have a filter to stop frequencies of <40Hz from passing to the speakers.

Note that the Ultra Fidelity Computer Sound System also rolls off at 40Hz (–3dB at 35Hz), making it perfect for rock music and especially the bass guitar!

VINTAGE RESTORATION

In response to Philippe Trolliet's letter ("A Moving Experience, Nov. '01, p. 84), by a strange coincidence, I am in possession of both a Thorens TD-124 and a Leak Stereo-30, which I obtained used some time ago. I have not bothered to restore the Leak unit, which is an old solid-state integrated amp that I have little interest in. The TD-124, however, needed little effort to restore to like-new condition and was a premium component when new. Mr. Trolliet is right; it is built like a tank.

He will not need a service manual to refurbish it. He can simply lift off the platter and thoroughly clean the bearing with solvent, being careful not to lose the ball thrust bearing at the bottom. Also pry off the C-ring securing the idler puck and then clean and relubricate all bearing surfaces.

I recommend replacing the idler puck, the drive belt from the motor pulley to the idler puck pulley, and maybe the socketed neon lamp that illuminates the strobe markings on the underside of the platter that indicate rotational speed. Thorens of America supports this product, and has replacement parts for it. As for the Leak amp, I can't help him with a repair manual.

The Thorens is the better of my two turntables, performs very well, and will probably outlast me. I wish him good luck in his endeavors.

Jerry Boncer
Carlsbad, Calif.

Philippe Trolliet responds:

Thank you very much for your kind letter. I also thank people who e-mailed me directly, particularly Mr. Guy Trépanier, who also wrote me a long letter.

The Leak amplifier, I know, is not only solid state, but includes germanium transistors. Its '50s look is charming, but I'm not sure it is of enough value to be repaired. Maybe it could serve as decor for a late '50s movie.

The Thorens TD 124 is another story. When I put it on the bench and plugged it in, the only thing that worked was the neon lamp! I looked at the motor, which was mechanically hard to turn. I put some fine oil into the bearing, turning it by hand, and, after a while, it started to turn, but was noisy.

Looking at its mechanical parts, I found the various oils and greases were more like hard jelly! So I decided to take the turntable apart, praying my memory would be good enough to reassemble it correctly! So I ended up with a collection of parts, which I individually cleaned and washed.

The riveted motor was a problem. I finally decided to drill out the rivets and took it apart. I even removed—noting the places and colors—all the wires from the motor, removed the iron and coils, and put the two halves with the bearings in an ultrasonic solvent bath. The oil was so hardened I had to leave it there for more than an hour before everything looked nice and clean.

I did another short cleaning in the bath with clean solvent, to be sure the felt around the bearings was thoroughly cleaned. I did that in the shop where I was working before I retired. And then I allowed them to dry for a full day.

I then put the two halves of the motor into a box just large enough to hold them and filled that with fine oil and left them for several days, so the bearings would be fully impregnated. I cleaned them with towels, removing the oil excess, putting the coils back, remounting the motor, keeping the two halves secured with screws and nuts in place of the rivets, and reconnecting the various wires. When I applied power, the motor turned, just silently and vibrationless—perfect.

I dismantled the platter bearing too, thoroughly degreasing and cleaning it, and then leaving the parts in an oil bath for several days. Remounting the rest, even the speed control, was far less daunting than I first imagined! It seems every part was logically designed, and I finally ended up with a complete turntable, with no extra parts left over!

The idler puck roller and the drive belt look OK, but I am pretty sure I will change them as you, and others, have suggested. And finally I found an old SME 3009 tonearm, so I redid the arm woodplate and installed the SME arm. The look is gorgeous. The only thing that shows age is the yellowish plastic of the bubble level.

To complete it, I must have my Ortofon SPU cartridge refurbished (I have kept that one since I was twenty!) and buy an original SME cartridge holder. I thank SME for a kind of service quite rare today. I e-mailed them various requests, and they quickly sent me all the necessary information and all old manuals in PDF form. Bravo!

I'm also sure the TD 124 will outlast me, and show far fewer signs of use than my old Bob Dylan long-playing records, which are about the same vintage!

PHONO PREAMP

Eric Barbour's use of an unusual tube in his preamp ("A Planar-Triode Phono Preamp," Nov. 2001, p. 18) grabbed my interest, but after a thorough reading of the article, I consider there are two potential problems with this design. This is from my experience—both personally and commercially—with a wide range of tube preamps.

