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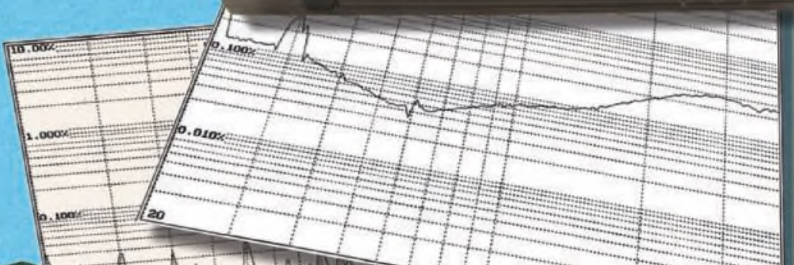

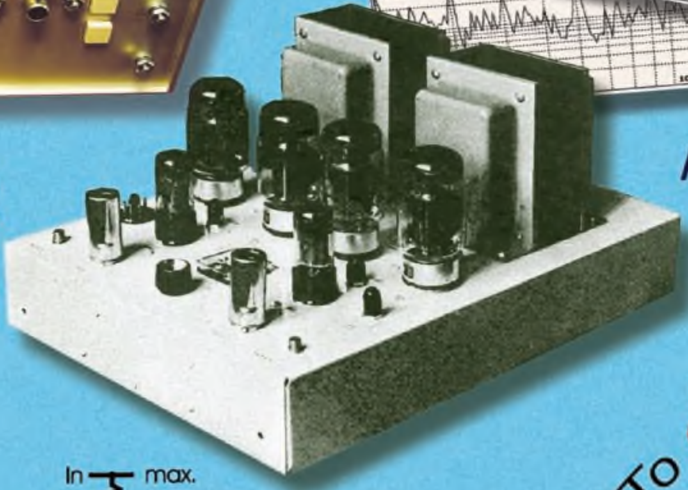

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R. K. Stonjek

How To Choose The Right Power Transformer
Pete Millett

Editorial

The Custom Touch

By Edward T. Dell, Jr.

Our town is remarkable for a number of reasons, despite its small size—about five thousand souls. Most remarkable to us, of course, is that it is home to some 30 periodicals, including two weekly newspapers. It has a world-class summer theater and is home to the MacDowell Colony, where forty artists of all kinds are in unfettered residence on a 400-acre preserve working uninterruptedly in isolated cottages, year round.

Most recently our local hospital added a first-class three-story Wellness Center, which offers a broad range of services for those recovering from physical trauma, as well as all forms of exercise and workout routines. Classes abound, two pools welcome, massage can be scheduled, and the dressing rooms offer every comfort including steam rooms and saunas. The planning survey indicated that membership might reach 300. Over a thousand have signed on and there is a waiting list.

The completeness and quality of this installation is what strikes the visitor most forcefully. Top quality and thoughtful planning are evident everywhere you look. The evident thoroughness of the planning is startling. They thought of everything. But not quite.

Peterborough has a significant elderly population, which has grown steadily partly because of a major retirement complex completed a few years ago. These older citizens make good use of the Center's facilities.

However, there was one thing the planners missed.

Those who have had the splendid pleasure of using a sauna will have noted that the benches are usually stepped, with the first at normal seat level above the slatted floor. But there is always a higher seat one level above the lower one. The steam is better at the upper level, of course; the sweat breaks out sooner.

For some of the older folks, however, that upper bench is hard to reach without help. One of these citizens noted the problem and went to the managers with an offer to help. A week later he presented them with two beautifully handcrafted handles, fashioned of oak stock, complete with bronze mounting screws. The maintenance staff mounted these on the rear wall, allowing those needing help a handy assist in accessing the top bench.

The volunteer made these handles in a home workshop. Thus a handcraft offering remedied a shortcoming in what is a wonderful, modern, and highly satisfying facility. But the architect and the engineers had not anticipated this particular, homely human need.

Yes, they had remembered all sorts of other needs. Large mirrors over wonderful sinks, lotions, hair dryers, tissue, cotton balls, hair spray, shampoo, body wash, stacks of fluffy towels, a water extractor for wet bathing suits—and much else. But no handle to hoist your aging body to the upper deck of the sauna bath.

Our human capacity to look creatively at what we are offered in today's world can see the small improvements that even the professionals may miss. How many of us have driven a car in which we find inconveniences any designer/engineer would have caught had s/he driven the vehicle for five minutes?

A whole new discipline named ergonomics has arisen to address this problem. Great strides have been made toward thinking ahead about such issues. Many products strike us as having origins where someone has given a lot of thought to the human experience in using them. But there is almost always some small, and sometimes large, omission, or some adjustment which only real-world use will uncover.

Such situations in audio equipment are exactly what amateurs are best at assessing. Most consumer products are, as we all realize, a result of a compromise of features to meet a marketing price point. Such compromises are not "mistakes" or even shortcomings, seen from the perspective of the art of the possible. But the history of this periodical includes a long list of improvements—starting with the first issue of *Audio Amateur* in 1970. A remarkable number of these have become part of the industry's standards.

There is really no substitute for creative interaction with the things we use every day. Such thought and response has been one of the bedrock foundations of the audio amateur enterprise.—**E.T.D.** ❖

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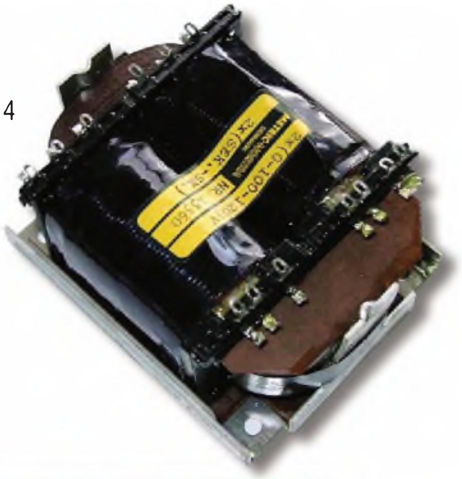

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> *The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.*
>
> JOHN STUART MILL

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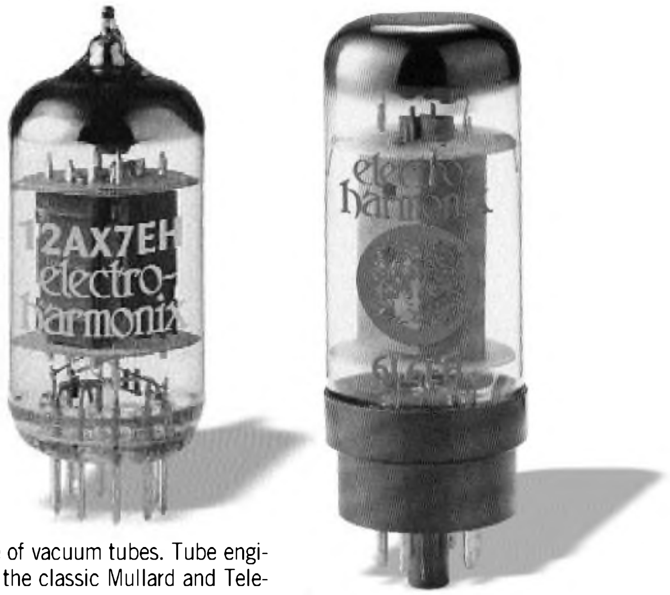
Audio News

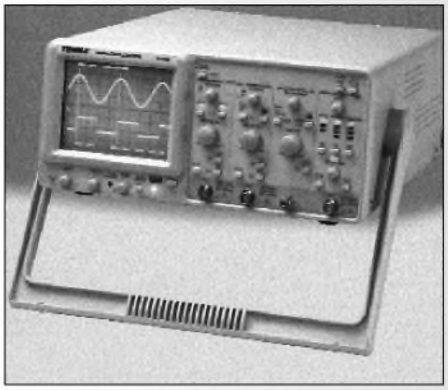


➲ ELECTRO-HARMONIX VACUUM TUBES

Electro-Harmonix added the 12AX7EH and the 6L6EH to its EH line of vacuum tubes. Tube engineers borrowed from the high-gain, low-noise, ultra-linear designs of the classic Mullard and Telefunken tube types and added second-order harmonics to produce the 12AX7EH, designed specifically for guitar amp applications. The 6L6EH, modeled after the vintage RCA 6L6GC "Blackplate," features larger plate dimensions and improved grid structure for increased power handling capabilities. Electro-Harmonix, 20 Cooper Square, New York, NY 10003, (212) 529-0466, (800) 633-5477, FAX (212) 529-0486, website: www.ehx.com, e-mail: info@ehx.com.

■ MADISOUND APPOINTMENT

Madisound was recently appointed North American Stocking Distributor for Morel (IL) ltd of Israel, and welcomes inquiries regarding the Morel product line. Madisound Speaker Components Inc., 8608 University Green, Box 44283, Madison, WI 53744, (608) 831-3433, FAX (608) 831-3771, website: www.Madisound.com, e-mail: info@madisound.com.



■ TENMA OSCILLOSCOPE

Tenma Test Equipment announces their Model 72-6820 100MHz Oscilloscope with Cursor Readout. It features dual-channel, dual-trace, and dual-time base operation, as well as delayed sweep mode with delay time from 1ms to 5s, cursor readout with seven measurement functions, ten memory front panel settings, 6″ rectangular CRT with internal graticule, trigger hold-off, and trigger lock. MCM Electronics, 650 Congress Park Drive, Centerville, OH 45459, (937) 434-0031, FAX (937) 434-9084, website: www.mcmelectronics.com, e-mail: talk@mcmelectronics.com.

■ RUSSOUND WEATHERPROOF SPEAKERS

Russound announces three new weatherproof speakers for their Outback series. The three models, the OB-6.1, OB-5.1, and OB-4.1, offer enhanced styling, improved sonic performance, and increased protection from the elements. Russound, 5 Forbes Road, Newmarket, NH 03857, (603) 659-5170, FAX (603) 659-5388, website: www.russound.com.

■ CARA 2.0

Rhintek Incorporated announces the availability of CARA 2.0 in the US and Canadian markets. CARA, Computer Aided Room Acoustics, is a popular German room acoustic modeling software package used by hobbyists and professionals to accurately determine the properties of sound waves in custom room designs. Dr. Ulrich Thomanek, German physicist and acoustic expert, developed the program using comprehensive mathematical models, incorporating both classical and modern theories of sound propagation, absorption, and reflection. Rhintek Computer Engineering, PO Box 220, 8835 Columbia 100 Pkwy, Columbia, MD 21045, (410) 730-2575, FAX (410) 730-5960, e-mail cara@rhintek.com, website: www.rhintek.com.

■ LAWSON TUBE MIKE

Lawson, Inc. introduces the Lawson L251 tube microphone. It features a faithful reproduction of the capsule used in the ELAM 251, a rare vintage mike renowned for its "airy" highs and warm, solid lows. The Lawson L251 also features a true toroidal output transformer, level handling capability, vanishing distortion, extended frequency response, low frequency contour control, 10dB pad, continuously variable patterns, and a cardioid-only position for a 3-4dB lower noise floor. Lawson, Inc., 2739 Larmon Dr., Nashville, TN 37204, (615) 269-5542, fax (615) 269-5745, e-mail: mail@LawsonMicrophones.com, website: www.LawsonMicrophones.com.

➲ VALVE AMPLIFIERS

Bachmann Electronik introduces two new valve amplifiers. These 200W amplifiers have six 6550 reflector tubes and input impedance of 10kΩ (symmetrical). They are ultralinear with two separated windings on the transformer. The inputs and driver tubes have regulated heater power supplies. The chassis are aluminum (10mm) and have special aluminum profiles on each side to aid cooling. Bachmann Elektronik, Werner Weberstrasse 9, CH-8630 Rüti, 055/240 40 87, FAX 055/240 42 27, website: www.bachelag.ch.

King-Size Quarter Horse Power Stereo Amplifier

In this age of solid-state devices, it is unusual to find a constructor who still resorts to high-quality commercial practice to produce a high-power amplifier. This unit seemed to have sufficient merit to warrant the full treatment, and all seasoned audio buffs know that superb performance can still be obtained from vacuum tubes. **By Robert M. Voss and Robert Ellis, Photographs by Edwin F. Meers**

This article appeared originally in Audio, *October 1966.*

Transistors may be here to stay, but tubes are not yet ready to be written off and placed in museums. The amplifier described here was designed and constructed to see just how far vacuum-tube design has progressed; it is a state-of-the-art device.

Before you read further, a word of caution: if you are interested in compact or miniaturized equipment, this is not for you. The amplifier and power supply are each built on 13″ × 17″ chassis, whose combined weight is over 80 lbs. The construction of the power supply (about 55 lbs) required installation of a small hydraulic jack underneath the workbench to avoid sag and ultimate collapse of the bench. Those who look upon the amplifier marvel greatly; their awe is brought about by the overall massiveness and, particularly, by the size of the power transformer, which by normal home standards is immense, although it would not hold a candle to some units in professional installations.

BASIC AMPLIFIER CIRCUIT

The basic amplifier circuit (*Fig. 1*) is, up to the output stages, an adaptation of a Genalex design, and a number of its features are deserving of further comment. First of all, the RC networks between C3 and C4 and R14 and R15 are there for the purpose of rolling off response at both frequency extremes to avoid any tendency toward motorboating or high-frequency oscillation at the resonant frequency of the output transformer and associated circuitry.

Incorporation of a driver stage between the phase-splitter and power-amplifier stages, in addition to supplying some needed gain, isolates the output stage from the shaping circuits and allows both output tubes to be driven by sources of the same impedance.

Circuits that drive the output stage directly from the paraphase inverter may overload asymmetrically, because, although in schematic form this type of inverter tends to look completely symmetrical, such is not at all the case. The top half of the inverter is a simple voltage amplifier with no voltage feedback, while the bottom (the half which is actually the inverter) is a voltage amplifier with sufficient feedback to reduce its gain to unity and its output impedance to a comparatively much lower value.

Because of the large amount of drive needed for the output stage, the plates of the drivers are fed from the same 650V source as is the output stage. Very large coupling capacitors are used to the output stage to hold rolloff and consequent phase shift to a point well below the frequency determined by the earlier shaping networks.

THE POWER SUPPLY

Aside from these few refinements, the circuit is very straightforward, and this leads to ease of adjustment and a large stability margin, in addition to general-

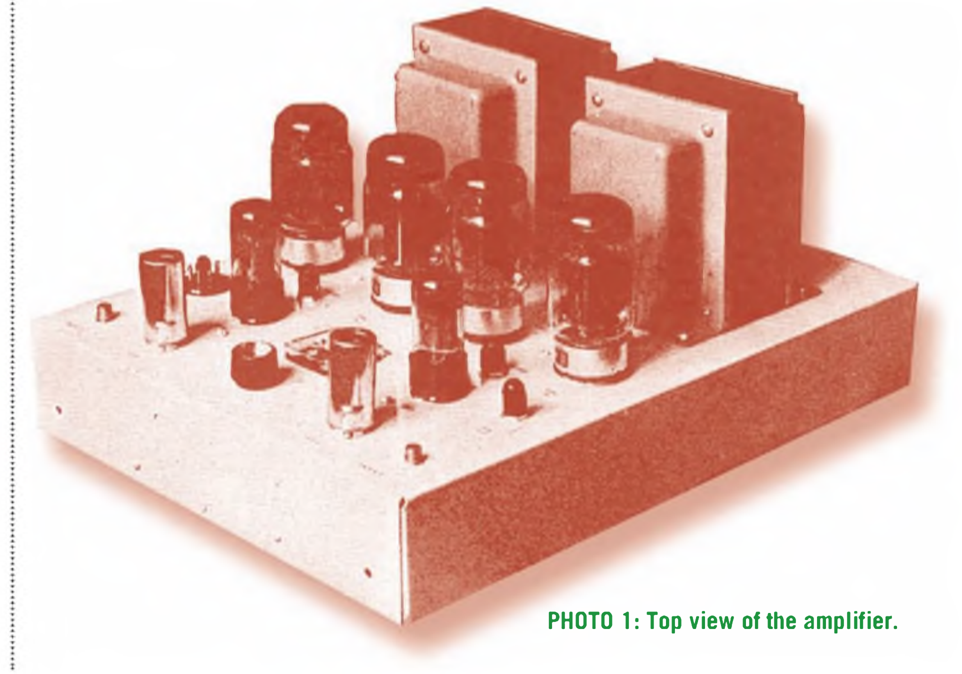


PHOTO 1: Top view of the amplifier.

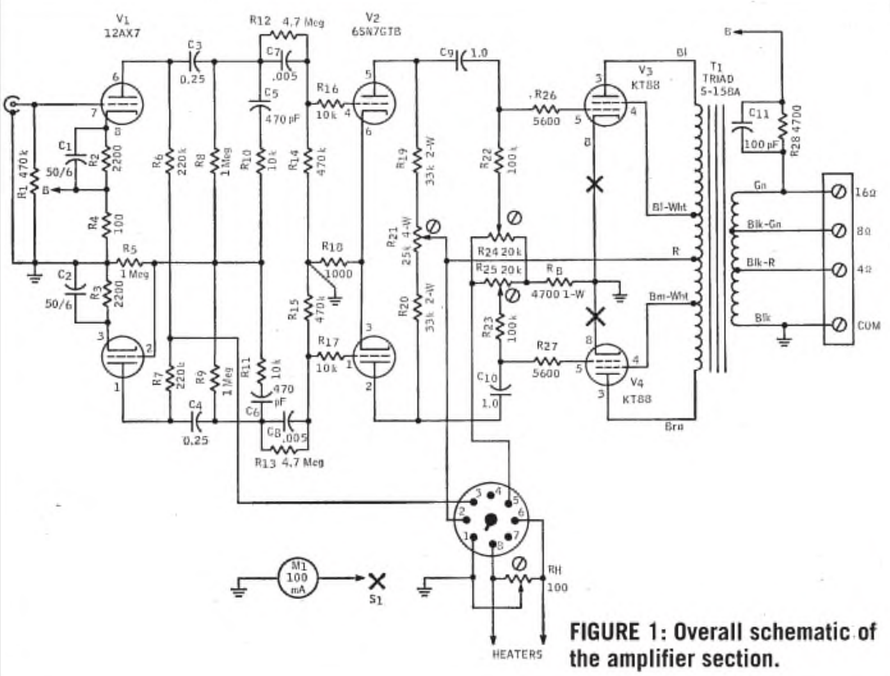


FIGURE 1: Overall schematic of the amplifier section.

ly superior performance. One possible source of difficulty, however, was the large swing in plate current drawn by each output tube from 50mA at zero signal to 150mA at 100W output per channel.

To accommodate this variation (equivalent to a total amplifier drain of about 250mA idling and 650mA at 200W out) we have used a silicon rectifier feeding a choke-input filter (*Fig. 2*). The rectifier (Sarkes-Tarzian S5162) is meant as a tube replacement, and was the most economical way of meeting the current and peak-inverse-voltage requirements. (Strings of lower-voltage diodes, with voltage-equalizing rc networks would have cost about the same, and are a makeshift solution at best.)

Those with a real aversion to solid-state components can use push-pull 5R4GYs in place of the silicon rectifier: this will reduce power output somewhat because of both larger initial voltage drop and poorer regulation. In addition, it will require redesign of the physical layout of the power supply since 5R4GYs cannot be expected to live very long underneath a chassis.

The power transformer and chokes are rated at 400mA continuous output, which means that the power supply runs at a comfortably low temperature, even after many hours of operation. If you are testing the amplifier for maximum output, or running it at high power for any other purpose, replace the ½A B+ fuse with 1A for the duration of the test, and avoid driving it at sustained full power for more than ten minutes or so.

This does not mean that the power supply has an inadequate safety margin: quite the contrary, the amplifier is quite suited to fill your living room (or major league baseball park) with large amounts of sound for indefinite periods for a long time. When initially wiring the power supply, test the filament windings to make sure they are in phase rather than bucking by momentarily touching the red leads (*thick* ones, not the thin red high-voltage leads) to the green; if you get a fat spark (only 6V here, no shock hazard) they are out of phase; reverse one set of leads and solder.

BIAS-SUPPLY TRANSFORMER

Since the power transformer has no bias winding, a separate transformer is used for the bias supply. Any isolation transformer will do as long as it can deliver about 15mA; the exact output voltage is not critical since it is adjustable at each tube. If you can't locate a packaged bridge rectifier, just about any four silicon diodes will work just as well.

Photo 1 is the top view of the amplifier; each channel is on one side of

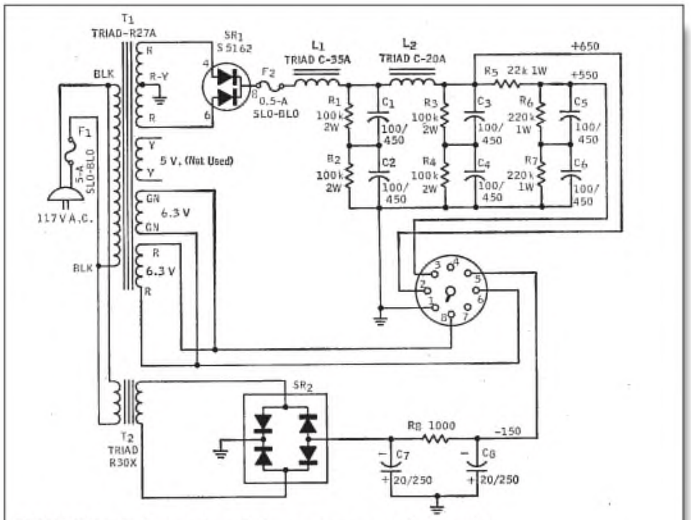


FIGURE 2: Schematic of the power-supply section.