Near the bottom of page 21 he says he found he could run the tube with a grounded cathode because it self-biased correctly (–0.5V) due to the grid current flowing in the 47k input resistor. Half a volt developed across 47k, indicating a grid current of just over 10μA, and this will be the cause of the problems I foresee:

1. The tube may correctly "contact bias" with the 47k resistor, but what happens when a cartridge is connected? No gain figures are offered for the overall unit, so it's not immediately apparent whether it is moving-magnet or moving-coil sensitive. A Shure V15 is a typical MM cartridge and a brand new one here measures at 975Ω, so if the grid current remains similar the bias voltage will drop to something like 10mV (10.6μA × 975R). This will almost certainly cause a major shift to the operation point of the tube, and may well take it out of its linear region.

2. Such a MM cartridge generates in the order of 5mV into a 47k load, which equates to approximately 0.1μA (100nF). If the tube produces 10μA of grid current and most of it flows through the cartridge windings, this is 100 times more than the signal current at which the cartridge is intended to operate—and there is a good chance it will become internally magnetized by this current flow. It's unlikely that 10μA will irreversibly damage a MM cartridge, but unless it is revitalized with a cartridge "demagnetizer" regularly, there is a good chance it will lose its sonic properties over time.

Moving-coil cartridges have less DC resistance than MM cartridges, down to an ohm or two, and while this will not make the bias situation effectively any worse than with a MM, I would not want a constant 10μA running through my prized IKEDA, vdHul, and Clearaudio masterpieces!

A polypropylene input blocking cap of 1μF or so will solve both of these problems, but I dislike using any caps in the signal path at such a sensitive location. Unless Eric can find a bias system that holds the tube at its correct operating point at all input resistances, and eliminates the seriously excessive grid current, I believe there are many better tubes for this job, including the 6922 I consider he slanders unjustly!

Allen Wright
Vacuum State Electronics

Eric Barbour responds:

Yes, the DC resistance of the phono cartridge can affect the idle point of the planar tube. However, it is a lot less than you think. To reflect my point, I took the original prototype of the preamp and measured the voltage on the anode of one planar triode with different grid resistances to ground. During these tests the plate supply voltage was 103V, and the preamp had been allowed to warm up for several minutes before testing.

With the input open-circuit (47kΩ resistor only):	*Vp=58.0V*
With the input connected to ground directly:	*Vp=41.6V*
With the input grounded through a 1kΩ resistor:	*Vp=42.6V*

I obtained similar results on the other channel in the preamp. No visible distortion on an oscilloscope trace occurred when I played a test LP through the preamp, using an old Empire turntable equipped with an Audio-Technica arm and ADC Mk. III cartridge. Even when playing loud peaks, the maximum voltage swing observed on the plate of the planar triode never exceeded about 2V p-p.

You see, Allen, the world is not always as simple as it seems. You are forgetting to include the effects of secondary emission, dissimilar-metals potential (sometimes called "contact potential"), gas effects, and the like. Tubes do not always behave strictly according to Kirchhoff's laws; they contain hot cathodes, which generate energy above and beyond that provided by the plate supply. So even when the grid and cathode of such a tube are grounded, the tube still biases as though there were a grid resistance.

This is especially noticeable in very high-mu planar triodes such as this one. You see extreme examples of this in transmitting tubes that use high-temperature thoriated filaments—often, to allow for such effects (depending on what resistance the grid sees externally), the grid bias needed to achieve a desired operating point must be made more positive than the plate curves of the tube indicate.

An old magazine article (ELECTRONICS, Dec. 1952, p. 192) describes how you can use a 12AU7 as an amplifier or oscillator with no plate supply of any kind. Some cheap 12AU7s of the era had so much of this effect that you could ground the plate resistance of the tube and still see a few volts DC at the plate—up to four volts, according to the article. Enough to use with low-level signals.

Oh, and by the way, your comments about grid current also don't hold water. With an input open-circuited, I measured 0.12V across its 47k grid resistor, giving a DC current through it of about 2.7μA. I then connected a 1kΩ load resistor to that input with a sensitive microammeter in series. If a current goes into the cartridge, it must be less than 1μA, because I could see only a slight indication on the ammeter.

Allen, I would appreciate it if you obtained a copy of Kohl's classic textbook, "Materials and Techniques for Vacuum Devices," and read chapter 17 carefully. I would be willing to provide the prototype of this preamp for testing to a third-party laboratory, if my personal tests do not sufficiently convince you. ❖

Panymo
from page 26

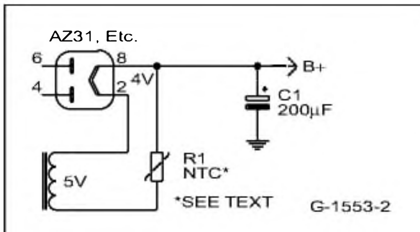


FIGURE 2: A simple and reliable 3.5-minute NTC delay.

age. If you hook an AZ31 in series with the NTC to a 5V winding, you have the perfect condition for this delay. ❖

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The LP track carries both stereo channels vs. alternating pits on the CD. The LP remains superior with "tight coupling for audio," $15; using an "OTL amplifier circuit," $35; driving efficient "Bass Horn" plans, $50. Visa, MC. ***STEEN AUDIO LAB***, PO Box 2185, Vancouver, BC V6B 3V7, CANADA.

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Test Tracks

Step carefully into the world of "audiophile" source material—I won't call all of it music. The following, however—to paraphrase Duke Ellington—is all good music: seven of my favorite test tracks, excellent in composition, performance, and production. These will make a top-notch system come alive, and help point out shortcomings in a less-than-optimal setup. Some of these discs are not in the catalog anymore, but with used CD sources such as eBay, they all are obtainable. Yes, they are all older material, but have stood the test of time.

1. *Schubert Impromptus*, Melvyn Tan, EMI CDC 749102 2, any track.
Mr. Tan is one of the world's best at performing these very difficult works. Using a period instrument, the sound is both spacious and intimate, but mostly real, with no ten-foot-wide piano sound as is common on many keyboard solo albums. It should sound as though Melvyn is right there with you, unamplified. Wonderful music, beautifully recorded.

2. *Dexter Blows Hot and Cool*, Dexter Gordon, Boplicity Records (Ace), CDBOP006, "Cry Me A River."
Carl Perkins (not that one) starts it off with a stunning piano intro, and then Dexter takes it and lays down one for the ages. You are in a New York City jazz club at midnight looking up through the smoky haze (it's 1955, folks), and that's how it should sound. The octave-to-octave balance should be perfect, and on the right system, it can sound as though Dexter is onstage with you—sax sound to die for. Jazz musicians and devotees who have never heard this cut are usually very moved by it. Ask your listeners if they even noticed that it was in mono.

3. *Dots Will Echo*, Self-Titled, Windham Hill (also issue #9 in the Boston Acoustics *Up Close* series), "The Rain."
A state-of-the-art recording of a better-than-normal garage band with in-your-face sonics and air—you can hear the synthesizer hiss and amplifier hum. Prepare to be very pleasantly surprised by a really good rock song you have never heard—well played and sung. The recording quality, however, sets a standard that is met on very few contemporary rock issues. Full-frequency detail and a gorgeous mix—you should be able to play the song through and follow any instrument or vocal part easily. The rest of the CD is equally listenable.

4. *Lyle Lovett and his Large Band*, Lyle Lovett, MCA 42263, any track.
Half big-band eclectic and half-country, there is a selection on this disc for everyone. Most people don't think about Lovett, because he cannot be put in a box or categorized. Neither can this disc, but it is well worth seeking out for its offbeat musical diversity, great original songs, and jaw-dropping production. The bass pumps, and the Large Band just cooks. If your toes aren't tappin', you have work to do.

5. *In The Square*, Patty Larkin, Philo CD PH-1136, "The Letter."
One incredibly talented woman who can just play the hell out of a flat top guitar. A live performance in Boston Common with Larkin on guitar accompanied by only an acoustic bass. The air, detail, and ambience of the hall space are present in the recording, and you will get a big, wide, multi-dimensional soundstage. The Box Tops probably never knew there would be this cover version, but I bet they all have copies.

6. *88 Elmira Street*, Danny Gatton, Elektra 9 61032-2, any track.
The late Danny Gatton was one of the masters, and this very well-recorded album is a testament to his musical prowess. On this CD, Danny uses his collection of vintage guitars and amps to get a wide variety of sounds, tossing off many styles and formats with his usual alacrity. His fluidity and warm tone remind me of the great Les Paul. If you really know your vintage equipment, you can name the axes Danny is using as the tracks progress.

The CD crackles with great production sound, and a couple cuts flat-out rock. The different musical venues explored will be a challenge for even the best systems—check that the overall signature of the mix is not lost among the tunes. Even though the songs are diverse, all cuts should have the same feel. RIP Danny.

7. *The Gift*, Liz Story, Windham Hill 11151-2, "Bring a Torch"...track #1.
Besides being a well-recorded CD of beautiful music, this disc has a Christmas theme. It is a 1994 issue, but I believe it is still a best-seller for Windham Hill. This may be the definitive natural-sounding recording of piano and acoustic bass, just perfect from the first note. The production is superb, the songs are all classics, and Liz Story is a gifted player.

This is a "lights out" CD—soundstage left to right, front to back, and up and down. You should be able to localize both instruments in front of you. The songs are deceptively simple, and Ms. Story makes it seem easy. If you're a piano player, try and play them.

Andrew Pennella
Stamford, Conn.

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1,000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to "Test Tracks", Audio Amateur, Inc., Box 876, Peterborough, NH 03458. We will pay a modest stipend to readers whose submissions are chosen for publication.