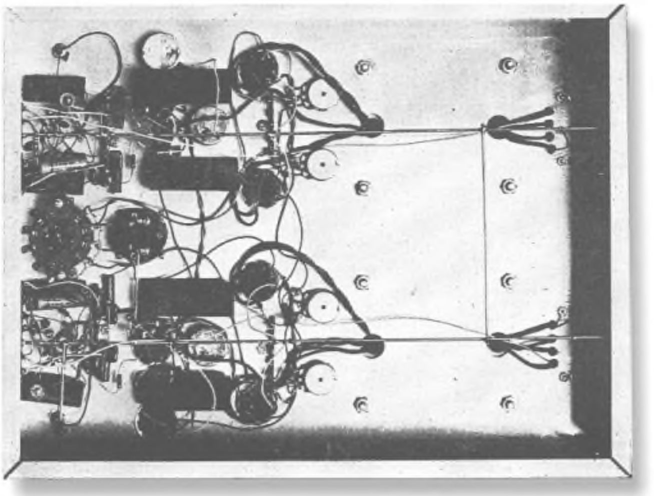


PHOTO 2: Under-chassis view of the amplifier, showing relatively neat construction.

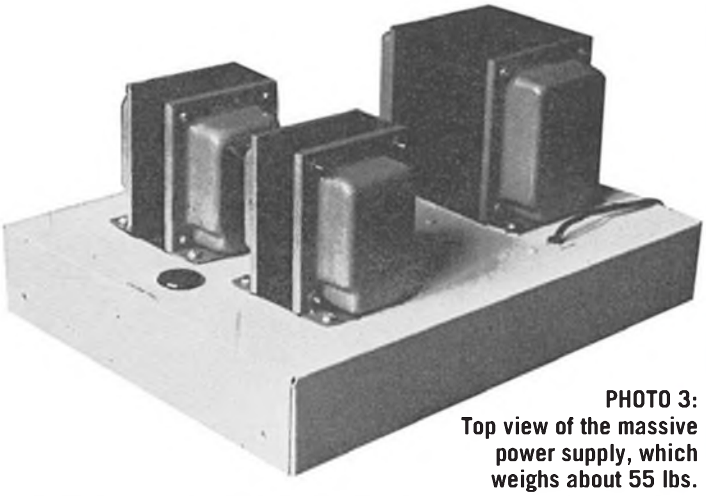


PHOTO 3: Top view of the massive power supply, which weighs about 55 lbs.

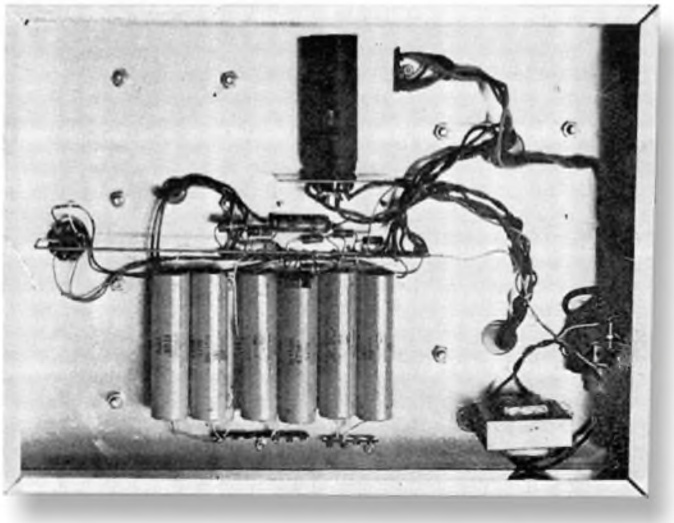


PHOTO 4: Underside of the power-supply chassis is similarly neat and uncluttered.

TABLE 1
PARTS LIST

AMPLIFIER **(TWO OF EACH COMPONENT UNLESS OTHERWISE SPECIFIED)**	
C1, C2	50µF, 6V, electrolytic, Sprague TE 1100
C3, C4	0.25µF, 600V, paper, Sprague 6TM-P25
C5, C6	470pF, Sprague 47192
C7, C8	.005µF, paper, Sprague 6TM-D50
C9, C10	1.0µF, paper, Sprague 6TM-M1
C11	100pF, ceramic, Sprague 10TCC-T10
M1	100mA meter, 1 only, optional
R1, R14, R15	470kΩ, ½W (all resistors 5%)
R2, R3	2200Ω, 1W
R4	100Ω, ½W
R5, R8, R9	1MΩ, ½W
R6, R7	220kΩ, ½W
R10, R11, R16, R17	10,000Ω, ½W
R12, R13	4.7MΩ, ½W
R18	1000Ω, ½W
R19, R20	33,000Ω, ½W
R21	25,000Ω, 4W potentiometer
R22, R23	100kΩ, ½W
R24, R25	20,000Ω potentiometer
R26, R27	5600Ω, ½W
R28	4700Ω, ½W
Rb	4700Ω, 1W (1 only, common to both channels)
Rh	100Ω potentiometer (1 only)
S1	Meter switch, Mallory 1400L (1 only, optional)
T1	100W, tapped screen, 4500Ω output transformer, Triad S-158-A)
V1	12AX7
V2	6SN7GTB
V3, V4	KT88
	Chassis, jacks, sockets, hardware, etc.
POWER SUPPLY	
C1–C6	100µF, 450V electrolytic, Sprague 1718
C7, C8	20µF, 250V, electrolytic, Sprague 1508
F1	5A, slow-blow fuse
F2	0.5A, slow-blow fuse
L1	Triad C-35A swinging choke, 20-4H, 40–400mA
L2	Triad C-20A choke, 6H, 400mA
R1–R4	100kΩ, 2W (all 10%)
R5	22,000Ω, 1W
R6, R7	220kΩ, ½W
R8	1000Ω, ½W
SR1	tube-replacement type silicon rectifier, Sarkes Tarzian S5162
SR2	Bridge-type silicon rectifier
T1	Triad R-27A
T2	Triad R-30X
	Chassis, line cord, fuse mountings, power cable, hardware, and so on

the chassis, with the common components (meter and switch) in the center. Hidden behind the output transformers are the output terminal strips. The male octal power connector is on the far side of the chassis, the hum balance control on the near side. The two other capped potentiometers (between the 6SN7s and pairs of KT88s) are the AC balance controls.

Photo 2 shows what the amplifier looks like underneath. There are plenty of components used in the input stages, but the large chassis leaves plenty of room. Just make sure nothing is touching that isn't supposed to touch. The locations of the meter and switch are obvious. The four large capacitors are the coupling to the output stage; the balance controls are between them. The four bias pots are on the other side of the output tube sockets. Two bus bars are used, to which all grounds are connected; they are tied together and grounded to *one* of the input sockets; this is the only chassis ground.

The top of the power supply is shown in *Photo 3*; the only components here are the power transformer and chokes, the octal power connector, and the line cord. More is to be seen underneath (*Photo 4*). On the lower right are the components of the bias supply: transformer and (partly hidden) R8, R2, C7, and C8. The B+ rectifier is mounted on an angle; there is no need to put it on top of the chassis since its life is indefinite. Underneath the rectifier is the high-voltage fuse; underneath the fuse are all of the resistors and capacitors in the high-voltage circuit.

The output of the power supply is fed to an octal socket; a matching cable is used to deliver power to the amplifier. Hookup is by means of a six-conductor cable. Since the total filament drain is over 8A, the two wires carrying filament current should be at least #18; the others can be #22. We used Belden 8446, whose #22 conductors are rated at only 200V, but as in other applications, we have had no difficulty with insulation breakdown at many times this rating. Perhaps lacing six single, well-insulated wires might be a more elegant solution; in any case keep the length to no more than is necessary.

TESTING AND OPERATION

Before connecting the amplifier to the power supply, make a preliminary resistance check from pins 2, 3, 5, 6, and 8 of the male octal power connector on the amplifier to ground. Pins 2 and 3 should be wide open (infinite resistance), pin 5 about 10k, and pins 6 and 8 varying from 100Ω to zero as the hum balance control is rotated.

When measuring from pin 5, turn all four bias controls from one end of rotation to the other and make sure this has no effect on the resistance to ground at pin 5. Failure to make this check cost us a KT88: a sliver of wire caught under one of the bias pots removed all bias from that tube, and before we could get to the power cord, completely stripped the cathode of the tube. Any such short would show up as a resistance variation in the above check.

Assuming that the amplifier checks out, plug in the power supply with the amplifier not yet connected (the dimming of lights is normal) and measure the output voltages (B+ should be about 700V with no load). If everything is normal, unplug the power supply and connect it to the amplifier. With the 12AX7s and 6SN7s in their sockets (no output tubes yet) plug the power supply in again and make sure the tubes light up.

If you wish to make plate voltage checks here, keep in mind that they will be a bit higher than normal. Connect a signal generator to one of the inputs (about 1kHz) and adjust it for about 20V at pin 5 of either of the output tubes of that amplifier. Now adjust R21 so that the signal will be the same at pin 5 for both output tubes. Repeat the adjustment for the other amplifier; we used caps on the balance controls since this is a semipermanent adjustment which cannot be checked on the meter.

Now connect each side of the amplifier to a dummy load (or a common load to the paralleled outputs), turn each bias control fully *negative* (counterclockwise in our amplifier) and plug in the output tubes. To check output-tube current, we used a 100mA meter with a switch which inserts the meter into any one cathode circuit or cuts it out of the circuit. If you see no purpose in building into the amplifier a meter which will be used rarely, make some other provision for measuring cathode current (such as closed-circuit phone jacks).

In any case, with the controls supplying full bias, the tubes should draw little or no current. After they have warmed up, slowly adjust each bias pot for 50mA cathode current. Since they interact, go back and adjust each pot a couple of times until all of the tubes are drawing the same 50mA. After about ten hours of operation, repeat the adjustments to compensate for initial aging.

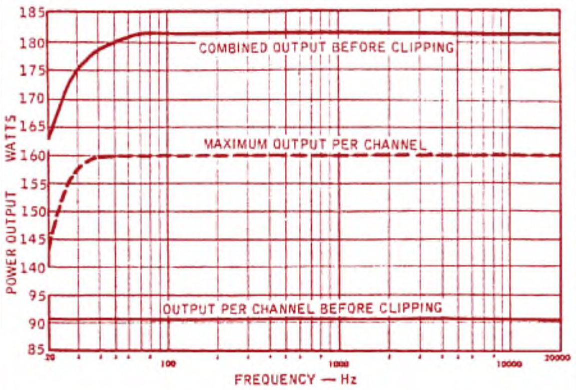


FIGURE 3: Curves of output of each individual section before clipping, and of both sections together, along with maximum output per channel, all versus frequency.

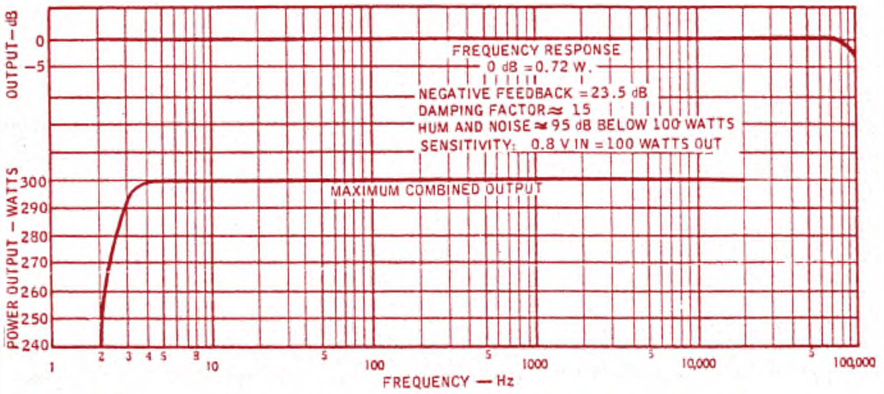


FIGURE 4: Frequency-response curves for 0.72W per channel, and for maximum combined output of both channels.

Before connecting the amplifier to your speaker system, check each side with a voltmeter (dummy load attached) to make sure it is not oscillating; if you have made a wiring error causing the feedback phase to be reversed, the amplifier would be producing about 160W of oscillation, which you will need to correct before it sends your speaker up in smoke.

Adjustment of the amplifier is now completed. If you have the test equipment, you might choose to trim balance controls R21 for minimum IM distortion; adjustment for equal output is sufficient for anything less than laboratory measurement uses.

PERFORMANCE

Frequency response, power output, and other data are given in *Figs. 3* and *4*. No distortion measurements are given since these have been shown (by transistor research, to a great extent) to have little to do with sound quality. Suffice to say that square waves at 1kHz and 10kHz are closer to the original than those coming out of any other amplifier we have seen. 20Hz square waves show a great deal of tilt because of subsonic rolloff, and high-frequency square waves are rounded because of the high-frequency rolloff.

Note the large difference between maximum output before clipping and absolute maximum output in both the single channel and combined measurements. This is indicative of the symmetrical, gradual clipping which turns sine waves almost into square waves before sides are deformed. Maximum power without audible distortion is about 100W, music power about 115W (each channel), and the total peaks in the half-kilowatt region.

Frequency response is dead flat throughout the spectrum, another 15dB of feedback will produce neither oscillation nor detectable sound improvement, the noise is inaudible, the gain is very high (input gain controls may be needed in some applications), and the damping factor (remember that?) is good for most speakers. The amplifier has been tested into a capacitive load approximating an electrostatic speaker and shows no signs of instability or sound degradation.

A word of caution: if you have never experimented with anything this big, be very careful about plugging and unplugging leads. An open ground could easily mean 160W at 60Hz fed to your speaker. Also, switching clicks may contain so much energy that they may cause your house lights to flicker. This does not mean oscillation; it simply means that the amplifier can reproduce a sharp thump fed into it at the equivalent of many watts of power. If you have chronically noisy switches on your preamp, you might consider an infrasonic filter.

RATIONALE

Why, to sum things up, should you go out of your way to listen to an amplifier whose only claim to fame seems to be the great gobs of power it can produce, an amplifier whose flat frequency response, low noise, and high gain can be duplicated by most of the commercial units on the market today, an amplifier which is hardly easy on the back muscles and which, if mistreated, may vent its rage by destroying any speaker made in a fraction of a second?

The reason is sound. For pure sound quality, this amplifier sounds noticeably better than anything we have ever heard. Using it in place of a respected 70-watter elicited immediate comments from listeners. Not only is it exceptionally clean at normal levels, but the feeling of completely unlimited power handling capacity is a new experience.

We invariably listen to the amplifier at considerably higher levels than before, but it is a new type of loudness, absolutely free of listener fatigue and truly approaching live sound. This is an amplifier that can be relied upon never to call attention to itself (sonically, at least) no matter what it is called upon to do; it will always be a strong link in the chain of sound reproduction. ❖

Power Transformers for Audio Equipment

Discover the different types of power transformers, and why you might choose to use each type in audio equipment. **By Pete Millett**

Newcomers to the fine art of audio equipment design (you don't really believe it's a science, do you?) might think that any power transformer providing the needed voltage and current ratings will work fine in their design. As you may have learned from experience, this is far from the truth. This article addresses issues with low-frequency (50–60Hz) line transformers, but many of the same issues apply to audio-frequency transformers as well.

HOW TRANSFORMERS WORK: A QUICK REVIEW

Grossly oversimplified, a transformer works by converting an AC current to a varying magnetic field, and then back into an AC current. Coiling wire into a "winding" and passing current through it produces the magnetic field. Conversely, the field passing through another winding induces a current in it, which is used to drive the load.

In a power transformer, the "primary" winding is driven with AC line voltage—the power that comes out of the wall. The voltages required for the rest of the system are generated in "secondary" windings. All the windings are placed on a "core" made of an iron alloy. This is done because the permeability, or magnetic conductance, of iron is much higher than that of air, which allows a transformer to work much more efficiently. There are many different ways to build the core using different materials, processes, and shapes. I'll describe a number of the more common types in detail (*Photo 1*) a little later.

MAGNETICALLY INDUCED NOISE AND STRAY FLUX

If you've built much audio equipment, chances are that at one time or another you've run into a problem with line-frequency noise getting into the audio signal path. Sometimes this problem is simply the result of not enough filtering or regulation of the DC power supply, or perhaps a ground loop in an input circuit. But often, magnetic coupling from the AC power transformer is the culprit.

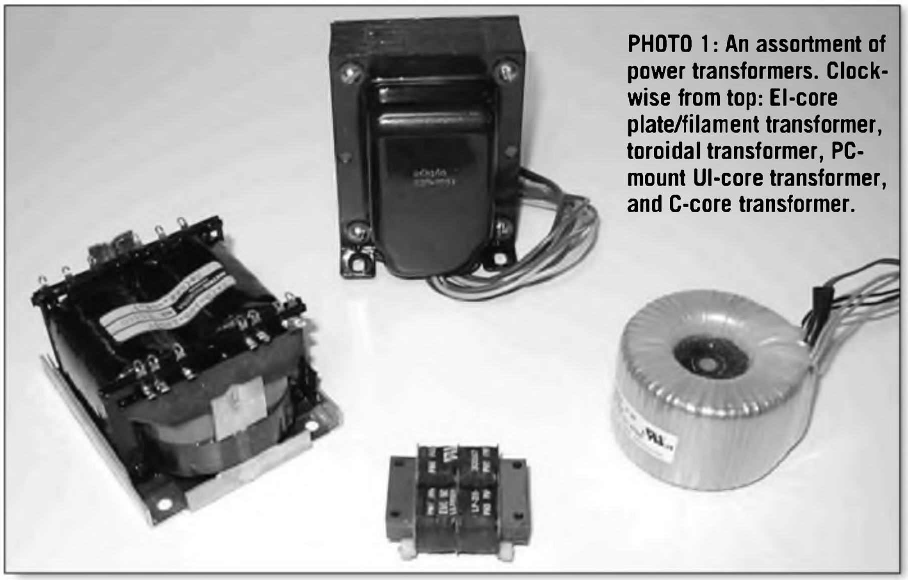


PHOTO 1: An assortment of power transformers. Clockwise from top: EI-core plate/filament transformer, toroidal transformer, PC-mount UI-core transformer, and C-core transformer.

In a perfect transformer, the entire magnetic field generated by the primary winding would be contained entirely within the transformer and pass through the secondary winding(s). Of course, nothing is perfect, and there is always at least some part of the magnetic field that escapes from the transformer. This field, called "stray flux," is a primary concern in the selection of a transformer for audio equipment.

If an AC magnetic field that leaks from the power transformer intersects a wire (or PCB trace), it will induce a small current into that wire, just as if it were a secondary winding on the transformer. The resultant noise voltage produced is generally very small, but in audio equipment, even a few millivolts of noise in a sensitive circuit can be audible. Tube circuits, in particular, with their high impedances, are very susceptible to picking up noise from power transformer stray flux.

Stray-flux problems usually manifest themselves as a line-frequency noise—more of a buzz than a hum—that is unrelated to power-supply ripple or filtering. The noise waveform on an oscilloscope is usually not sinusoidal—instead, it looks more like the combined waveform of the current drawn from the secondary(s), which has a much sharper peak, resulting from the filter capacitors charging on the rising edge of the AC waveform. You can often see large spikes that correspond to the reverse recovery current in solid-state rectifiers, which is probably a subject worthy of an article itself.

Photos 2 and *3* show oscilloscope patterns of stray-flux noise in a tube headphone amplifier, which I'll talk

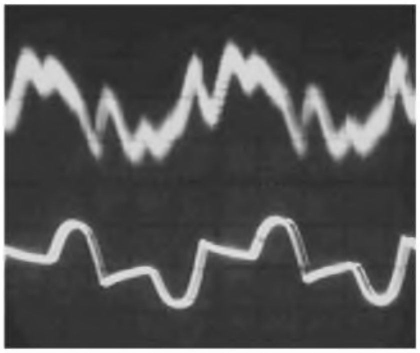


PHOTO 2: Oscilloscope pattern showing output noise (top trace) and the current through the plate winding, using the vertical EI-core transformer.

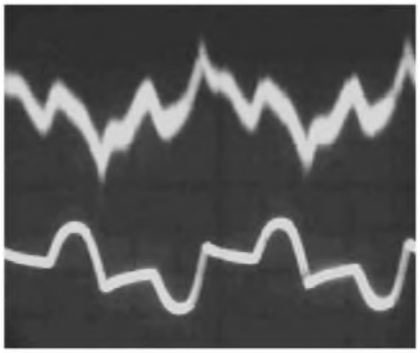


PHOTO 3: Oscilloscope pattern showing output noise (top trace) and the current through the plate winding, using the horizontal EI-core transformer. Note the spikes from diode reverse recovery.

about more later.

The easiest way to see whether you have a stray-flux problem is to move the transformer away from the rest of the circuitry. Since the strength of a magnetic field drops rapidly with increasing distance from its source, often moving the transformer only an inch or two will diminish the noise dramatically.

Sometimes—especially in power amplifiers—just keeping the transformer away from low-level circuits is an effective method of dealing with stray flux. But in other situations, as in a phono preamp, the design of the transformer itself needs to be addressed to provide acceptably low noise in the finished product.

CORE TYPE—THE MANY DIFFERENT WAYS TO BUILD A TRANSFORMER

You can build low-frequency power transformers in many different shapes and configurations. I'll discuss a number of them in detail, starting with the most commonly used types, describing their construction and their suitability for use in audio equipment.

Almost all power transformers share some common characteristics. Typically, they are wound on cores made from thin sections of an iron alloy, usually a type of steel made just for this application. Thin sections are used instead of a solid piece to prevent currents from being induced in the core itself—iron, after all, is an electrical conductor as well as a conductor of magnetic flux.

Different physical configurations and manufacturing methods for transformers have evolved over the years in an effort to design a better product. One goal is to design a transformer that is nearly 100 percent efficient—meaning that all the energy in the primary winding is coupled to the secondary, and is not wasted by heating up the core or windings, or leaking magnetic flux outside the core. Another goal is to design transformers that are inexpensive to manufacture. As you might expect, these two goals are generally at odds, and the better the transformer is, the more expensive it is to build.

The EI-Core Transformer

The most common type of AC power transformer is called an EI-core transformer, because the laminated iron core that it's wound on—before assembly—looks just like the letters "E" and "I." *Photo 4* shows an EI-core transformer with the outer casing removed.

PHOTO 4: A typical EI-core transformer with the case removed.

In an EI-core transformer, the windings are wound around the center leg of the "E"-shaped core piece, and the "I"-shaped part is joined to the "E" to form a closed magnetic path. In reality, the thin, stamped "E" and "I" laminations are stacked up in alternating directions and assembled with a form, or "bobbin," containing the pre-made windings. The joint between the "E" and "I" is alternated between the two sides of the transformer to give the core greater mechanical strength.

While the EI transformer is the least expensive and most common type of transformer, it is also one of the worst in terms of its stray flux. The problem is all those "E" to "I" interfaces—there is always an air gap, no matter how tiny, between the laminations. Each of these gaps (there are three for each lamination), visible in the close-up (*Photo 5*), provides an opportunity for magnetic flux to leave the core.

Several "fixes" applied to EI transformers help lower the stray flux. The most common is to place a conductive "flux band" around the transformer. This band, usually made of a thin sheet of copper, is wound around the transformer in the same orientation as the windings, but completely outside the core. It is soldered together to form a continuous loop.

The flux band works by acting as a shorted turn around the outside of the transformer. Any flux lines that cut through the band induce an eddy current that produces an opposite magnetic field, which tends to cancel out the original flux.

PHOTO 5: Close-up of the lamination gaps in an EI-core transformer.

Another often-used fix is to weld an iron strap around the outside perimeter of the core. This tends to contain, rather than cancel, the stray flux, since the iron strap has a much higher permeability than the air around it. Similarly, the entire transformer can be contained, or potted, inside a ferrous metal can.

The Toroidal Transformer

The toroidal transformer, or "toroid," is a familiar sight inside high-end audio equipment. The toroid looks like a doughnut, with windings equally spaced around the diameter of the transformer (*Photo 6*). Toroids are also available potted inside metal or plastic cans, or molded inside plastic resin and equipped with pins to mount directly onto a PC board.

Like the EI-core transformer, the core of a toroidal transformer is made of an iron alloy, but instead of being composed of multiple, stacked laminations, it is wound from a single strip of metal, much like a roll of tape. The fact that there are no discontinuities in the core makes the toroid very efficient and reduces stray flux to around 10% of that of a comparable EI transformer. Toroidal transformers still leak magnetic fields, mostly because the windings are not symmetrical—the wires are spaced farther apart on the outer diameter of the core than the inner.

Generally, toroidal transformers are smaller (by up to 50%), quieter (magnetically and sonically), and more efficient (95% versus 80%) than comparable EI-core transformers. If they are so much better, you might wonder, why doesn't everybody use them? The answer, as

PHOTO 6: A small toroidal transformer for a tube headphone amplifier.

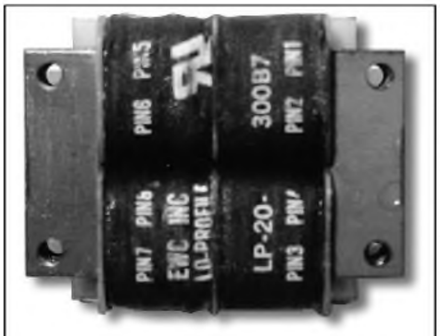


PHOTO 7: A small PCB-mount "semi-toroidal" UI-core transformer.

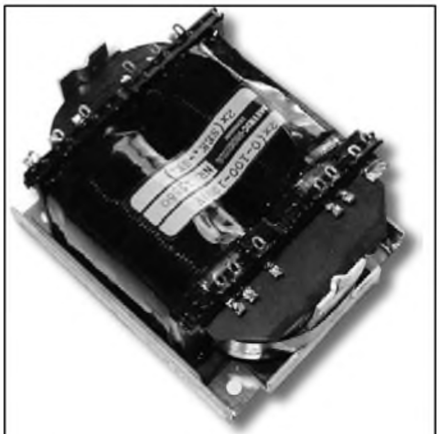


PHOTO 8: A C-core power transformer.

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you might guess, is because at power levels under 500W, they're much more expensive. This cost premium is mostly because they are much more difficult to manufacture.

Since there is no "open end" on a toroidal core, you cannot wind the wire onto a bobbin and slide it onto the core, as is done for an EI transformer. For each turn of wire, the entire length of the winding must pass through the hole in the core, making the toroidal transformer much more difficult and time-consuming to wind.

The UI-Core Transformer

UI cores are similar to EI cores, without the middle leg of the "E." Also like EI cores, UI cores are made of stacked laminations, so they suffer from the same gaps in the magnetic path that EI cores do. These discontinuities result in flux escaping from the gaps in the core.

UI-core transformers are sometimes made with the windings all on one leg of the core (a common construction for very high voltage transformers), but usually the primary and secondary windings are placed on separate bobbins on opposite sides of the core. This results in a "semi-toroidal" construction, with the windings oriented to help cancel out any stray magnetic fields.

The most common application of this type of core for power transformers in audio equipment is in small PCB-mount transformers (*Photo 7*).

A small PCB-mount UI-core transformer is often a better choice than a conventional EI-core transformer for low-power audio equipment use. These inexpensive, compact types tend to radiate a little less magnetic interference than a comparable EI-core transformer. They still have much more stray flux than a toroidal transformer, so keep them a good distance from low-level circuits.

C-Core Transformers

C-core transformers are made on a core that is wound from a single strip of material, like a toroid. The core is wound with two straight sides, so it is shaped more like an oval than a circle. After the core is wound and impregnated with a glue to hold it together, it is cut into two pieces, each in the shape of the letter "C." This allows assembly of the pre-made windings onto the core, which is then put back together.

C-core transformers can be made with a single core (one magnetic "loop") or with two (often called a "double C-core transformer"). Like UI-core transformers, they can also be constructed with the windings wound all on one side, or on two opposite sides of the core.

As far as performance is concerned, you can consider C-core transformers in between toroidal transformers and EI-core transformers. Though they still have a break in the magnetic path, they have only two, which can be minimized with careful finishing of the core. The wound construction of the core also results in higher magnetic efficiency than a stacked core. When constructed with windings on opposite sides of the core, the symmetrical construction helps to cancel stray magnetic fields as well.

Although C-core audio transformers for tube amplifiers have had fairly wide acceptance (at least in Europe and Japan), C-core power transformers are not common in audio equipment. *Photo 8* shows a medium-power C-core power transformer, a single-loop transformer with windings on both legs of the core.

R-Core Transformers

A more recent development is the "R" core, which you can think of as a cross between the C-core and a true toroidal core. R-cores are wound from a continuous strip of metal and are formed into a shape with two straight sides, like a C-core. Instead of using a constant-width strip of metal, the R-core is wound from a strip of varying width, so that the finished core winds up with a circular cross section. Unlike the C-core, the R-core is not cut to assemble the windings—instead, the windings are done on a bobbin that is assembled over the circular cross section of the core, then rotated to wind on the wire.

The R-core transformer is nearly as good as the toroid in terms of stray flux. It has an advantage over the toroid because the turns of wire are spaced equally around the core, since they are wound on a straight section of the core. The windings on a toroid are spaced closer on the inside of the core, and wider on the outside. The fact that the winding is done on a bobbin on a straight section of the core makes the R-core much easier to wind, lowering the cost of the finished transformer.

R-core transformers are currently used in mid- to high-end consumer electronics equipment from Japan. In the US, they are still quite uncommon and almost as expensive as toroids, but I hope this will change as they become more popular here. The R-core transformer has the potential to become the predominant choice for use in audio

PHOTO 9: Tube headphone amplifier, with vertical EI-core power transformer (at right). The output noise caused by the power transformer flux leakage was –54dBm.

PHOTO 10: The headphone amplifier, with the EI-core transformer mounted horizontally on a bracket. Output noise due to flux leakage of this configuration was –58dBm.

PHOTO 11: The final solution, using a toroidal transformer on a steel bracket. The resultant output noise due to the transformer was unmeasurable.

TABLE 1

COMPARISON OF EI-CORE AND TOROIDAL TRANSFORMERS IN THE TUBE HEADPHONE AMPLIFIER

	EI, VERTICAL	EI, HORIZONTAL	TOROIDAL
Cost (each, @ 25 pcs.)	$22	$26	$48
Flux-leakage noise	–54dBm	–58dBm	Unmeasurable

equipment, providing all of the benefits of toroidal transformers at a lower cost.

EI VERSUS TOROIDAL TRANSFORMERS—A CASE STUDY

I did some real-world comparisons of EI and toroidal transformers in the recent design of a tube headphone amplifier. I had custom EI-core and toroidal power transformers built for the amplifier, both with the same ratings. The transformers have two secondary windings—one 6.3V filament winding and one plate voltage winding.

I also considered an R-core transformer, but I could find only one or two vendors in the US who could produce such a transformer, and their prices and lead times were unacceptable (it is possible that they were actually going to have them manufactured in Asia, and import them).

My first approach was to mount the EI transformer on the amplifier chassis in its "normal" vertical orientation, (*Photo 9*). As soon as the amplifier warmed up, it was apparent that there was a problem: a noticeable line-frequency buzz coming from both channels.

An oscilloscope trace of the noise waveform using the vertically mounted transformer is shown in *Photo 2*. The bottom waveform is the current in the transformer plate winding (measured with a small current transformer), and the upper trace is the noise observed at the amplifier output. Note the rough correlation between the two waveforms. The noise waveform is quite complex—it is far from sinusoidal, and it does not resemble power-supply ripple.

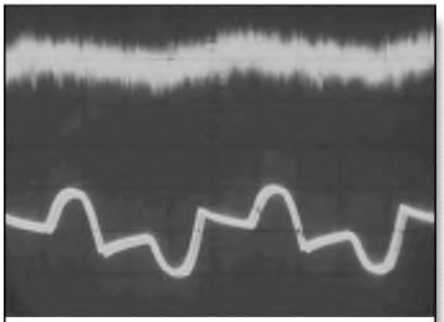

PHOTO 12: Oscilloscope pattern showing output noise (top trace) and the current through the plate winding, using the toroidal transformer. The noise from the transformer stray flux is gone.

Moving the transformer away from the PCB made the noise disappear, so it was evident that there was a stray-flux problem. If I moved the transformer about 2″ away, the noise diminished to the point of inaudibility—and if I moved it closer to the input stage of the amplifier, the noise became much worse.

I found by experiment that if I placed the transformer in a horizontal orientation (*Photo 10*), the noise was much less noticeable. Apparently, the stray flux from an EI transformer radiates the most in the plane of the windings. Leaving the transformer vertical and rotating it made little difference in the noise level.

Photo 3 shows an oscilloscope pattern of the noise with the transformer mounted horizontally. The amplitude of the noise is slightly lower than it was with the transformer mounted vertically, and somewhat different in shape. Note the large spikes that correspond to the diode reverse recovery current—and the diodes used were soft-recovery diodes! It is interesting that these spikes were not so evident in the vertical orientation.

Because the noise level was still unacceptable, I experimented with all kinds of magnetic and electrostatic shielding around the transformer (including a flux band), but met with little success. No matter what I did, the EI transformer caused unacceptable levels of stray-flux induced noise into the low-level section of the circuit. The only way I could reduce the noise level was to move the transformer, which wasn't an option, since the enclosure was already designed and built.

Though I preferred not to spend the money for a toroidal transformer in production, I was running out of ideas, so I turned my attention to the toroid. The transformer was mounted using a steel bracket into the same space as the EI transformer (*Photo 11*).

In the headphones, the difference between the transformers was immediately obvious. With the toroidal transformer, there was no noise to be heard. As shown in *Photo 12*, the remaining (inaudible) noise was composed of a combination of a small 60Hz component (interestingly, not 120Hz as you would expect from power-supply ripple) and the random noise generated by the components. Some radio-frequency noise was present as well, probably originating from a nearby AM broadcast transmitter. This noise was below the –60dBm level that I could meaningfully measure with my test equipment.

A comparison of the three transformer options is summarized in *Table 1*. The total cost of the three options shown include mounting brackets and hardware. Even though the toroidal transformer turned out to be twice as expensive as the EI transformer, it was the only acceptable solution for this application.

CONCLUSION

If you're designing or building a piece of quality audio equipment, carefully consider your choice of power transformer. There are trade-offs to be made between the cost and performance of the transformer itself, as well as considerations in the rest of your design (such as where to locate the power transformer) that you need to think about.

Certainly, if cost is not the deciding factor, a toroidal transformer in audio equipment will provide superior performance in almost every respect when compared to a conventional EI-core transformer. But if you are careful with the location of the transformer, you can usually obtain adequate performance with EI transformers in all but the most noise-critical or space-constrained applications.

I hope more manufacturers (especially outside Asia) will invest in the production equipment to manufacture R-core transformers, and the cost of this superior technology will drop enough to make its use compelling. Other transformer technologies continue to be developed as well by leading consumer electronics companies, always striving to build better, smaller, and less expensive audio products. ❖

The Virtual Crossover, Part 2

In Part 2 the author explains the hardware implementation of the Virtual Crossover. **By Richard Mains**

In Part 1 I explained the motivation and theory behind the Virtual Crossover, a device that can either temporarily or permanently replace the real crossover circuit in a speaker system. In a temporary application, you can use it to optimize the component values for the real crossover that will eventually replace it; however, if you use it in place of a real crossover, it allows you greater freedom to do things that are not possible with either passive or active real circuits.

Before I proceed to discuss my implementation, I should mention that I decided to build this as a stand-alone device that can operate independently of a PC, although it can also interface to a PC through the RS-232 serial port. I'm sure it is also possible to implement it using some kind of plug-in card in a PC, but I chose not to do this for several reasons.

First, the plug-in cards that I thought might have the capabilities I required were either very expensive, or it was not clear to me how to write specialized software for them. Second, the idea of trying to carry out a real-time simulation using Windows was a bit daunting. Finally, since I ultimately intended to use this device to play music, I preferred not to start up a PC every time I wished to play my stereo. Nevertheless, I'm sure a PC design would also be a valid approach for this device, and if you should develop such a design, I would be very interested to hear about the results.

IMPLEMENTING THE BASICS

Because it will be important to the discussion of the components in the Virtual Crossover, I repeat the following equation from Part 1:

$$V_{out}(n) = \sum_{i=0}^{N} h_{imp}(i) V_a(n - i). \qquad (9)$$

Equation 9 shows how to calculate the output of the crossover, $V_{out}(n)$, at time $n\Delta t$, given an audio input, V_a and the impulse response of the crossover circuit, h_{imp}. The equation is just a discrete form of the convolution integral, and it is also the expression for the output of a class of digital filters called FIR (finite impulse response) filters.

The most basic requirement for the Virtual Crossover is to perform the above calculation efficiently. Actually, you must calculate equation 9 more than once within each time interval: for each crossover filter and for both left- and right- channel audio signals, resulting, for example, in four separate calculations for a two-way speaker system.

The logical choice to implement this calculation is one of the available DSP (digital signal processing) chips. These chips are specifically optimized to carry out the type of calculation in equation 9, which can be described as a series of "multiply-accumulate" steps. I chose the ADSP21061 chip made by Analog Devices, which can perform a complete multiply-accumulate step in a single processor cycle. In fact, it can do more than that; it can multiply-accumulate and fetch new values from a calculation buffer in a single cycle.

This is also a floating-point chip, which is necessary to calculate equation 9 accurately for a wide range of audio inputs. The clock speed for the device I am using is 40MHz, so that a multiply-

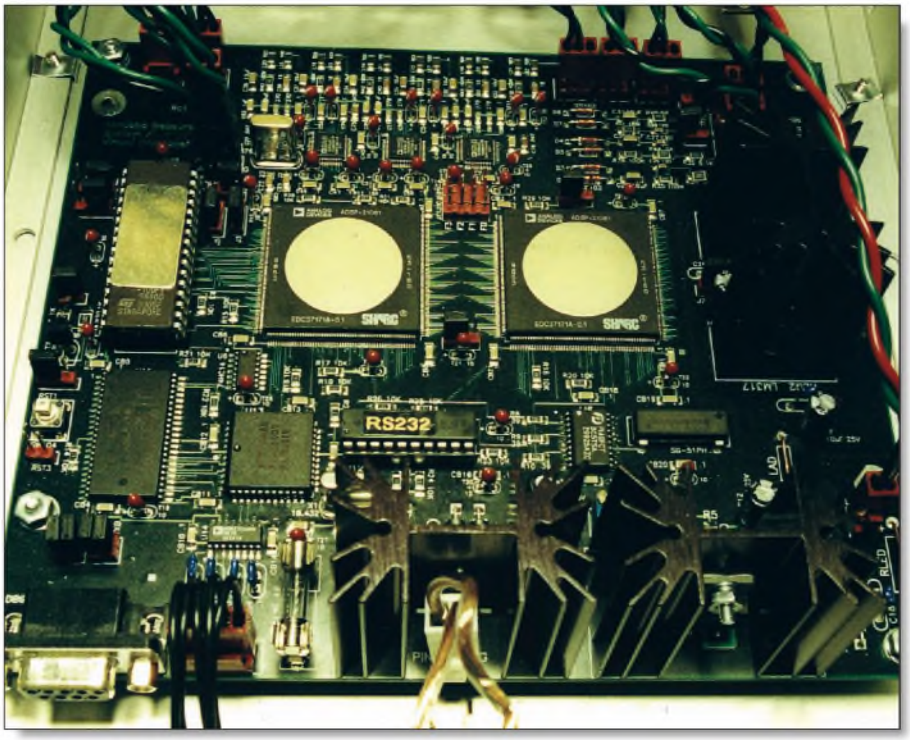

PHOTO 1: Close-up of the first Virtual Crossover DSP board prototype.

accumulate cycle is carried out in 25ns.

The 21061 chip is available in a surface mount package, but it is a difficult one—there are 240 pins, with lead spacing of about a quarter of a millimeter. Hand-soldering a device with such a fine lead spacing is nearly impossible, or at least extremely difficult. I had never designed a surface-mount board before, so the prospect of using such a package was intimidating.

However, in checking with a local PCB fabrication company (Hughes Electronics in Livonia, Mich.), I discovered that the usual procedure is to have such fine-pitch ICs mounted by a machine at the time the board is fabricated. That was encouraging, but the down side is that in order to place the solder paste used for the surface-mount ICs, a stencil is required that adds a couple of hundred dollars to the start-up costs of the board.

A MULTILAYER PCB

I ended up designing a multilayer printed circuit board (*Fig. 8*) for this project, with eight layers (including those on top and bottom) and with surface-mount components on top of the board. To design it, I used an excellent software tool called Tango PCB, made by Accel Technologies. I used version 1.07 for DOS, which I purchased several years ago and which I don't believe is available anymore; however, Accel Technologies offers more recent versions of this product for Windows.

I can't really recommend trying to fabricate this board as a DIY project unless you have a lot of experience with such designs; instead, if you are thinking of building this device, I would recommend obtaining a fabricated board from me, with at least the ICs already mounted. You can solder the discrete components by hand if you wish, although even that requires patience and a steady hand.

In addition to carrying out the equation 9 calculation, the other basic requirement for the Virtual Crossover is the capability of sampling analog audio signals, converting the results to digital form, sending the samples to the DSP for processing, and converting the processed results back into analog

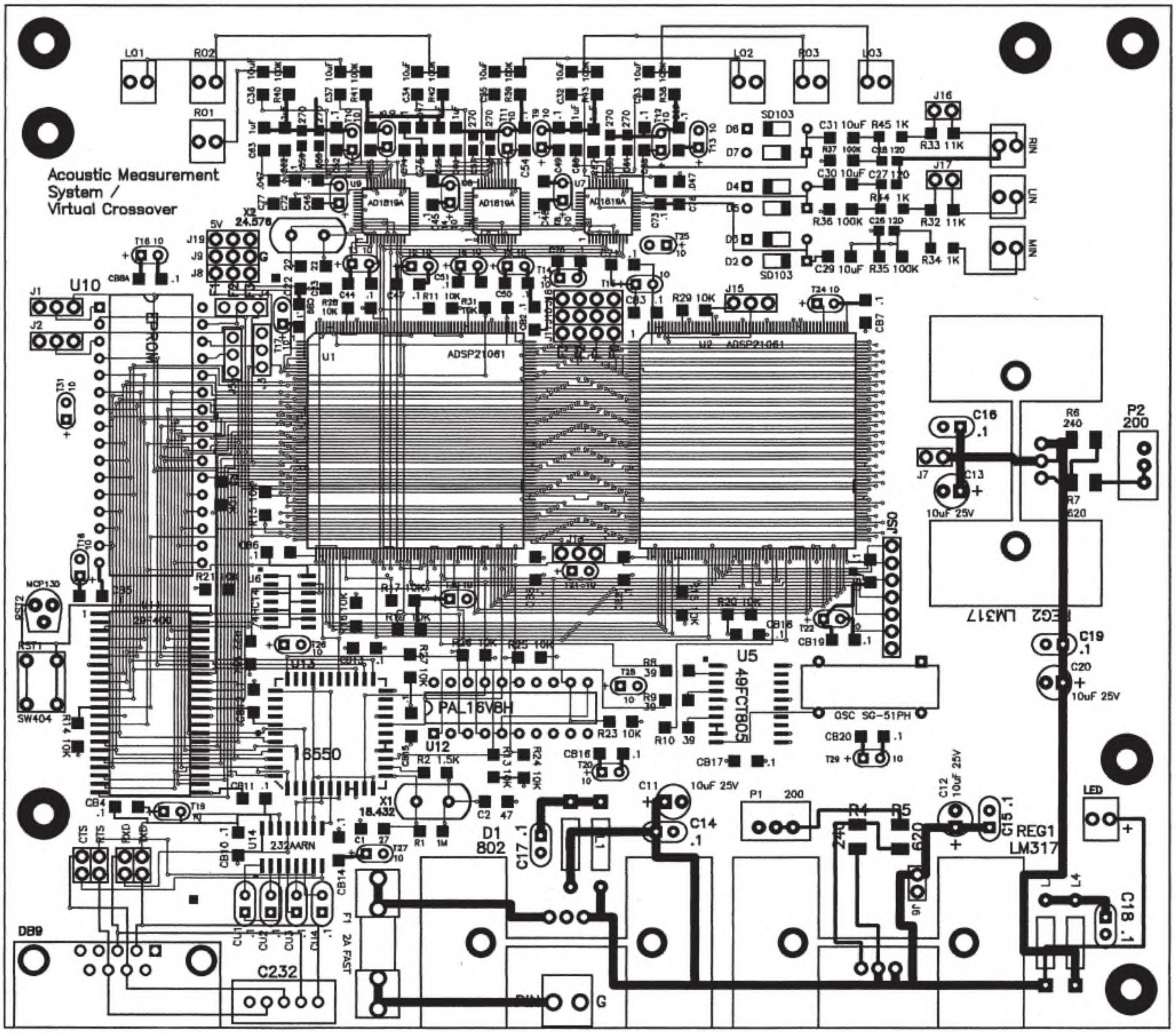

FIGURE 8: Actual size printout of the eight-layer DSP printed circuit board. (100%)

form to drive the speakers. To perform all these functions, I chose Analog Devices' AD1819A SoundPort Codec, which packs quite a bit of audio processing in a small package.

The AD1819A carries out all the functions I've listed, including band-limiting of the audio signal with both analog and digital filters. It is intended to interface with a DSP controller such as the ADSP21061, communicating with it through the synchronous serial port, using a protocol referred to as AC-Link.

This protocol sets up a continuous, bidirectional flow of serial information between the DSP controller and the Codec, with each complete transfer occurring at a standard 48kHz sampling rate. In each 48kHz sampling interval, the Codec sends to the DSP the most recent values it has sampled from the audio inputs, and the DSP simultaneously returns the most recent values it has processed (according to equation 9) that are ready to be applied to the speakers.

The AD1819A uses a 16-bit A/D converter to discretize the audio input signal. It also has digitally controlled input and output gains, which you can select in steps of 1.5dB.

POWER AMPLIFIERS

I should mention at this point that between the output of the Virtual Crossover and the speakers, power amplifiers are required; in fact, you need a separate stereo power amplifier for each driver in the system, so that two power amps are necessary, for example, for a two-way speaker such as the hybrid ESL/TL system.

PHOTO 2: View showing DSP board and Output board mounted on front panel.

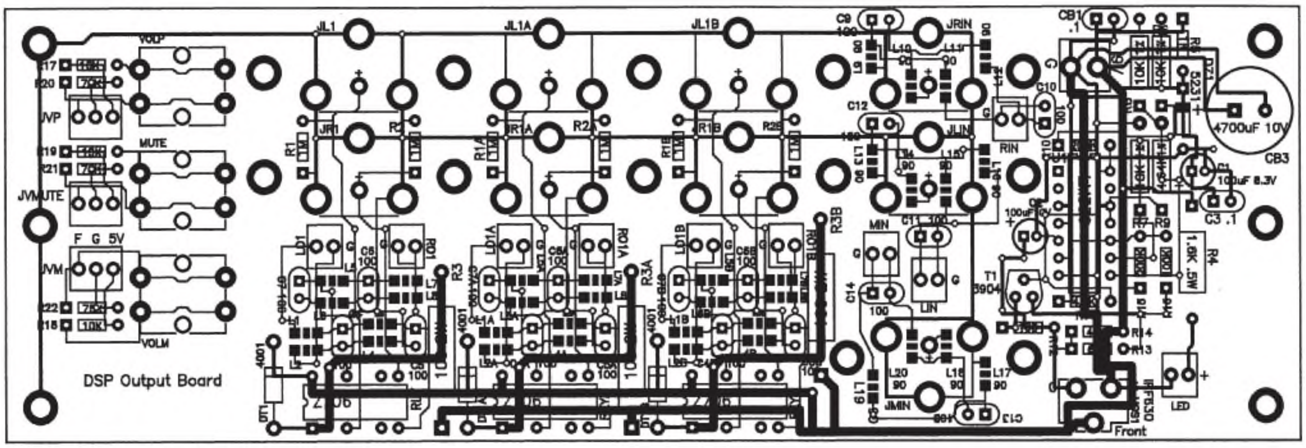

FIGURE 9: Printout of the two-layer Output printed circuit board. (90%)

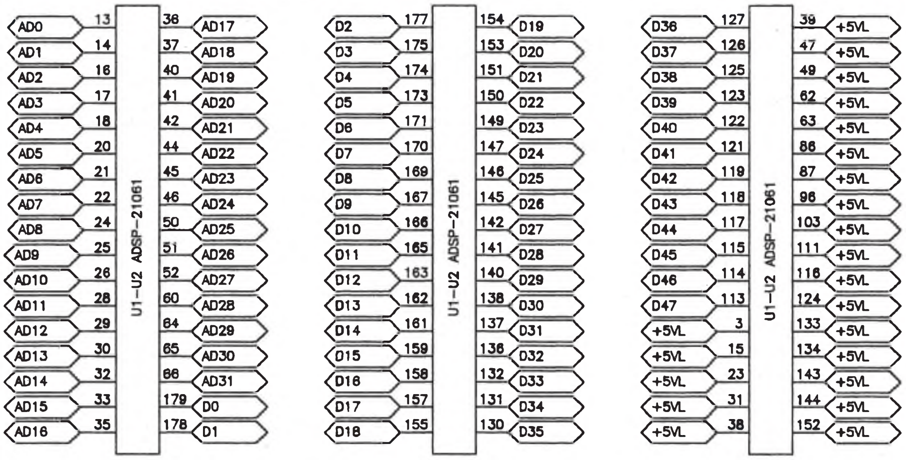

FIGURE 10: Schematic of some common connections between both ADSP21061 processors, U1 and U2.

PHOTO 3: Front panel of Virtual Crossover.

This requirement has motivated me to become involved with power-amplifier design.

One of the reasons I decided to use the AD1819A Codec and ADSP21061 DSP is that a lot of information about these parts is available on the Analog Devices website at www.analog.com, including design examples. As a novice in this area, it was important that I gain sufficient design information if there was to be any hope that my board would actually work. Unfortunately, there is not as much information on the AD1819A Codec as I would like, but there was enough for me to get my very first board working.

In analyzing the impulse responses of some of the crossover circuits I intended to use for my hybrid ESL/TL speakers, the first intended application for this device, I concluded that I would need to process in the range of 500–600 points per channel within each 48kHz sampling interval, or a total of up to 1200 operations of the type in equation 9.

Since the clock speed for the version of the ADSP21061 I used is 40MHz, this would require a processing time of 30µs, which is greater than the 48kHz sampling interval of 20.83µs. Therefore, I decided it would be necessary to use at least two ADSP21061 chips in parallel to carry out the required calculations. Fortunately, these devices are designed to be easily connected together, and examples of multiple-chip systems are shown in the documentation available from the Analog Devices website. I ended up using two of these devices in my first implementation of the Virtual Crossover.

The AD1819A Codec provides a single stereo pair of outputs, but this is not sufficient for the multiple drivers of a speaker system. Each impulse-re-

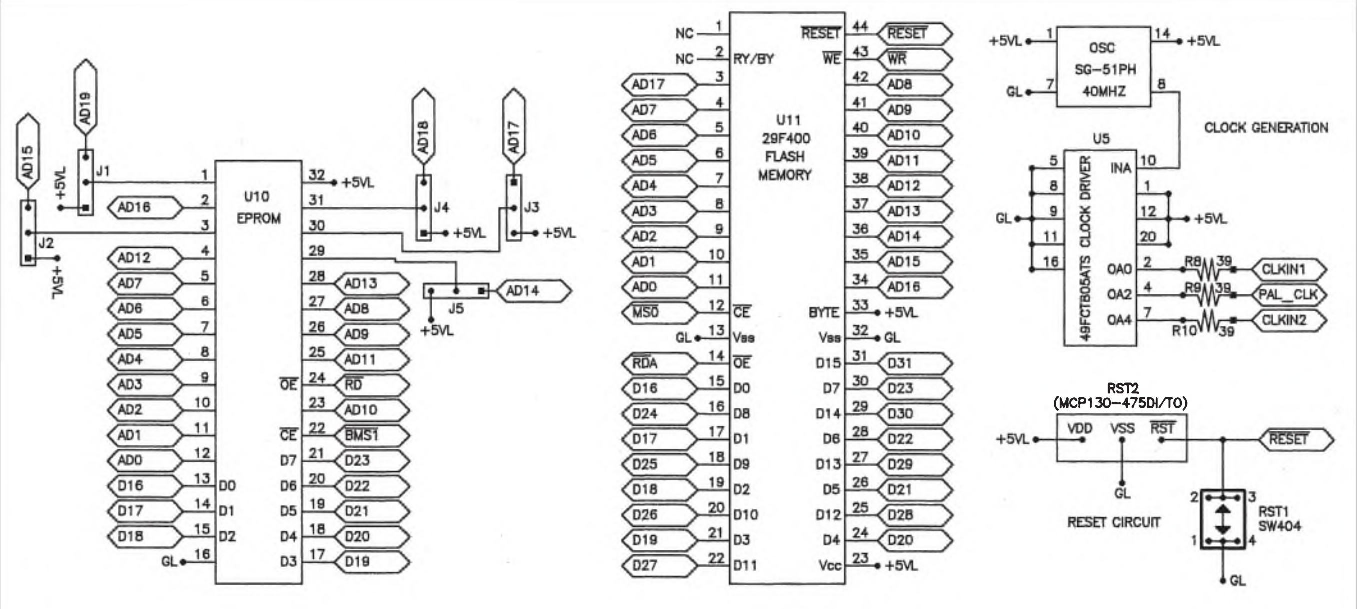

FIGURE 11: EPROM and FLASH memory connections, together with the clock signal-generation and reset circuits.

sponse calculation of the type in equation 9 is for a particular driver; for example, one calculation using the left-channel audio input is for the left-hand transmission line in my speaker system, and a separate calculation using another impulse response is for the left electrostatic driver. Therefore, I chose to cascade three AD1819A chips, which is sufficient to supply a three-way stereo speaker system. Again, it was fortunate that these devices are designed to be easily combined, and multichip designs for the AD1819A Codec are also available on the Analog Devices website.

PERIPHERAL CONSIDERATIONS

In addition to the chips that will carry out the basic functions required by the Virtual Crossover, several peripheral

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features are required. I will describe these systems briefly in this section, then present further details with the schematics.

The most obvious and important support you must add is some sort of memory. The ADSP21061 contains 1MBit of internal SRAM memory, but this disappears when the power goes off; therefore, it is necessary to provide an EPROM chip that boots up the ADSP21061 chips and loads the Virtual Crossover program at startup. I designed my board to permit using several different EPROMs by including selectable jumpers. The EPROM I used initially is the 27C801, which provides 8Mbits, which is more than enough memory for this application.

In addition, considering how the Virtual Crossover will typically be used reveals another type of memory requirement. I envisioned first interfacing the device with a PC through the serial port. During this stage, I would download to the DSP and try out many different crossover circuit impulse responses. I would also carry out acoustic measurements during this stage to determine the effects of changes in the crossover.

However, once I had determined an optimum crossover design, I wished to download and store the impulse-response information semipermanently in the Virtual Crossover memory so that, for example, I could power up the device without a PC and have it use the crossover-circuit information to play music. To implement this type of operation, some kind of erasable and reprogrammable memory is required. I chose to use a

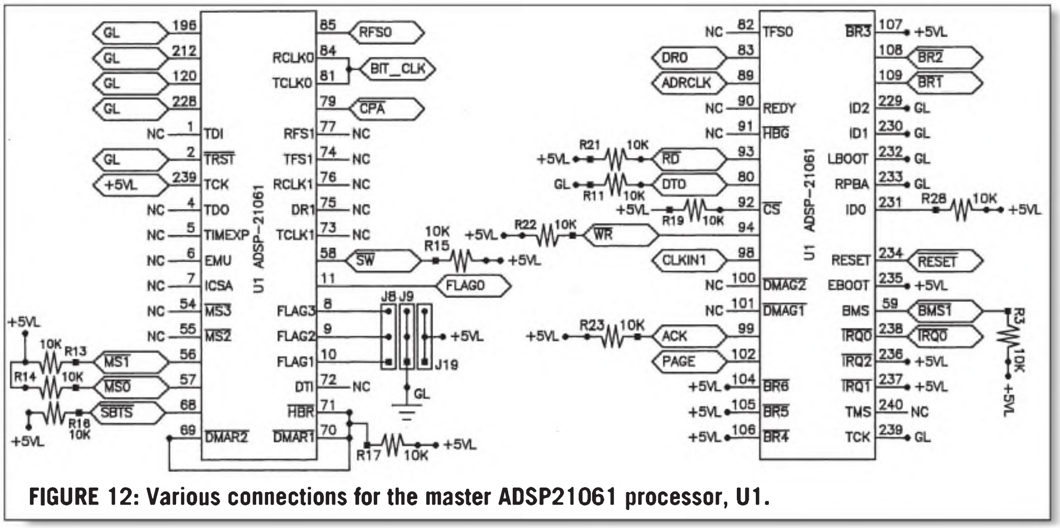

FIGURE 12: Various connections for the master ADSP21061 processor, U1.

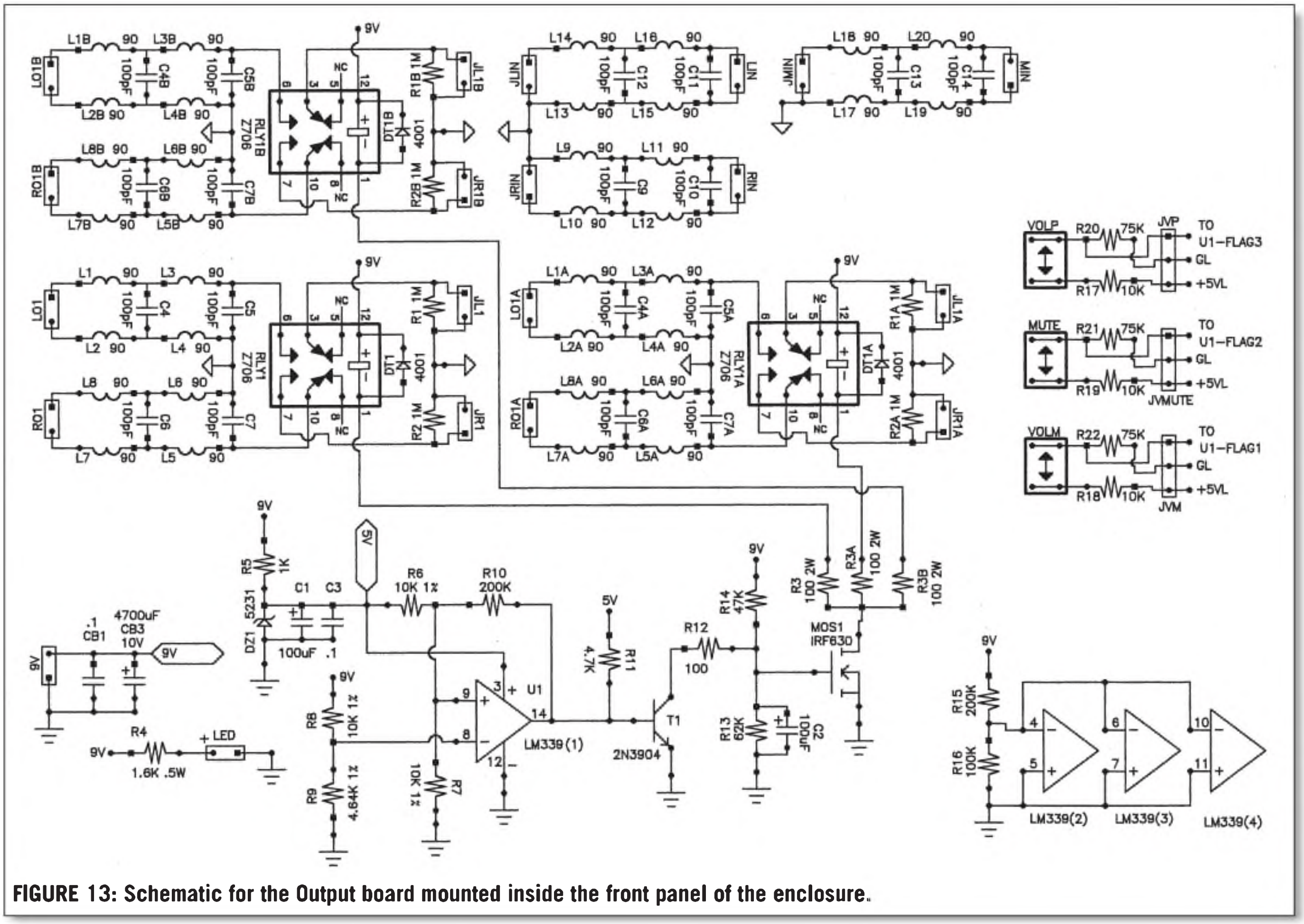

FIGURE 13: Schematic for the Output board mounted inside the front panel of the enclosure.

FLASH memory chip, the AM29F400 by AMD, which provides 4Mbits of programmable memory.

Another important peripheral aspect of this system is the serial interface with a PC, requiring the usual UART and RS-232 level shifter, but additional consideration is also necessary. In order to interface the ADSP21061 with the 16550 UART chip that I used, it is necessary to insert a buffer chip between the two; simply adding wait states to the ADSP21061 memory read/write cycles is not sufficient, because control signals in proper sequence must also be supplied to the UART.

I first became aware of this problem from the *ADSP-2106x SHARC EZ-KIT Lite Reference Manual* by Analog Devices, where I saw a solution using a GAL16V8 programmable chip as the interface between the ADSP21061 and UART. I developed a similar program for the GAL16V8 and used it in my board design. For the design details, including the .JED file that you can use to program the GAL16V8, please visit my website.

THE OUTPUT BOARD

I first made a single PCB for this project, which I referred to as the DSP board. However, when I tested it I discovered that the audio outputs of the AD1819A Codecs go through a rather nasty transient when they are powered up or reset. If the Codec outputs are connected directly to power-amplifier inputs, as I envisioned they would normally be, the resulting transient from the power amps might destroy speakers unless the power amps included a timed relay-protection circuit at power-up.

Even if the power amps were protected with relays, a sudden dip in power to the crossover might cause the processors and Codecs to reset, initiating a transient while the power amps were connected. It became evident, therefore, that I needed to include adequate relay protection in the Virtual Crossover itself to avoid such mishaps.

In addition, when it came time to think about placing the crossover in an enclosure and interfacing the three inputs and six outputs to the outside world, it became clear that I needed to

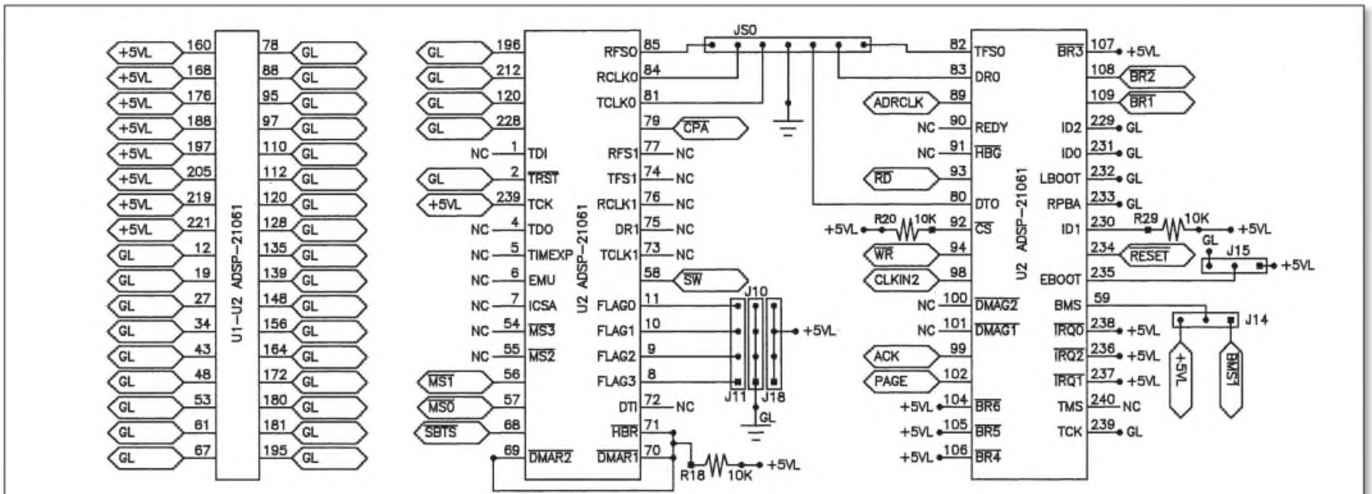

FIGURE 14: Additional common power connections for U1 and U2, together with various connections specific to slave DSP U2.

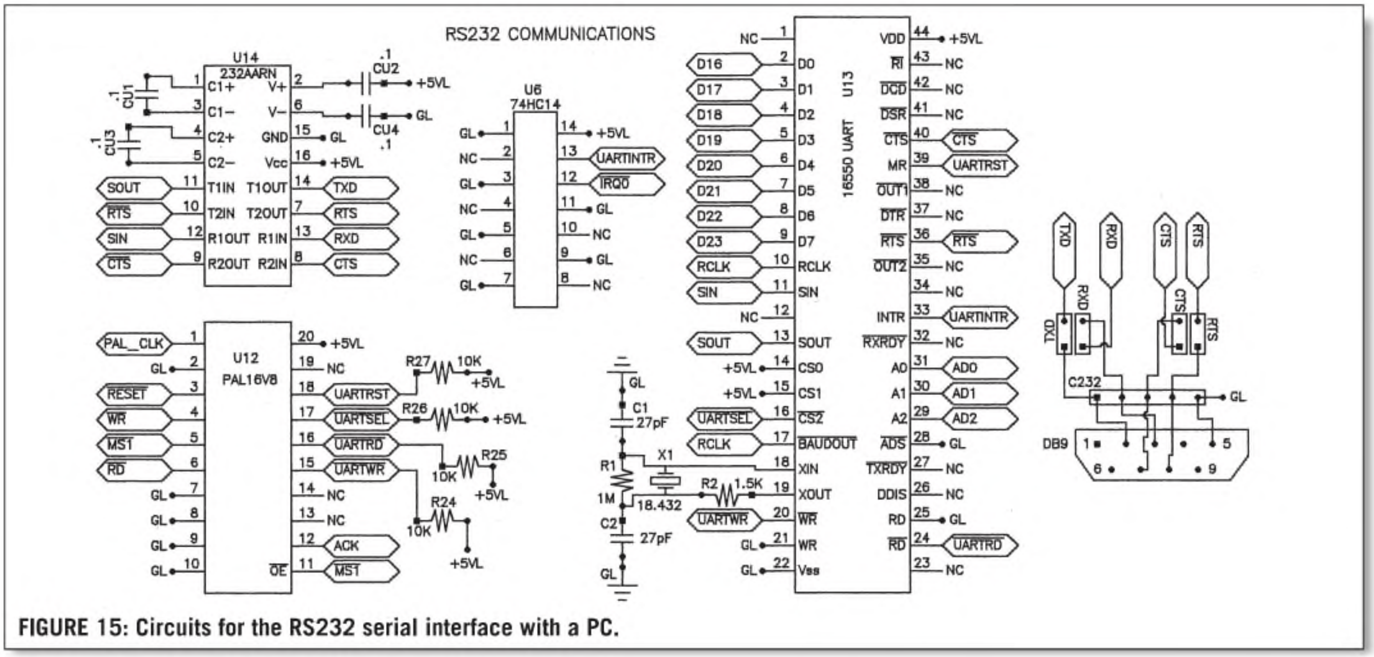

FIGURE 15: Circuits for the RS232 serial interface with a PC.

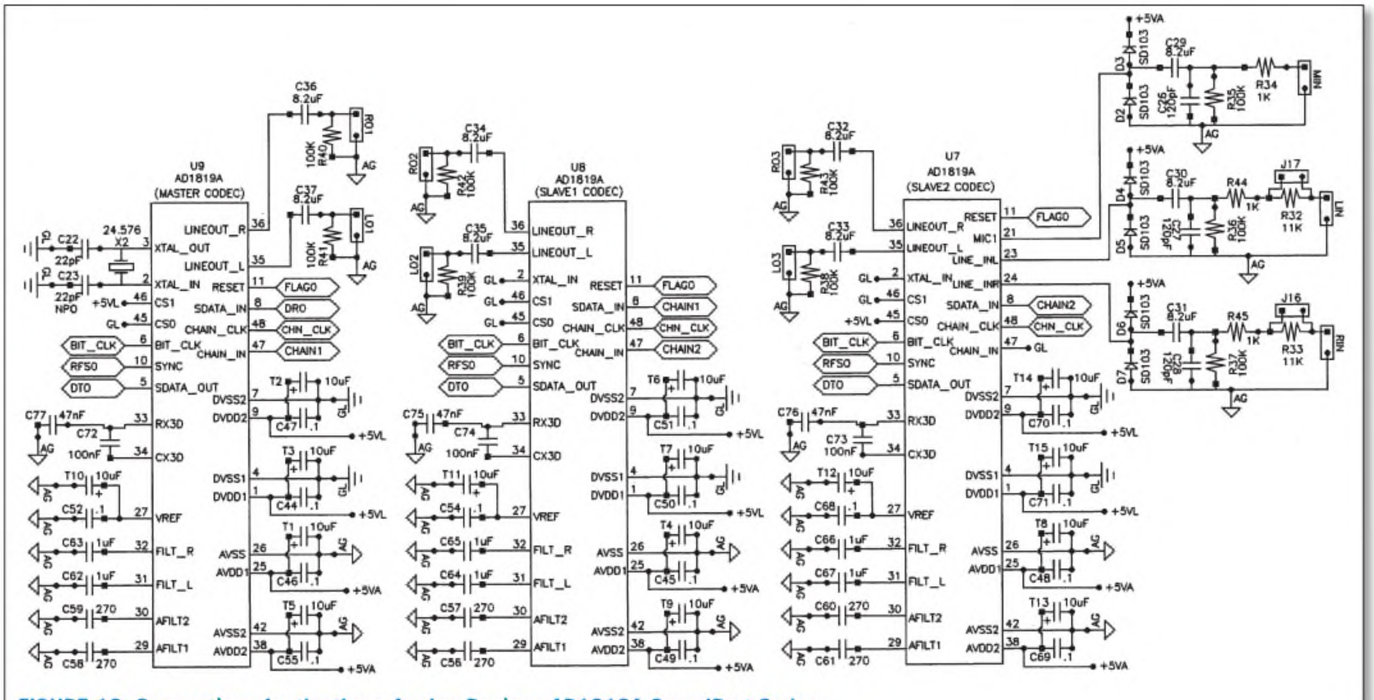

FIGURE 16: Connections for the three Analog Devices AD1819A SoundPort Codecs.

design a second board, which I called the Output board, that would include phono jacks for the front panel as well as relay protection.

Photo 1 shows a close-up of the first DSP board that was completed. The Output board is mounted on the inside of the front panel of the enclosure; *Photo 2* is a view showing both the DSP and Output boards. I used PC-mount phono jacks from Switchcraft (no. 3514PC) so that I could mount them directly on the Output board. *Photo 3* shows the front panel; the phono jacks and LED are directly soldered in the OUTPUT board. I haven't attempted to label the front panel yet, but I should because it is difficult to remember where all the phono jacks are connected.

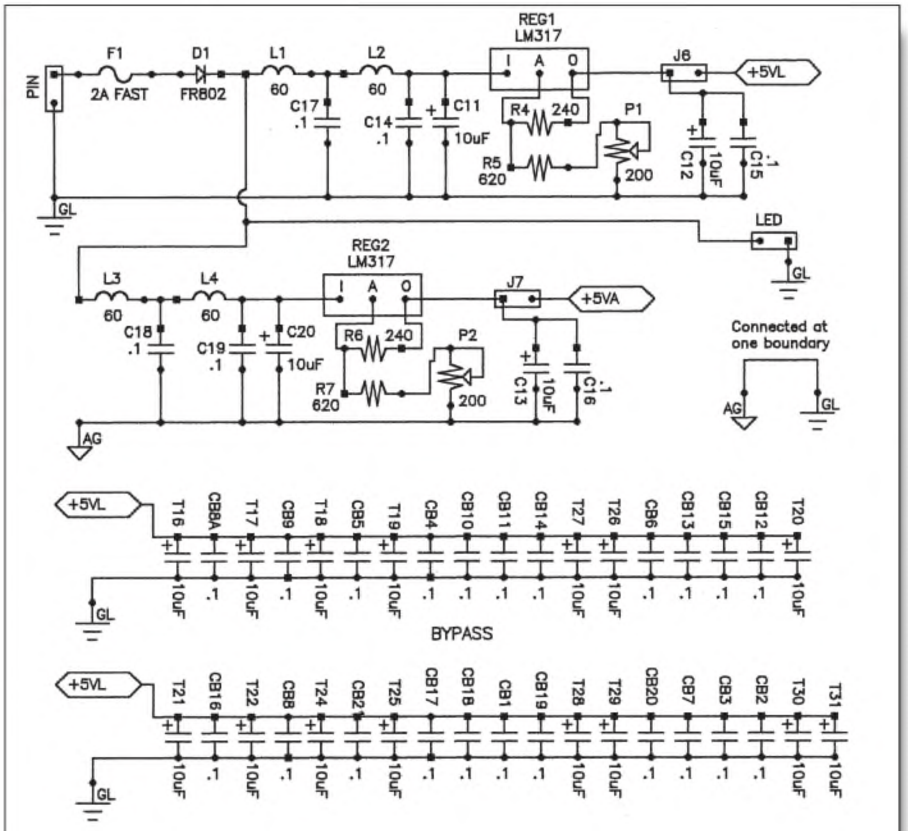

FIGURE 17: Digital and analog power-supply circuits for the DSP board.

I should mention that the schematics do not correspond exactly to the boards shown in these photos. Based on my experience with this first prototype, I have decided on some changes I will make with future boards. I have included these changes in the schematics, since they will be improvements in the Virtual Crossover.

Figure 8 shows an actual-size printout of the DSP board, with the changes that I mentioned already implemented. Keep in mind that this printout shows all the metallization layers, and since the figure is a single color, it is not possible to tell which lines are on which layers. *Figure 9* shows a printout of the Output board, which is a two-layer board with no internal layers.

SCHEMATIC DETAILS

In this section I present detailed schematics of both the DSP and Output boards. With each one, I will discuss important aspects of the design that pertain thereto.

Figure 10 shows some common con-

TABLE 1
PARTS LIST FOR DSP BOARD

(ALL PARTS FROM DIGI-KEY UNLESS OTHERWISE NOTED)

QUANTITY	DESCRIPTION	PART NO./SOURCE	COMPONENT NUMBERS
1	DB9 connector	104951 Jameco	DB9
1	DB9 connector	15771 Jameco	(chassis mount)
4	ferrite core	P9820BK-ND	L1–L4
10	2-pin header	WM2700-ND	LED, LIN, LO1–LO3, MIN, RIN, RO1–RO3
10	2-pin housing	WM2011-ND	(not on board)
1	5-pin header	WM2703-ND	C232
1	5-pin housing	WM2003-ND	(not on board)
25	crimp terminals	WM2312-ND	(not on board)
13	2-pin jumper	WM2722-ND	CTS, J1–J7, J16, J17, RTS, RXD, TXD
15	shunts	S9002-ND	(not on board)
5	3-pin jumper	WM2723-ND	J8, J9, J14, J15, J19
3	4-pin jumper	WM2724-ND	J10, J11, J18
3	3-pin housings	WM2801-ND	(not on board)
13	crimp terminals	WM2512-ND	(not on board)
1	7-pin header	WM2727-ND	JS0
1	7-pin housing	WM2805-ND	(not on board)
3	.047µF caps smt	PCC473BCT-ND	C75–C77
10	.1µF chip caps	P4910-ND	CU1–CU4, C14–C19
40	.1µF caps smt	PCC1870CT-ND	C44–C52, C54, C55, C68–C74, CB1–CB21, CB8A
1	1.5k res smt	P1.5KECT-ND	R2
30	10µF caps tant	P2013-ND	T1–T22, T24–T31
9	100k res smt	P100KECT-ND	R35–R43
20	10k res smt	P10KECT-ND	R3, R11, R13–R29, R31
9	10µF caps smt	PCC1939CT-ND	C29–C37
4	10µF 25V elect	P6746-ND	C11–C13, C20
3	120pF caps smt	PCC121CGCT-ND	C26–C28
1	16550 uart	PC16550DV-ND	U13
1	18.432 crystal	CTX061-ND	X1
3	1k res smt	P1.0KECT-ND	R34, R44, R45
1	1M res smt	P1.0MECT-ND	R1
6	1µF caps smt	PCC1893CT-ND	C62–C67
2	200 pot	3296W-201-ND	P1, P2
2	22pF caps smt	PCC220CCT-ND	C22, C23
1	232AARN	ADM232AARN Future	U14
1	24.576 crystal	CTX092-ND	X2
2	240 res smt	P240VCT-ND	R4, R6
1	27pF cap smt	PCC270CCT-ND	C1
6	270pF caps smt	PCC271CGCT-ND	C56–C61
1	29F400 flash mem	AM29F400BB-90SC-ND	U11
1	2A fuse (fast)	283-2335-ND	F1
3	39 res smt	P39ECT-ND	R8–R10
1	47pF cap smt	PCC470CCT-ND	C2
1	49FCT805 clk buf	PI49FCT805ATS	U5
2	620 res smt	P620VCT-ND	R5, R7
1	74HC14	MM74HC14M-ND	U6
1	802 diode	FR802	D1
2	11k res smt	P11KECT-ND	R32, R33
1	2-pin .156 header	WM4620-ND	PIN
1	2-pin .156 housing	WM2122-ND	(not on board)
2	.156 crimp terms	WM2300-ND	(not on board)
3	AD1819A Codecs	AD1819AJST Future	U7–U9
2	ADSP21061 DSPs	ADSP-21061KS-160 Future	U1, U2
1	EPROM	M27C801-100F1 Future	U10
2	LM317 volt reg	LM317AT-ND	REG1, REG2
1	MCP130 pwr mon	MCP130-475DI/TO	RST2
1	PAL16V8H	PALCE16V8H-10 Jameco	U12
6	SD103 Schottky	SD103ACT-ND	D2–D7
1	SG-51PH osc	SE1907-ND	OSC
1	SW404 pshb swtch	SW404-ND	RST1
3	heatsinks	HS188-ND	part of REG1, REG2, and D1
1	32 pin IC socket	A95326-ND	U10
1	20 pin IC socket	A9420-ND	U12
3	#4 ⅜″ screws	H781-ND	to attach devices to HS
	heatsink grease	CT40-5-ND	to attach devices HS
7	#4 flat washers	H777-ND	
7	#4 lock washers	H236-ND	
7	#4 nuts	H216-ND	
1	chassis	B2H10-001A Lansing	(top should be perforated)
4	½″ tubular spacers to mount DSP board on chassis		
4	#4 ¾″ screws	H350-ND	
1	power connector	151589 Jameco	
1	9V DC 2.3A or 1.67A wall transformer	167564 or 161550 Jameco	
1	On-off power switch	GC 35-691 (from local retailer)	

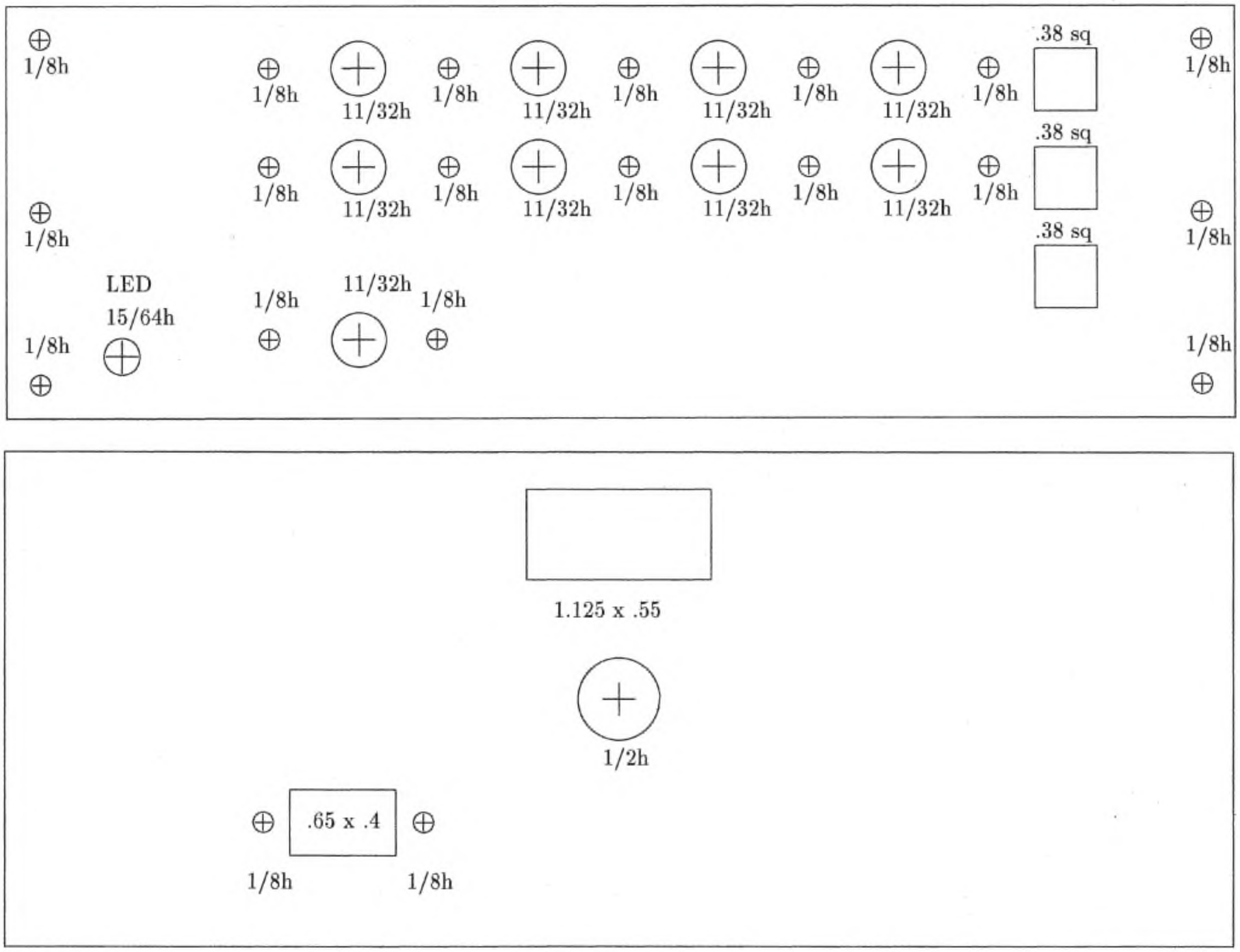

FIGURE 18: Templates used to machine the front and back panels of the B2H10-001A enclosure, available from Lansing Instrument Corp. (90%)

nections between the two ADSP21061 processors; the notation "U1-U2" indicates that identical connections are made for both processors, U1 and U2. The labels in boxes indicate common connections that may appear elsewhere on other schematics. For example, "AD0" is the lowest-order bit on the address bus; *Fig. 10* indicates that this line is connected both to U1 pin 13 and U2 pin 13. However, this same line will appear elsewhere in the schematics; for example, in *Fig. 11* it is also attached to the EPROM and FLASH memory chips.

The common-address and data bus connections in *Fig. 10* are necessary to connect the two processors in parallel. Analog Devices provides information and examples of multiprocessor systems on its website. (These are not the only lines the processors must share; others will appear on later schematics.) In addition to address and data bus connections, *Fig. 10* also shows various required power connections for the processors (+5VL, where "L" refers to Logic, as opposed to Analog, power).

Figure 12 shows some connections for the master DSP processor, U1 ("GL" in these diagrams refers to Logic Ground, and "NC" denotes No Connection). MS0 and MS1 lines are used to select external memory, and they must be connected together between U1 and U2; MS0 selects FLASH memory, and MS1 selects the UART. BMS1 is used to select "Boot Memory," which in this case is the EPROM chip U10. Note that many of these lines need to be pulled up.

Several lines in *Fig. 12* are involved with the serial interface between the master DSP processor and the Codecs. This interface follows a protocol referred to as "AC-Link," which provides a method of continuously transferring data and commands at a standard 48kHz rate. BIT_CLK is a 12.288MHz clock that is generated by the master AD1819A Codec, U9, as the timing for this interface. Each information-exchange at the 48kHz rate is initiated by a synchronization signal, RFS0, which is generated by the master DSP, U1. During each information exchange interval, data is transmitted from the master DSP to the Codecs via the DT0 line, and data is simultaneously sent from the Codecs to the DSP processor on the DR0 line.

In the application information for interfacing the AD1819A Codec to a DSP controller, Analog Devices recommends pulling down the DT0 line with a resistor in the 10-20kΩ range. When I first assembled the DSP board, I used a 15kΩ resistor, which did not work—the AD1819A Codecs did not function properly (Analog Devices explains that they can enter "factory test mode" unless DT0 is pulled down). In order to get the Codecs to work, I needed to use a 10kΩ resistor, which is shown as R11 in *Fig. 12*.

FOUR FLAGS

Each ADSP21061 processor has four flags that you can configure either as inputs or outputs. I used FLAG0 of DSP processor U1 as an output to reset all three Codecs. The remaining flags, FLAG1–3, I configured as inputs and brought out to header J8. These flags I used as inputs for the volume up/down and mute controls from the Output board, which are push-button switches. You can see these controls in *Fig. 13*, the Output-board schematic, and in *Photo 3* on the front panel. The adjacent power and ground connections on headers J19 and J9 were also required for these Output-board controls.

RD, WR, and ACK are important signals for interfacing the DSPs to each other and to peripheral devices, and you can find these signals in several locations in the schematics. IRQ0 is the interrupt line to the master DSP processor from the UART (after going through inverting buffer U6), which indicates that the UART has received a byte of information from a PC over the RS-232 serial port. (It is not necessary to have a PC on the serial port for the Virtual Crossover to function, however, if there is no PC, all it can do is play music, using the information that has been stored previously in the FLASH memory.)

BMS1 is the boot-memory select line, which selects the boot-up source when the DSPs are powered up or reset. You can see in *Fig. 11* that this line selects the EPROM chip, U10. ID0–2 are used to identify master and slave DSPs in a multiple-processor environment; they are set to select U1 as the master, and U2 as the slave.

Figure 14 shows some power connections common to processors U1 and U2, together with some connections specific to slave DSP processor U2. For U2, all four flags are brought out to header J11, even though these flags are not currently used. The flags are very valuable tools in the debugging stage of DSP program development, so they should all be made available even if they will not be used during normal operation of the device.

The lines associated with serial port 0 of U2 are brought to header JS0 in order to accept direct digital input from a CD player, rather than having to sample the input through the A/D converter. I have not yet attempted to do this; it remains as a future project.

The BMS and EBOOT lines control the boot-up sequence when the DSP is powered up or reset. Since I was not sure of the best method to boot up the devices, I brought these lines from U2 out to jumpers J14 and J15 so I could change the settings if there were problems. I ended up setting EBOOT high for both processors and connecting their BMS pins, which means that both devices boot up from the same EPROM at startup. The boot loader provided with the Analog Devices evaluation kit allows each processor to boot a different program from the same EPROM, which is necessary since the master functions somewhat differently than the slave DSP.

EPROM CONNECTIONS

Figure 11 shows the connections for the U10 EPROM, which stores the programs that the U1 and U2 DSPs load at startup, along with the connections for the FLASH memory, which stores the impulse-response information and other data that allows the Virtual Crossover to function without a PC. Of course, you must first program the EPROM; I used the Chipmaster 2000 programmer available from Logical Devices, the same programmer that I used to program the PAL16V8 chip for the RS-232 communications interface.

If you are interested in seeing the programs I used for the DSPs, I will make the source code available on my website; these programs, written in assembly code, are called DSPMSTR.ASM for the master ADSP21061 (U1), and DSPSLAVE.ASM for the slave ADSP21061 (U2).

I provided jumpers for the U10 EPROM so that you could use different EPROMS. As I indicated previously, the 27C801 device I am currently using has much more memory than is really needed.

Also shown in *Fig. 11* is the clock-generation circuit for the DSPs and the PAL16V8. The clock signal is one of the most important on the board, and Analog Devices recommends using a clock-driver chip together with series resistors to attenuate reflections on the clock lines. It also recommends laying out the clock lines to reach the devices at approximately equal distances from the clock driver, so that all the devices in the system are synchronized.

Finally, *Fig. 11* also shows the RESET-signal generation circuit. I used a "microcontroller supervisor" chip, the MCP130-475DI/TO, which generates a reset pulse whenever the voltage drops below a certain value; this chip is guaranteed to reset somewhere in the range 4.5V to 4.75V.

Also, since this device has an open-drain output with pullup resistor, you can connect a reset switch (RST1 in the figure) to manually reset the board. (One word of caution: if you use this board, do not manually reset it while it is connected to live power amps connected to speakers; if you do, the relays that I mentioned earlier will afford absolutely no protection from the reset transient.)

SERIAL INTERFACE WITH A PC

Figure 15 shows the connections that implement the RS-232 serial interface with a PC. As explained previously, the PAL16V8 chip acts as a buffer between the DSP and UART, allowing signals to be presented in the required sequence and timing. U12 must be programmed, and the program I used (RS232.JED) is available on my website. U6 is an inverter, required only because the 16550 UART generates a positive polarity interrupt signal, whereas the U1 DSP requires a low IRQ0 interrupt signal.

U14 is an RS-232 level shifter that translates the TTL-level signals from the UART to the approximately ±10V required on the serial line. I provided jumpers at the RS-232 interface output so that, for example, you could also use a null-modem cable. There are two RS-232 output connectors mounted on the board, a header C232 and a 9-pin DB9 connector; I included the DB9 so that you could test the board with a PC while it is removed from the enclosure. On the back panel of the enclosure, there is also a DB9 connector that would normally be used with a PC; wires from this panel-mount connector plug into header C232.

Figure 16 shows the schematic for the three AD1819A Codecs on the DSP board. These Codecs are connected together in a master-slave type of configuration, with U9 being the master, U8 the slave1 and U7 slave2. Analog Devices provides information on its website about how to interface multiple Codecs, both among themselves and to a controlling DSP. I needed three Codecs because each one has only a single stereo output, whereas I require three stereo outputs for a three-way stereo speaker system.

There are three inputs to the Codecs; LIN and RIN would typically be used for a stereo music input signal, for example, from a CD player. Note that I included with these inputs two 11kΩ series resistors, R32 and R33, which you can optionally short out with jumpers J17 and J16. The input impedance of the AD1819A is specified as 10k in the data sheet, and the maximum input voltage that this device can accept before clipping is specified as 1V RMS, or 1.414V peak.

Since the analog output of a CD player can typically be up to 3V peak, it could not be connected directly to the AD1819A input. However, with R32 and R33 in the circuit, together with the 1kΩ and 100kΩ resistors already there, 3V peak input is lowered to 1.29V at the input to the AD1819A. I included jumpers in case you use a lower-level input source, and this extra attenuation is not needed.

The remaining input, MIN, might be used for a microphone or any other input that requires amplification. MIN provides an optional 20dB gain that you can select or de-select through software. I intended that the MIN input would probably only be used for acoustic measurements.

CODEC FUNCTIONING

The overall function of the Codecs is as follows. Each one is responsible for the output of the left- and right-channel signals associated with one of three crossover circuits. For example, the master Codec, U9, outputs the left and right channels for the first crossover filter at outputs LO1 and RO1. This filter might be the low-pass circuit for the woofers, for example. Slave1 (Codec U8) handles the outputs for the second crossover filter, and slave2 (Codec U7) handles the third filter. In addition, the slave2 Codec is responsible for sampling the audio input signals through its A/D converters and sending the resulting samples to the DSP U1 for processing.

The details of this information exchange may be found in the AD1819A data sheet and other literature available on the Analog Devices website. The Virtual Crossover can also operate as a measurement system, in which case test signals (typically filtered pulses) are output by the Codecs, and the response (from a microphone, for example, after a specified delay) is relayed back to the DSP for transmission to a PC over the RS-232 port.

Figure 17 shows the circuits for both the digital and analog +5V power supplies. Analog Devices recommends using separate power supplies if possible, with the grounds connected at one point or in one localized region to avoid ground loops. I used LM317 adjustable-voltage regulators for both supplies. Especially for the digital supply, it is important to provide a hefty heatsink and

TABLE 2
PARTS LIST FOR OUTPUT BOARD

(ALL PARTS FROM DIGI-KEY UNLESS OTHERWISE NOTED)

QUANTITY	DESCRIPTION	PART NO./SOURCE	COMPONENT NUMBERS
9	RCA jacks PC mnt	SC 3514PC Switchcraft	JL1, JL1A, JL1B, JLIN, JMIN, JR1, JR1A, JR1B, JRIN
3	push button swtch	SW413-ND	MUTE, VOLM, VOLP
3	switch cover	SW452-ND	(not on board)
9	2-pin header	WM2700-ND	LIN, LO1, LO1A, LO1B, MIN, RIN, RO1, RO1A, RO1B
9	2-pin housing	WM2011-ND	(not on board)
27	crimp terminals	WM2312-ND	(not on board)
3	3-pin header	WM2701-ND	JVM, JVMUTE, JVP
3	3-pin housing	WM2012-ND	(not on board)
1	2-pin .156 hdr	WM4600-ND	9V
1	2-pin .156 hsg	WM2123-ND	(not on board)
2	crimp terminals	WM2300-ND	(not on board)
2	.1µF caps	BC1084CT-ND	C3, CB1
1	1.6k .5W res	1.6KH-ND	R4
1	100 res	100EBK-ND	R12
18	100pF caps	BC1013CT-ND	C4–C7, C9–C14, C4A–C7A, C4B–C7B
3	100 2W res	P100W-2BK-ND	R3, R3A, R3B
1	100k res ⅛W	100KEBK-ND	R16
1	100µF 6.3V cap	P10195-ND	C1
1	100µF 10V cap	P10219-ND	C2
1	4700µF 10V cap	P5524-ND	CB3
3	10k res ⅛W	10KEBK-ND	R17–R19
3	10k res 1%	10.0KXBK-ND	R6–R8
1	1k res ⅛W	1.0KEBK-ND	R5
6	1M res ⅛W	1.0MEBK-ND	R1, R1A, R1B, R2, R2A, R2B
2	200k res ⅛W	200KEBK-ND	R10, R15
1	3904 transistor	2N3904	T1
1	4.64k res 1%	4.64KXBK-ND	R9
1	4.7k res ⅛W	4.7KEBK-ND	R11
3	4001 diode	1N4001MSCT-ND	DT1, DT1A, DT1B
1	47k res ⅛W	47KEBK-ND	R14
1	5231 zener	1N5231BDICT-ND	DZ1
1	62k res ⅛W	62KEBK-ND	R13
3	75k res ⅛W	75KEBK-ND	R20–R22
36	90 ferrite smt	240-1037-1-ND	L1–L20, L1A-L8A, L1B–L8B
1	IRF630 MOSFET	IRF630	MOS1
1	transistor socket	WM2550-ND	MOS1
1	LM339 comp	LM339AN-ND	U1
3	Z706 relays	G5A-237P-DC5	RLY, RLYA, RLYB
4	14-pin IC sckt	AE8914-ND	U1, RLY, RLYA, RLYB
1	LED	94511 Jameco	LED
18	plastic spacers	3103K-ND	
18	#4 ⅝ screws	H164-ND	
18	#4 nuts	H216-ND	
18	#4 flat washers	H777-ND	
18	#4 lock washers	H236-ND	

ventilation through the top of the enclosure. I included jumpers at the power-supply outputs, J6 and J7, so that I could adjust the supplies before applying power to the rest of the board.

The power input, PIN, is from a 9V DC wall transformer obtained from Jameco. Protection is provided through fuse F1 (this board draws approximately 1.1A when operating normally), and diode D1 protects against accidental application of reverse-polarity power. Ferrites L1-L4 and capacitors C14, C17, C18, and C19 form low-pass filters to isolate digital noise generated on the board from the input power.

RELAY PROTECTION

Figure 13, the Output board schematic, was primarily designed to provide relay protection against the reset transient from the Codecs. It also provides filtering for the input and output signals, isolating them from digital noise generated on the DSP board. Finally, the volume and mute controls are also implemented on this board.

The relay protection works as follows. Comparator LM339(1) controls whether or not the relays are energized. A 2.5V signal is present at the positive input of the comparator, which is compared to a divided-down 9V power signal from the wall transformer at the negative input. At power-up, regardless of the state of the output of this comparator, it takes a few seconds for capacitor C2 to charge up to the point that MOSFET MOS1 turns on to switch the relays so the Codec outputs are connected.

Under normal operation, the divided-down 9V signal at the negative input would be about 2.85V, so the output of the comparator would be grounded, T1 would be off, and MOS1 would be on. If the 9V power input drops below 7.88V, the output of the comparator goes high. As a result, T1 switches on, quickly draining the charge from C2 through 100Ω resistor R12, which turns off MOS1 and disconnects the Codec outputs. Zener diode DZ1 provides 5V power for the comparator; this 5V is held steady during power transients by capacitor C1.

The series Ls and shunt Cs in the input and output signal paths in *Fig. 13* filter out digital noise generated on the DSP board. PC-mount phono jacks are soldered directly to the Output board and protrude through the front panel. A power-indication LED is also soldered directly to the board and protrudes through the front panel.

ENCLOSURE

I used an enclosure available from Lansing Instrument Corp. in Ithaca, N.Y.; the part number for the standard enclosure is B2H10-001A. However, it is desirable to have the top panel perforated at least in the areas above the DSP chips and the +5VL regulator heatsink. Lansing will punch perforations in the top panel of this enclosure for a very reasonable charge, although there is a $60 additional setup fee for each custom part order.

The standard enclosure itself is $69.71, and the fee for punching the top panel (according to instructions that I supplied to Lansing) is $12.53. Lansing assigned custom-part numbers to this modified enclosure: B2H10-7895 for the complete enclosure with modified top panel, and 38946-7896 for the perforated top panel itself.

I attempted to machine the front and back panels of this enclosure myself; *Fig. 18* shows the templates that I used. As you can tell from *Photo 3*, my machining capability is rather primitive. I ended up needing to do a lot of filing to get everything to fit in the openings.

COST AND AVAILABILITY

As you have probably already surmised, this is not an inexpensive project. The first prototype DSP board cost $658, including a one-time fee of $270 for the stencil used to mount some of the more difficult surface-mount components. This cost is for the board itself; it does not include any of the components mounted on the board, although there is a labor charge included to mount some of the surface-mount components when the board is fabricated.

The most expensive component on the board is the ADSP21061 DSP processor, which is about $75 in small quantities, and there are two of them on the board. The initial Output-boards I had made were about $93 each. If you are interested in this system, I may make these boards available at a price considerably lower than that of the initial prototypes, the reduction depending on how many people are interested. If you think you might want to use these boards, please contact me at my e-mail address so that I can give you an idea of what the quantities and prices will be.

In Part 3 I will show how you can use the Virtual Crossover to design the crossover for a hybrid electrostatic/transmission line speaker system similar to that described in the excellent series of articles by Roger Sanders. I hope to demonstrate some of the unique capabilities of this device that cannot be accomplished with either conventional active or passive real crossovers.

ACKNOWLEDGMENTS

I would like to thank R.J. Smith and Tom Bewick of Hughes Electronics in Livonia, Mich., whose expert and very patient guidance prevented my first surface-mount PCB project from becoming a disaster.

❖

REFERENCES

1. Sanders, Roger R., "An Electrostatic Speaker System—Parts I, II, and III," *SB* 2/80, 3/80, and 4/80.

RESOURCES

I have posted the software used in the development of the Virtual Crossover on my website at www.usol.com/~rkm/audio. Questions regarding the software or these articles can be directed to my e-mail address at rkm@usol.com.

PARTS SOURCES

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

Jameco Electronic Components
1355 Shoreway Rd.
Belmont, CA 94002-4100
1-800-831-4242
www.jameco.com

Lansing Instrument Corp.
705 Willow Ave.
Ithaca, NY 14850
1-800-847-3535
www.plantfloor.com/ny/lansinginstrumentcorp.htm

Future Active
1-800-655-0006
www.future-active.com

Accel Technologies, Inc.
12348 High Bluff Dr.
San Diego, CA 92130
www.acceltech.com

Logical Devices, Inc.
2062 Stout St.
Denver, CO 80205
303-308-9600
www.logicaldevices.com

A Different Kind of Line Amplifier

What does it take to build a successful line stage? This author shares his design experience to show you how it's done. **By John L. Stewart**

Many line drivers have claimed and sometimes delivered better performance. Here is another look at the problems encountered on the way to a circuit able to drive both terminated 600Ω transmission lines or a length of high-capacitance audio cable.

Each of the two amplifiers in this article has its own unique problems, the most important of which I have investigated in some detail.

THE OUTPUT STAGE

Among several possibilities, there appear to be two favorite topologies. One of them is in fact a power-amplifier stage, usually a triode with a grid driven by a transformer. The other is some kind of single triode or a pair of twin triodes connected in a variation of the so-called mu follower.

The first of these is well able to drive either of the loads I've referred to. However, at high levels it may deliver more distortion than you can tolerate. It also tends to be an expensive solution, since it needs two wideband transformers.

The second may be adequate at low levels driving a short piece of audio cable, many of which bring with them 30pF per foot of capacity. It doesn't take a rocket scientist to realize you will soon have more capacity than is practical for the mu follower to drive. It will go into overload quickly as the current limits of the upper and lower triodes are reached, and this will limit the circuit's ability to drive either capacitive loads or a terminated 600Ω transmission line.

THIS MIGHT WORK

Instead of the usual triode working into an output transformer, I tried something different (*Fig. 1A*). The circuit appears to be somewhat unusual, but it is able to drive commonly encountered loads with authority. Using a common triode (or triode-connected pentode), the signal is driven into a split-primary audio output transformer that was originally meant for a push-pull amplifier. The 600Ω output is taken from one of the 43% ultralinear taps, which results in a load of 13kΩ reflecting back to the triode.

All the curves showing distortion in triode versus load impedance tell us that distortion increases as load impedance diminishes. In a power stage, you design for a load impedance that is at least twice, and more often three times, the plate resistance. A triode-connected 6K6GT used at this stage (I tried) has a published plate resistance of 2.5kΩ (at 250 plate volts). Thus the plate load is five times the plate resistance, and low distortion is guaranteed.

There are other advantages to this split connection. It provides local negative voltage feedback of half the total output signal, which further reduces both the residual distortion and the driving resistance as seen by the load. The output capacitor only needs to be low voltage, but nevertheless of good quality.

You must take care with the primary

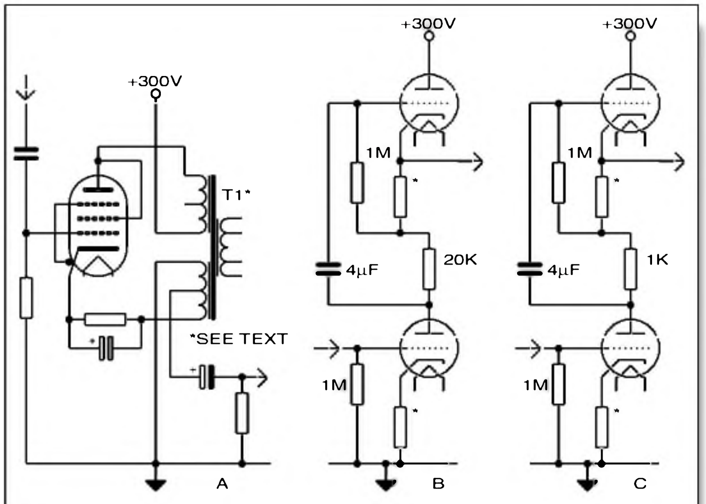

FIGURE 1: Line drivers.

TABLE 1
DISTORTION, GAIN, AND SOURCE RESULTS OF POWER STAGE—300V POWER SUPPLY

CONDITIONS			RESULTS			
TUBE	PLATE MA	LEVEL V RMS	GAIN	2^{ND} –DB	3^{RD} –DB	SOURCE RESISTANCE
6BL7GT	9.4	2 into 1.2k		53.7	58.5	–
6BL7GT	9.4	7 "		29.3	36.3	–
6BL7GT	8.5	2 "		47.1	63.3	336Ω
6BL7GT	8.5	7 "		29.9	38.5	"
6BL7GT	8.5	2 into Open		52.3	67.3	"
6BL7GT	8.5	7 "		43.1	–	"
6BL7GT	8.5	20 "		47.1	40.3	"
6BL7GT	8.5	30 "		26.7	28.2	"
6BL7GT	27.5	2 into 600Ω	X7.5	44.9	64.7	555Ω
6BL7GT	27.5	7 "		33.3	51.4	"
6BX7GT	41	2 "	X5.8	54.6	57.5	305Ω
6BX7GT	41	7 "		43.2	58.1	"
6V6GT	38.3	2 "	X0.34	56.8	66.0	60.7Ω
6V6GT	38.3	7 "	"	48.2	64.9	"
*6V6	38.3	2 "	X5.5	58.3	66.9	60.7Ω
*6V6	38.3	7 "	"	51.2	67.0	"

*Driven by 6SN7. See text.

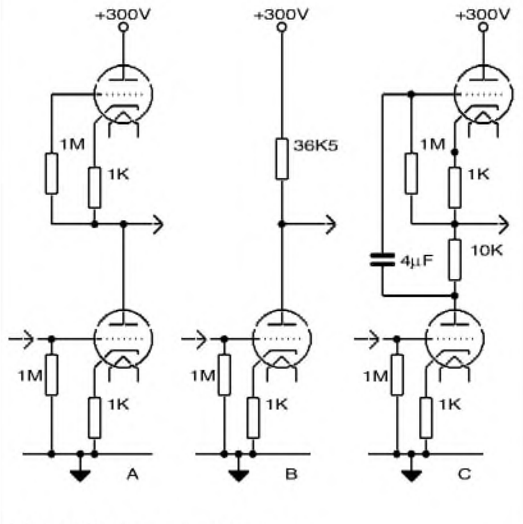

FIGURE 2: Some mu followers.

connections to achieve proper phasing. Otherwise, no output will result. You do not normally use the secondary, although it could provide a second overall feedback path when properly connected. You would connect feedback from the secondary to the cathode of the input stage. If the output-signal level increases, you need to reverse the transformer secondary connections. The secondary can also directly drive a loudspeaker with ease.

Audio power transformers with split primaries are not common, but Hammond, Sowter, and others can easily provide these. If you have a transformer specially made for this application, it should have an air gap, since the circuit is not balanced for DC, and core saturation could be a problem. The transformer I had on hand for this project is a Japanese device I acquired around 1960 with the intention of using it in a push-pull power amplifier wherein the DC would be cancelled. It seems to have enough iron so that I had no core-saturation problem. Nominal impedance of this transformer is 8kΩ.

A COMPARISON

In order to get comparative data, I tried a mu follower line stage using several different biasing conditions (*Fig. 1B*). I used a 6BL7GT, since it is able to deliver a large output if properly utilized. I provided bias for the first condition with 100Ω biasing resistors, which resulted in a standing plate current of 9.4mA. I set another condition using cathode resistors of 200Ω, which yielded a standing current of 8.5mA (*Table 1*). Distortion measurements were made using a Pico Technology ADC-100 analogue to digital converter. The signal source was an HP 200CD oscillator.

Clipping begins at 7.5V RMS when driving a 1.2kΩ load. With a pair of 6BL7GTs (one each for the upper and lower triodes), these results could be directly related to a 600Ω load.

Performance is reasonably good, although it could use some help. Since the load is less than the plate resistance at this operating point, you'd think the distortion would be high. The answer appears to be that at 2V RMS only a 1.67mA AC (4.71mA P-P) signal is required in 1.2kΩ. That is still a large part of the 9mA standing current. Not shown in the test results is an ominous rise in higher-order harmonic distortion.

If the 6BL7GT were run at a higher plate current, the results would be better, but you would need to increase the plate-voltage supply to satisfy the voltage drop across the 20k resistor. Alternatively, you could reduce this resistor, but then distortion would increase.

I tried a third condition wherein the standing current was increased to 27mA (*Fig. 1C*). Notice the 20k resistor driving the upper grid was reduced to 1kΩ. This causes the source resistance to increase to 555Ω.

By replacing the 6BL7GT in *Fig. 1C* with a 6BX7GT, I established a fourth condition. The increased plate current resulted in a further improvement of the test results. Finally, test results of the split-load line stage are included in the test data for comparison (see *Table 1* entries for 6V6GT).

THE INPUT STAGE

From the beginning of this project, I had intended to use a mu-follower as the input stage because of its reported low distortion and large voltage-output capabilities. On the way to the final circuit, I came upon some not-well-documented shortcomings of the simpler versions of this circuit.

In its simplest form, the mu follower uses identical triodes in the upper and lower parts. These are equally biased in order to make available the largest possible voltage to each of the triodes so that you can realize maximum voltage output. At this point, I used a 6SN7GT with 1kΩ biasing resistors, resulting in a plate current of 4mA so that good bandwidth could be ensured. The plate curves of the 6SN7GT show the plate resistance to be 10kΩ under these conditions (*Fig. 2A*).

Using the formula for calculating plate resistance with cathode feedback, the operating plate resistance with a 1kΩ cathode resistor is 30kΩ. That would seem a worthwhile improvement as the plate load for the lower triode. However, the lower half has also gone to 30kΩ, which sets up the condition where the load is one times the plate resistance.

When you treat the upper and lower triodes equally, the only advantage appears to be the low driving impedance available at the upper cathode. That would be useful if this stage were driving the transmission line.

The upper triode will always present to the lower triode a load equal to its own plate resistance. As well, that load will be the nonlinear curve of the upper triode's plate characteristic, guaranteeing that you will always have more distortion than need be.

OTHER TESTS

I made a further test of this conclusion by inserting in the plate circuit a 36.5kΩ resistor in place of the upper triode (*Fig. 2B*). This maintained the same operating conditions for the lower triode so that a valid comparison was possible. I measured distortion at 10V RMS

TABLE 2
RESULTS OF DISTORTION MEASUREMENTS AT 10V RMS OUTPUT

CONDITION/HARMONIC	2^ND^	3^RD^
6SN7GT with 36.5k load	–47.8	–67.4
6SN7GT with 6SN7GT load	–49.1	–67.8
6SN7GT with 6SN7GT active load	–54.1	–66.8
6BL7GT with 36.5k load	–48.4	–67.5
6BL7GT with 6BL7GT load	–50.1	–67.3
6BL7GT with 6BL7GT active load	–54.9	–68.7

on both circuits (*Table 2*). The resulting driving impedance of this configuration is 16.5kΩ (36.5kΩ plate load in parallel with 30kΩ plate resistance), which is more than adequate to drive the output line amplifier.

In another test, I substituted a 6BL7GT into the circuit. The results indicated that the simplest form of the mu-follower circuit has little or no advantage aside from a reduced source impedance for the following stage over a plate-load resistor. I also concluded that a high-perveance triode such as a 6BL7GT is little better than an ordinary triode such as a 6SN7GT when operated at the same plate current.

In a final test of this part of the circuit, I inserted a 10kΩ resistor between the two triodes (*Fig. 2C*). Driving the upper grid with the signal developed across this resistor makes the upper triode an active current source. The result is a load of considerably higher resistance for the lower triode. Measurements showed the new active load for the lower triode to be 210kΩ. The measured distortion was about 6dB less than the simple load-resistor case.

Some of the differences shown in the test results are within the error range of the test equipment. The 6BL7GT may be 1dB better than the 6SN7GT. The circuit using the active current source is a definite improvement. I settled on the 6SN7GT because of its availability and good performance in the active current-source version, while its power requirements are modest.

In summary, the mu-follower output line-stage advantages are simplicity, no output transformer, and gain in the output stage. The advantages of the split-load output line stage are very low output impedance, very low distortion, and the ability to drive a loudspeaker directly.

THE FINAL CIRCUIT

This article should allow you to make a choice of which circuit best fits your needs. I chose RC coupling in the split-load version (*Fig. 3*), which lets each amplifier stage work where it functions best. V1 is a 6SN7GT, but a 6BL7GT works just as well. Any medium-mu triode such as a 12AU7, 5814, or 5687 would do.

V2 in the final circuit is a 6V6GT. A 6K6GT also works well, as would several other triode-connected pentodes and common triodes. Plate-power rating is an important consideration here. The tube you use needs to dissipate about 10W with a 300V power supply, and plate current should be around 30mA. You can make an adjustment by varying the 330Ω cathode resistor. The primary resistance of the transformer's winding forms part of the biasing circuit for this stage.

Performance of the complete split-load version is summarized in the *6V6 entries of *Table 1.* A comparison of the results with those for the 6V6GT show a small but measurable distortion reduction. It seems that some distortion cancellation is at work here between the 6SN7GT and the 6V6GT.

"ENHANCED TRIODE" POSTSCRIPT

During development of the line amplifiers there was a requirement for tube specifications not obvious from those published. This led me to utilize simple circuits to measure them. I derived some information from data in the published curves, and I describe both methods.

Accurate measurement of tube parameters normally involves the application of some rather complicated bridge circuits. The methods I describe are more convenient for the average experimenter, and the results obtained are accurate enough for the analysis here.

On one of the line amps, I tried a direct connection from the upper cathode of a mu stage to the screen grid of the output stage, hoping this would provide DC coupling and eliminate a few parts. The result was disappointing, but it also led to some interesting conclusions with regard to the so-called "enhanced triode" mode.

Performance of the circuit while the output stage was connected in the enhanced-triode mode indicated low transconductance (G_m) through grid 2, so I decided an investigation of grid-2 G_m was in order. From the performance of the circuit, I knew grid-2 G_m was low, but how low?

G_M AND PLATE RESISTANCE FROM THE CURVES

Several pentodes and beam tetrodes listed in various RCA tube manuals show screen grid versus plate curves wherein you can directly calculate the G_m and plate resistance. For example, on p. 320 of RC-19 for the 6973, you can see that when the screen voltage is stepped in 50V increments, the plate current changes about 30mA.

Since G_m is measured in mA/V, you need only to divide, and in this case the result is 0.6mA/V. Compare that to the G_m through grid 1, which under similar conditions is 4.8mA/V. That is not very enhanced.

The plate resistance is the same whether you're looking at the curves for grid 1 or grid 2. That is easily seen by inspection of the two graphs. Another way of considering this is to realize that plate resistance is change of plate voltage versus change of plate current (Ohms law). Plate-resistance specifications are always given with the grids held constant.

If the operating conditions are those published for 250V plate and screen, then plate resistance is about 73kΩ. That won't provide much damping for a loudspeaker in an enhanced-triode connection anymore than it would as a straight beam tetrode. You'll need to apply negative-voltage feedback to gain control.

Another example of a tube which could provide power and lots of it in an output stage is the 6DQ5 (see pp. 198–199 of RC-19). The upper part of the curves would seem to indicate a G_m of 6.8mA/V at grid 2, but you would never

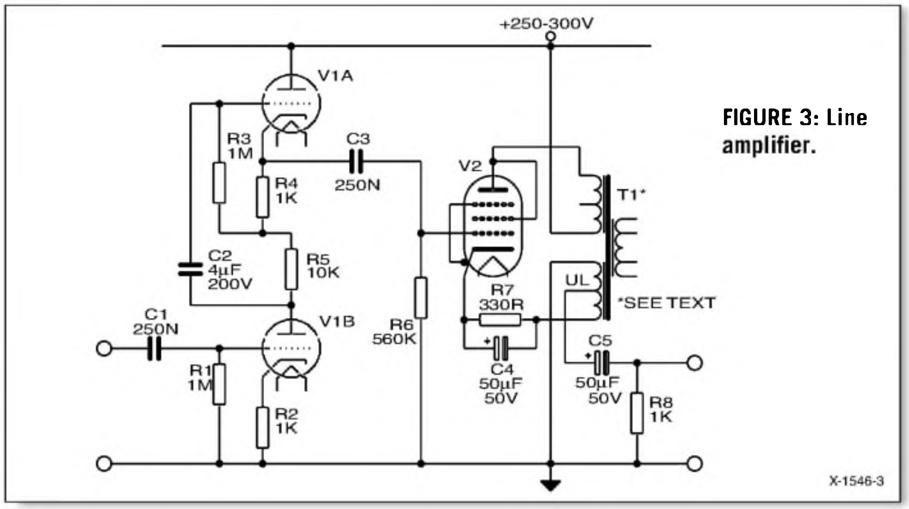

FIGURE 3: Line amplifier.

SPLIT LOADING

Some confusion may arise when you try to understand how to connect a split-load transformer. A comparison with the common push-pull stage will help, since it uses a similar output transformer. Winding polarity is indicated by dots (*Figs. 4* and *5*).

First, you can make the common assumption that current runs downhill, which helps a lot, even as it does for those working in the semiconductor industry.

In *Figs. 4* and *5*, the arrows indicate the commonly assumed direction of current flow, from positive to negative. In the case of the push-pull circuit, these currents are as equal as possible and running in opposite directions so that they will cancel most of the DC flux in the transformer core.

Not so in the split-load connections. Flux resulting from current in the two parts of the transformer winding is additive and may result in core saturation. You need a core with a built-in air gap, just as if you were dealing with a single-ended amplifier.

Again referring to the push-pull circuit, for a useful output to occur, you'll need as nearly as possible equal but opposite voltages at each end of the transformer primary. Since the grids are driven equal and opposite, that condition will result.

The same conditions are needed in the split-load circuit to produce an output. When the tube is turned on by a positive-going signal on its grid, the plate will go negative, while the cathode goes positive in order to follow the grid. A useful output occurs.

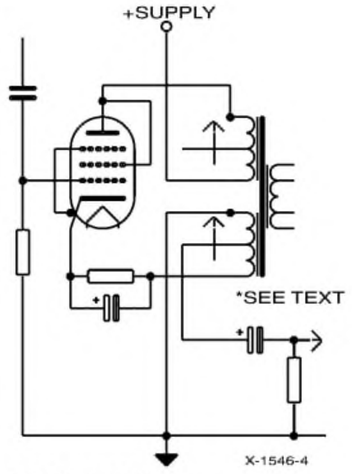

FIGURE 4: Split load.

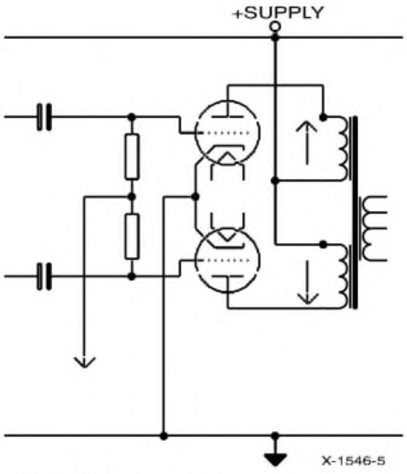

FIGURE 5: Push-pull.

go there in a real audio circuit. Plate dissipation in the upper part of the graph exceeds 100W. These tubes were designed for sweep circuits, where they endured either large voltages or large currents, but not at the same time. At a more realistic operating point, the grid-2 G_m drops to 4mA/V. Compare that to 10.5mA/V as a beam tetrode.

I used many of these as pulse generators in a core-memory research program in 1960. When driven hard, they are able to produce 1.5A pulses with rise times of 30ns into a 50Ω transmission line, but only for microseconds.

Again the plate resistance is the same whether you apply the signal to grid 1 or grid 2. From the curves in the safe operating region, this appears to be about 4kΩ, which is a long way from what you can obtain from a 2A3 or 300B. You will again need some negative-voltage feedback here to get control of a speaker load.

The only real advantage of the enhanced-triode mode appears to be the large plate-dissipation ratings of the tubes used. That is the bottom line when it comes to power.

TESTING FOR G_M AND PLATE RESISTANCE THROUGH GRID 2

I used the circuits of *Figs. 6* and *7* to measure the G_m and plate resistance of some tubes wherein the screen is the driven element. In each case, you need two separate power supplies isolated from one another. For this test, I used as the plate supply a regulated vacuum-tube power supply I had built about 1960.

For the screen, I needed to simulate conditions as they would be in the proposed enhanced-triode mode. I connected 17 9V batteries in series, which provided 158.6V—about the DC voltage available at the upper cathode of the input stage previously described. For these tests, current from the screen supply is small and intermittent, a technique that worked well. You must be sure to apply the plate voltage before the screen. The screen should never be left on alone.

Grid-1 bias was provided by one, two, three, or four 9V batteries as needed. The 250nF capacitor on grid 1 is a noise bypass to ground.

GRID-2 G_M

Referring to *Fig. 6*, I first applied a 300V potential (PS1) to the plate of the tube under test. I connected a 1kHz audio signal to grid 2 through the grid-2 battery (PS2) and measured it with a DMM or oscilloscope. At the same time, I measured the resulting AC voltage developed across the 100Ω current-sampling resistor.

The reason for the double power supply becomes obvious here. You need to isolate the screen current from the measurement. You

TABLE 3
MEASUREMENT OF G_M WITH GRID 2 AS CONTROL

CONDITIONS				RESULTS		
TUBE	G1 VOLTS	G2 VOLTS	TP1 V AC	TP2 V AC	G_M MA/V	I PLATE MA
6K6GT	−9	158.6	13.38	0.5	0.374	22.3
6K6GT	0	158.6	14.50	0.8	0.552	64.0
6V6GT	−9	158.6	12.67	0.5	0.395	24.0
6V6GT	0	158.6	12.33	0.694	0.563	72.0
6Y6GT	−18	158	12.51	1.51	1.21	63.0

TABLE 4
MEASUREMENT OF PLATE RESISTANCE

TUBE	PLATE AC VOLTS APPLIED	MILLIVOLTS ACROSS 100Ω SAMPLING RESISTOR	MICROAMPS AC CURRENT	PLATE RESISTANCE KΩ
6V6GT	28V	34mV	340µA	82kΩ
6K6GT	28	28	280	100
6L6GC	28	59	590	47.5

should measure only the plate current, since it is the change in that versus the voltage change at grid 2 that you need.

Measurements taken while grid 1 is at 0V must be performed quickly to avoid overheating the tube. Results obtained for various tubes are shown in *Table 3*. All of the results are poor for a power tube.

GRID-2 PLATE RESISTANCE

Note the similarity of *Fig. 7* to the circuit of *Fig. 6*. Plate resistance is a measure of the result of changing plate voltage on the plate current. In pentodes and beam tetrodes, the plate resistance tends to be high, and for this reason the driving signal must be large to produce a measurable change in the plate current. The audio oscillator is again at 1kHz, but this time applied through a step-up transformer.

The transformer also isolates the

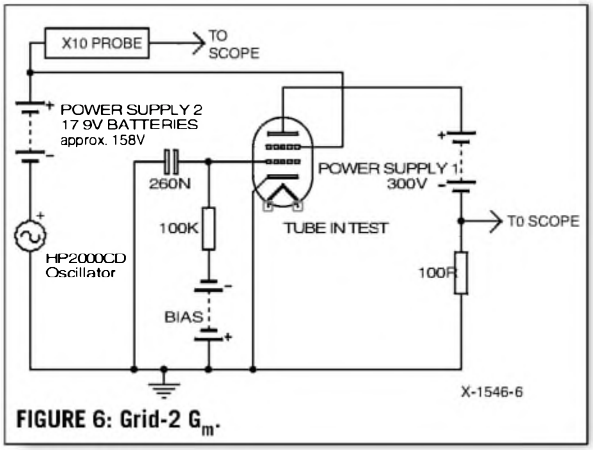

FIGURE 6: Grid-2 G_m.

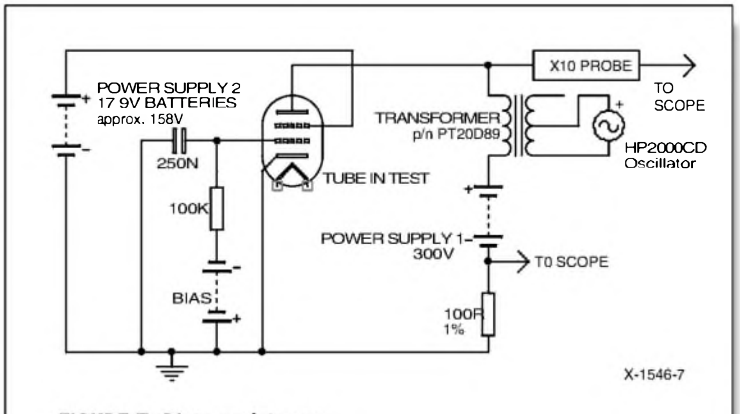

FIGURE 7: Plate resistance.

audio oscillator from dangerous voltages. The transformer I used here is available as part number PT-20D89 from Antique Electronic Supply, 6221 S. Maple Ave., Tempe, AZ 85283, 602-820-5411.

Plate-resistance results appear in *Table 4.* Note here that, aside from biasing, grid 1 shows up nowhere in the measurements, yet the plate resistance looks nothing like a triode. The measured results are not indicative of common audio triodes, enhanced or otherwise.

If you were to plot the plate family of curves with grid 2 as the variable, they would look very much like those of a zero-bias Class B triode, well known in RF-transmitter circles. ❖

EQUIPMENT USED

- HP 200CD Oscillator
- HP 302A Wave Analyzer
- GN GAG-810 Oscillator
- Rohde & Schwartz BOL 4 Trace 100MHz Scope
- RS 22-168A DMM
- Sanwa AX 303-TR Analogue Multimeter
- PicoScope ADC-100 Virtual Instrument
- Electronic Workbench Software

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The Pull-Down "T" Attenuator

This high-quality design addresses the drawbacks traditionally associated with attenuator layouts.

By R.K. Stonjek

In the early days of moving-coil loudspeakers, builders would place drivers into a baffle board giving little or no consideration to the enclosing cabinet. Later, in an attempt to improve low frequencies, they devised ever larger cabinets to keep the acoustic radiation—coming from the rear of the speaker—from encountering the front radiation. Only in the 1950s did a speaker company, Acoustic Research (AR), think of designing the speaker and cabinet in concert—no longer was the cabinet an afterthought.

ADD-ON SOLUTIONS

The phrase "high fidelity" originally referred to a sound system's ability to reproduce high frequencies. To achieve this, many speaker builders simply "added on" a high-frequency driver. Later, in an attempt to improve power handling, they added a capacitor to the tweeter. Only much later were crossovers introduced at the design stage as an integrated feature of the speaker system.

Biamping still has that add-on feel about it. Often, designers simply add an extra amplifier and subwoofer to a speaker system to improve the lows, much as the tweeter was added in the '40s to improve the highs. Alternatively, you can separate the lowest-frequency driver, usually the subwoofer or woofer, and amplify that separately in an existing sound system. Only the ActiPass Crossover System (*SB* 8/00) is a fully integrated active-passive crossover system with nothing added on.

In order to improve power handling and lower distortion at a given sound pressure level (SPL), it has become common practice to double up on the bass driver, and sometimes the midrange as well. This is definitely an add-on arrangement, with the only system adjustment to accommodate the new format being a change in the tweeter's shelving resistor.

Ultra Fidelity Sound Systems strove to design a very high-quality, fully integrated sound system meant to be a matching system from one end to the other—no components in the chain were simply "added on." We achieved this goal in 1996, just as the manufacturing business folded without producing a single unit. Two complete prototype systems exist, the "Concert" Mk.1 & 2. Another, the Mk.3, was partially built, and I am currently building an Mk.4 from the original blueprints as a kind of "capstone" to the failed project (see *SB* 8/00 article).

ABOUT THE AUTHOR
R.K. Stonjek is the owner of Ultra Fidelity Sound Systems, based in Tasmania, Australia.

THE PULL-DOWN T

The pull-down T attenuator is a type of stepped attenuator used to control the overall output of a sound system. Functionally, it plays the same role as the volume control. Some time ago, using a pair of fairly inefficient loudspeakers and a CD player known for its low output, I found I could bypass the preamplifier and plug the CD player directly into the power amplifier without suffering deafeningly loud levels.

The quality improvement—particularly in the treble, detail, depth, and apparent sound stage—was remarkable, but where was the extra quality coming from? What had I removed from the chain that was causing a sound-quality bottleneck? Since the preamplifier's frequency response was already well beyond my hearing range and the op amps I used were transparent, this left only the volume control itself—the potentiometer.

Stepped-attenuator circuits available to hi-fi enthusiasts have been imitation volume controls that you can simply use in place of the volume pot—a straight component substitution. The circuit is not designed around the attenuator and does not utilize its unique advantages. A typical high-quality input stage looks something like *Fig. 1.* Note the low resistance of the volume control. This is an attempt to reduce the thermal noise that volume controls contribute, particularly when set to around half volume.

To reduce noise, you must protect the volume pot from any DC in the circuit, but once you put in one capacitor you're compelled to add more. Furthermore, if there is no DC path to ground on the op amp's positive input, it is liable to "latch up" (snap to one of its voltage rails). While most op amps will still function while passing on ±15V DC, and the rest of the circuit is

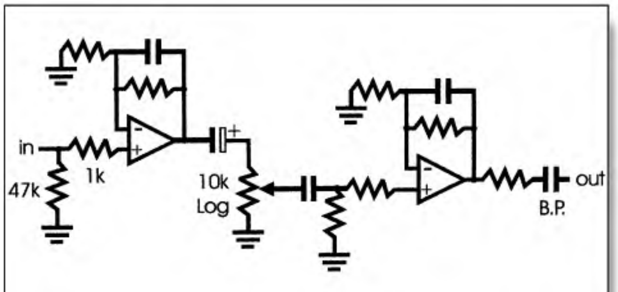


FIGURE 1: A typical high-quality preamplifier.

isolated via a capacitor, you can clearly hear a marked fall in quality and increase in distortion. The large-value capacitors required to filter this voltage and maintain frequency response often need to be the electrolytic or bipolar capacitors that you should never place in the signal path of high-quality equipment.

THE LADDER ATTENUATOR

A regular stepped attenuator looks something like *Fig. 2.* This is the "ladder" attenuator, where all the resistors are in circuit all of the time. The wiper simply selects various steps on the ladder as if it were a potentiometer, but making the steps right is notoriously difficult, requiring precision resistors.

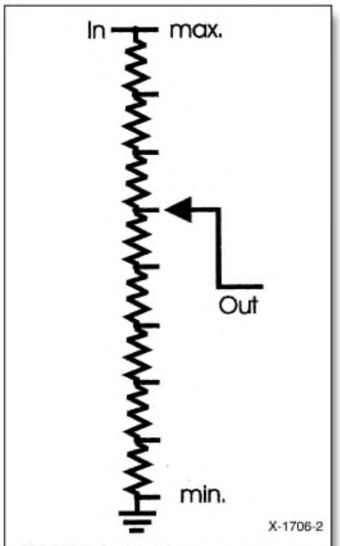


FIGURE 2: Ladder attenuator.

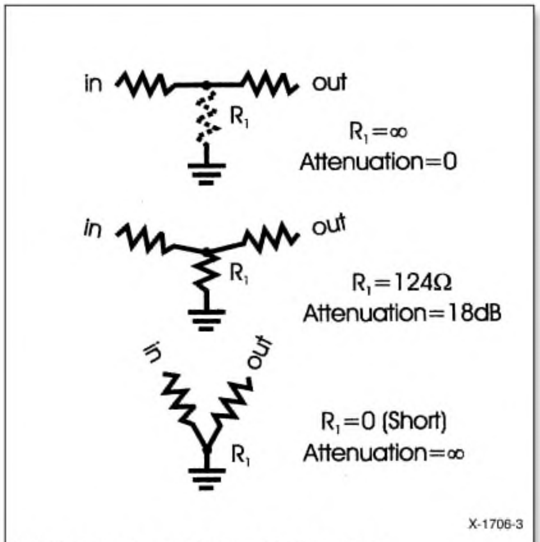
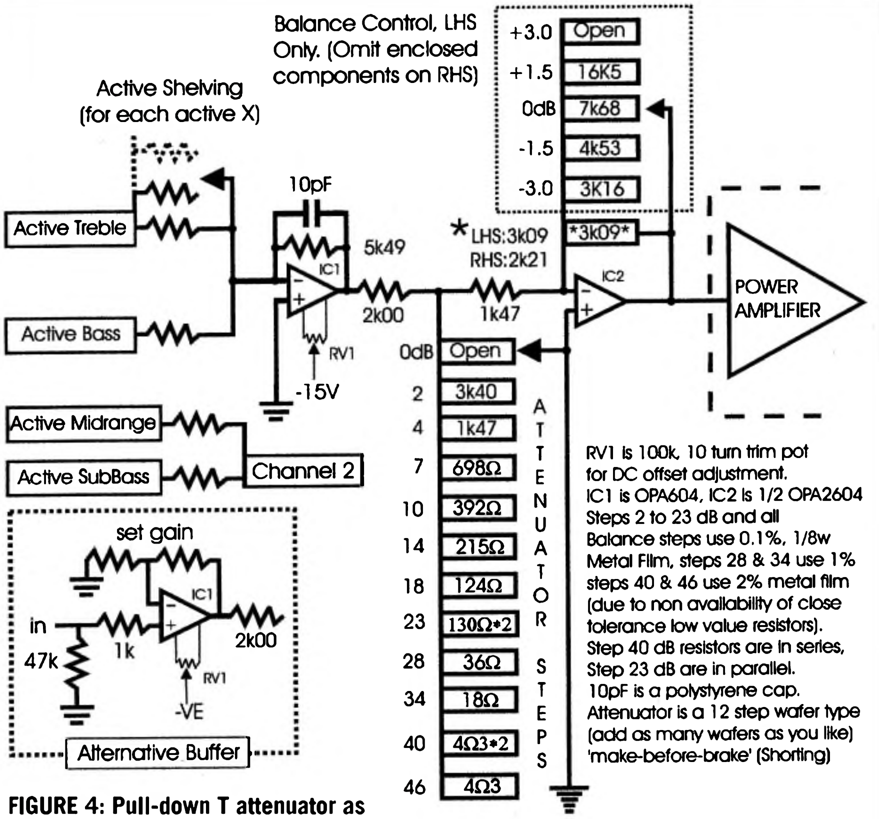


FIGURE 3: The pull-down T attenuator.

Figure 3 shows how the pull-down T attenuator works in principle. The input is buffered to two resistors that form the "T," the center of which is pulled down to ground by the attenuator. The output goes to the inverting side of another buffer. This inverting input forms a "virtual earth." The attenuator switches between a real and virtual ground. The pull-down resistor at the center of the T is switched by the attenuator—a different resistor for each step. At full volume, the pull-down resistor is removed altogether.

Figure 4 shows the pull-down T as it was used in the Ultra Fidelity Concert Sound Systems. The buffer IC also acts as the mixer for the ActiPass crossover. To achieve balance trim, you raise or lower the left-hand side channels by up to 3dB in 1.5dB steps. This setup ensures no crosstalk from the balance control. An alternative buffer for nonactipass systems is also shown.

With the components shown, the entire attenuator stage has the same specification as the op amps used. Using the



RV1 is 100k, 10 turn trim pot
for DC offset adjustment.
IC1 is OPA604, IC2 is 1/2 OPA2604
Steps 2 to 23 dB and all
Balance steps use 0.1%, 1/8w
Metal Film, steps 28 & 34 use 1%
steps 40 & 46 use 2% metal film
(due to non availability of close
tolerance low value resistors).
Step 40 dB resistors are in series,
Step 23 dB are in parallel.
10pF is a polystyrene cap.
Attenuator is a 12 step wafer type
(add as many wafers as you like)
'make-before-brake' (Shorting)

FIGURE 4: Pull-down T attenuator as used in the Ultra Fidelity Concert Sound System.

X-1706-4

Burr-Brown OPA604, you can expect harmonic distortion of 0.0003% and a flat frequency response from DC to more than 400kHz. Note the DC-offset trimpot, a ten-turn type. Once it is set, you never need to touch it. Trimming this resistor prevents "popping" between attenuator steps.

The steps of the Ultra Fidelity Attenuator may seem curious at first. Logically, you might tend to choose 3dB steps, but in practice this is too limiting. The expanded scale used "sounds" more logical; you find that all the steps are useful. At zero attenuation, the amplifier is at maximum for a 1V input. Note that there is no infinite attenuation—the infinite setting is a waste of a good attenuator step. The highest attenuation is the lowest practical listening level. With the close-tolerance components shown, the steps as marked are within a fraction of a decibel to those tested.

ADDITIONAL DESIGN NOTES

Both balance and attenuator switches are "make-before-break" types. The switch body used for the attenuator has a very stiff action, but you can remove one of the two ball bearings used to make the switch "click" into position.

This design may not be stable when inexpensive op amps are used.

You can solder the resistors directly onto the attenuator in a star-burst fashion, and then connect the ends to form a ring around the attenuator wafer.

The OPA2604 and OPA604 op amps' distortion figures are quoted for a load of 1kΩ. The pull-down T has a minimum of 2kΩ.

You can add the active-crossover shelving shown to all the orders—e.g., bass, midrange, treble—that you intend to use, though it is advisable to leave the shelving off at least one, preferably the midrange in a three-way system, and the bass in a four-way system. The Concert Sound System had the bass shelving set to the overall gain of the system, so that a 1V input would drive the bass speaker to its maximum power handling. There was no accommodation for low-level input signals. ❖

Low Down Power, Part 2

Reviewed by Thomas Perazella

I measured all amps for frequency response, maximum power output into both 4 and 8Ω at 20Hz and 150Hz, and distortion into 4Ω at power levels from 2W to maximum power. For distortion measurements, I chose a reference frequency of 50Hz to determine the power level. Since not all frequency responses were flat, you may see some strange humps in the distortion curves at some power levels because boosts in the frequency response cause the output power to go above the reference value.

In addition, I measured maximum gain and noise relative to both 1W and maximum voltage into 8Ω. I determined noise figures with shorted RCA plugs inserted into both RCA input jacks, and took results at three volume-control settings—minimum, 20dB gain, and maximum. *Table 2* lists the maximum power output, gain, and noise figures of the four sub amps.

APEX JUNIOR

The Junior turned out to be the junior of this group as related to power output. It fell short of its claimed 130W by 16 at 150Hz and 35 at 20Hz.

Looking at the 8Ω figures reveals the source of the problem is a weak power supply. The stiffer the power supply, the closer to double will be the 4Ω output compared to the 8Ω output. If this supply were stiff, the claimed 130W would have been reached. However, in terms of dollars/W, the junior was the king at just over 77.5 cents/W. Noise was respectably low and gain was sufficient.

Frequency response was very good as shown in *Fig. 1*. The top curve indicates the response at the 125Hz crossover setting, with the bottom representing the 60Hz crossover setting. Below 20Hz, there is a rolloff at 12dB/octave. This will help prevent excessive cone excursion, especially in vented enclosures where the driver is unloaded below port tuning. Above 20Hz, the response is relatively flat until the crossover frequency is reached.

I made three distortion measurements into a 4Ω load, one each at 2W (*Fig. 2*), 50W (*Fig. 3*), and 100W (*Fig. 4*). The distortion was very low at all measurements except for a slight rise below 35Hz at 100W. Overall, the measurements were quite good, especially in light of the modest price tag.

MARCHAND PM31

This amp was a good performer. It was rated at 150W into 4Ω and produced that level at 150Hz. At 20Hz, the maximum power dropped only slightly to 144W, showing the strength of the power supply. This power did not come cheaply. At $1.97/W, it was the most expensive in terms of dollars per watt. However, subs do not live by power alone, and the other performance measurements were quite good as well. Frequency response was almost ruler flat, being down only slightly over 1dB at 10Hz (*Fig. 5*).

Distortion results at 2W, 50W, and 150W are shown in *Figs. 6, 7,* and *8*, respectively. Distortion never goes above 0.1% at 2W, remains below 0.9% at 50W, and hovers around 2–3% at 150W.

The PM31 also came in with the lowest noise figures for any of the sub amps when in the operating mode. However, it also had the lowest gain, at 26dB, which could present a matching problem in

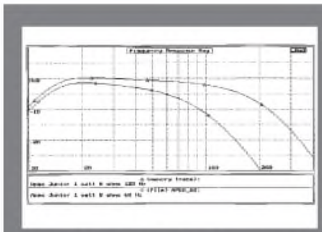


FIGURE 1: Apex Junior frequency response.

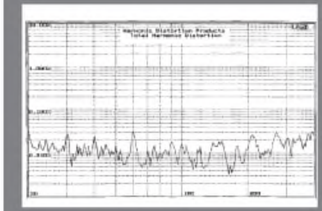


FIGURE 2: Apex Junior distortion—2W.

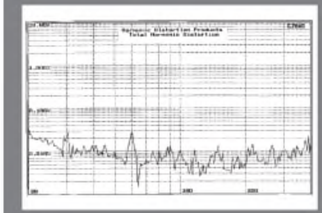


FIGURE 3: Apex Junior distortion—50W.

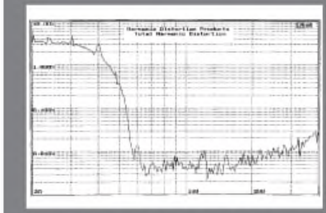


FIGURE 4: Apex Junior distortion—100W.

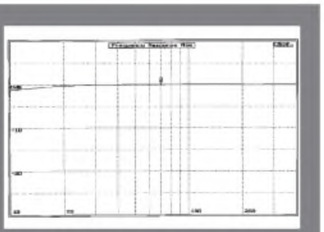


FIGURE 5: Marchand PM31 frequency response.

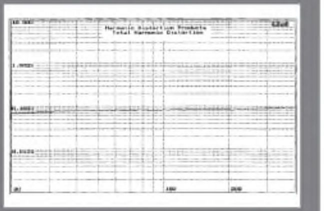


FIGURE 6: Marchand PM31 distortion—2W.

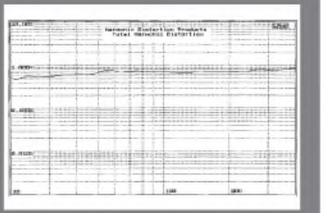


FIGURE 7: Marchand PM31 distortion—50W.

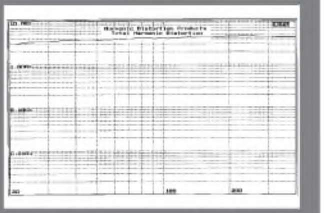


FIGURE 8: Marchand PM31 distortion—150W.

some situations. Overall, it provided a good showing, but was somewhat on the expensive side. All those heavy-duty parts do not come cheaply.

FARTS EXPRESS SW250A

This amp originally threw me for a loop when I measured it. My first test was for noise. If I had not tested other amps with the same setup immediately before, I would have thought my test setup was bad. My first reading showed a noise level only 33dB below 2.83V (1W into 8Ω). With the volume control set to maximum, it dropped to −15dB below 1W. I had never seen what appeared to be such a noisy amp.

To determine what was actually happening, I put a scope probe on the output and saw very little of the traditional 120Hz ripple you get from most power supplies. Instead, there was a lot of very high frequency hash. When I say high frequency, I mean in the range of 12MHz. That's even above dog-hearing country.

My true RMS AC voltmeters have very good high frequency response and were picking up this "noise." As I found out later, however, in real operation with a woofer, the amp was very quiet. The woofer could obviously not reproduce into the MHz range, so this hash was of no audible consequence.

The frequency response of this amp was far from flat, always exhibiting a rise in the 30Hz range. This is not uncommon for sub amps, as most drivers, especially very compliant ones, when placed in a small box exhibit a substantial rise in their resonant frequency. This boost helps restore some of the bass that would be lost due to the rolloff below resonance.

In theory it's not a bad idea, but I prefer a switch to defeat the boost if so desired. This amp had no such provision. Instead, it had an 18dB/octave rolloff below 30Hz to limit excessive excursion. Again, with a corner frequency that high, I would have preferred a method to either defeat the high-pass function or else lower the corner frequency.

Figure 9 shows the frequency-response curves with the crossover set for 160Hz and 40Hz. Both turnover points are actually a bit lower than indicated, but there are no real surprises here.

Because of the higher power rating of this amp, I actually made five distortion measurements. Power levels were 2W, 50W, 150W, 200W, and 250W. Results are shown in *Figs. 10*, *11*, *12*, *13*, and *14*, respectively.

Distortion at the 2 and 50W levels remains well below 1% at most frequencies except for a bump at 30Hz. At these power levels that bump does not appear to be excessive drive from the 30Hz frequency rise that was evident in the response curves. However, at 150W and above, the effect of the boost becomes clear. The distortion goes over 5% at 30Hz at 150W, over 8% at 30Hz at 200W, and over 14% at 30Hz at 250W.

Note that the scale has been shifted on the 250W graph to keep the results from exceeding the range. Looking at the 50Hz reference points on both the 200W and 250W graphs, it is clear that the amp is already into clipping at this point. The 30Hz boost only drives the amp further into clipping. In fact, maximum power at clipping was 186W.

Overall, this amp was the most controversial of the amps tested. The 250W rating was not met, but it still was quite powerful. The dollars-per-watt figure was $1.34, about in the middle of the pack. The amp stayed cool during operation and seemed to be more efficient than the others. The high measured noise did not translate to audible noise. It had the highest gain of any of the amps, providing the greatest range of matching with other parts of the system, but also the greatest difficulty in setting levels at the lower end of the volume range.

FARTS EXPRESS 300-794

This relatively new product from Parts Express is advertised as having an output capability of 250W into 4Ω, but the product sheet supplied with it states that the power is 272W into 4Ω. Who to believe? Actually, neither.

This amp had the highest output of any of the dedicated sub amps tested. It topped out at 286W into 4Ω at 150Hz, dropping to 237W at 20Hz. The price-per-watt figure was 79 cents/W, just slightly more expensive than the Apex Junior, but substantially more powerful.

Looking at the frequency-response curves revealed a smooth response, always peaking around 30Hz as with the previous PE amp. The most revealing characteristic of the response curves was the apparent lack of a defined plateau when higher crossover points were used. This amp has a continuous phase control, but that introduced some interesting results as can be seen in *Fig. 15*.

The top curve marked with a triangle represents the crossover set at 0° and 160Hz. The second curve marked with an X is with the crossover set at 0° and 40Hz. As you can see, there is a continuous drop in the output of the first curve from the turnover point of the 30Hz boost until around 160Hz, and then a very slight increase in the slope of the curve from that point forward. This is not your classic 160Hz crossover. The second curve looks more like a 30Hz bandpass filter.

The third and fourth curves are with the crossover set to 90° and 160Hz and 90° and 40Hz, respectively. The most immediate visible difference between these curves and the 0° curves is the substantial drop in amplitude as you rotate the phase control to produce a 90° difference.

By the way, the actual 90° point was not at the indicated 90° position on the rotary control. I determined the actual 90° point by using an oscilloscope. In fact, the phase change was quite nonlinear, with most of it occurring within about a 20° rotation of the phase control. This amplitude change will most likely produce a far greater effect on the perceived output of the sub than the actual phase shift of the signal. If you use this control, you must be careful to re-adjust levels with every phase adjustment; otherwise, you will be chasing two variables at one time.

The second most noticeable characteristic of the phase-adjusted curves is the apparent rotation of the curves around the corner frequency. It appears as though the two 90° curves have been rotated about 30° clockwise in relation to the 0° curves. As a result, the low-frequency rolloff of the 90° curves is shallower than the 0° curves, while the high-frequency rolloff is steeper. Again, I guess that the change in the slopes of the resulting frequency-response curves will have a greater impact on the perceived sound than the phase difference.

Also noticeable is the surprising similarity of the 90° 160Hz curve and the 0° 40Hz curve. This is not the result I would expect of two such varied crossover points. This means that you must be very careful when making these adjustments, as they interact to a substantial degree with the effects of each other. If you make multiple changes at a time, you may run around chasing your tail for quite a while.

Because of the power output of this amp, I again planned to make five measurements as with the previous PE amp. It actually turned out to be six measurements because as I was halfway into the measurement at 150W, the protective thermal switch on the amp tripped. This was the first time in this series of tests that a protective circuit activated. I waited a few minutes for the amp to cool down and repeated the test with no problems.

During these tests, taken with the LAUD measurement system, a series of sine waves at various frequencies is introduced into the amp and distortion measurements are taken. The duty cycle is about 50%; that is, the signal is on for 50% of the time and off 50% of the time. In between tests, I saved the results to a computer file, printed out the data, set a new power level for 50Hz, and began the test again. The duty cycle during the interim time is much below 50%.

To make sure there were no further interruptions in the testing of the 300-794, I allowed it to cool a few minutes before performing a subsequent test. I had no further interruptions and there appeared to be no lasting problems due to the thermal overload. I would not expect this to be a problem in normal use.

The distortion measurements

were quite good with this amp. The results are shown in *Figs. 16–20*. At any frequency, the distortion remained below 1% and below 0.1% above 20Hz, as long as the output was below clipping.

Starting with the 150W curve, the distortion begins to rise at 30Hz. This is because the bass boost centered near that frequency begins to drive the amp into clipping. The effect is more pronounced in the 200W curve and actually pushes the distortion over 10% on the 250W curve as the amp is driven heavily into clipping. Note again that the 250W graph has the scale changed to keep the curve from going off scale.

The results were good with the only caveats being the unorthodox operation of the phase control and the poor action of the crossover when set to higher frequencies.

AUDIOSOURCE AMP THREE

I saved this amp for last because it is really a different animal and not a direct comparison to any of the previous. However, it is a very real, although more complicated, alternate. The measurements are typical of a modern "perfect" amp.

For example, looking at the frequency-response curve of *Fig. 21*, you'll notice immediately that it is nearly a straight line. You might not notice, however, that the scale has been changed from 10dB per major division to 1dB. With the scale set the same as for all the other amps, the frequency-response curve disappeared into the 0dB line. Even with the greatly expanded scale shown here, there is only a slight droop of about 0.05dB at 10Hz. Enough said about frequency response.

With one channel driven, power output reached clipping at 319W into 4Ω. With both channels driven, clipping occurred at 267W. In bridged mono mode into 8Ω, the clipping point was 536W. The dollar-per-watt figure is about 65.5 cents/W. Remember, however, this cost does not include any electronic functions other than level control. An external active crossover will add considerably to the price.

Because of the high power output of this amp, I ran six distortion measurements. The last was at 300W with one channel driven. The results are shown in *Figs. 22–27*.

The curves from 2W to 150W are essentially in the residual noise of the measuring system, being below 0.01%. Up to 250W, the distortion remains below about 0.5%. At 300W, the distortion breaks 1% at 20Hz, but is below 0.5% from 30Hz up. Very good results, indeed.

LISTENING

Before describing the listening tests I performed with these amps, a disclaimer is in order. The sub I use in my system is an excellent reference but has spoiled me for most commercial subs. It consists of eight DV12 12″ woofers using a 450ft^3 loft as an enclosure. For details, refer to the article "True Bass" in *SB* 5/96.

In a nutshell, the f_3 is 12.5Hz and the sub can produce 122dB at 16Hz at the listening position before reaching 10% distortion. I did not think that I should use that woofer to test these sub amps, since it certainly would not be representative of most applications.

Instead, I chose two excellent 12″ woofers, the Audio Concepts SV12 and HSU Research ASW1201 and used both in the same 2.7ft^3 sealed enclosure. Since that enclosure was normally used for other purposes, I did not make cutouts to directly mount the amps in the enclosure, but rather connected the lead wires to the terminals on the enclosure.

The SV12 is the spiffier-looking of the two, with a cast frame, gold-plated terminals, and gold lettering on the dust cap. It also has an X_{MAX} of 12.8mm and nominal impedance of 4Ω. I've used this woofer before in other tests with good results. The last time I checked, price on this driver was $129.

The ASW1201 is a little more pedestrian in the looks department, with a stamped frame and solder terminal connections. The looks are very deceiving, however, as this is also a very competent woofer with an X_{MAX} around 13mm and a nominal impedance of 8Ω. I've used this driver for several years in the range between my sub and midrange drivers. The last time I checked price it was $109.

The SV12 has an f_S of 19Hz and the ASW1201 has an f_S of 29Hz. However, the SV12 has a relatively high V_{AS} and the HSU a relatively low V_{AS}. When mounted in the sealed box, the f_B of the SV12 combo was 40.4Hz, and the f_B of the HSU combo was 43.1. That provided a pretty close match for testing purposes.

The subs in this test covered the range up to 80Hz. Above that, I used two Sequerra Met 7 monitors. High-pass filtering for the Met 7s was provided by the high-pass section of a pair of Orban model 672A parametric equalizers. They provided a 12dB/octave slope at 80Hz. I positioned them on stands 4′ out from the front wall and 6′ apart. I positioned the sub in use in the front-right corner of the room.

To get a rough estimate of the sound-pressure levels produced during the tests, I used my Radio Shack sound level meter set to C weighting and fast response. Power for the Met 7s came from a second AudioSource Amp Three that I own. All source material was from CDs delivered by a Sony 707ESD player. The preamp used is a custom-built unit.

I chose the following source material to cover various types of bass, including piano, bass guitar, drums of different sizes, and very low frequency special effects:

Columbia C2K68519
Pink Floyd—*The Wall*

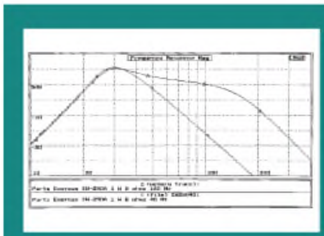


FIGURE 9: Parts Express SW250A frequency response.

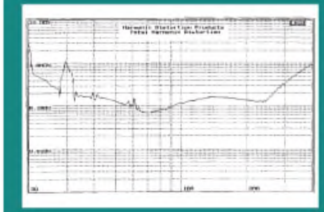


FIGURE 10: Parts Express SW250A distortion—2W.

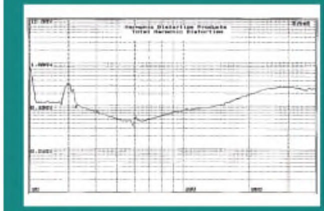


FIGURE 11: Parts Express SW250A distortion—50W.

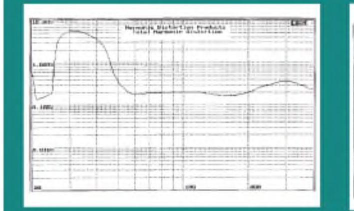


FIGURE 12: Parts Express SW250A distortion—150W.

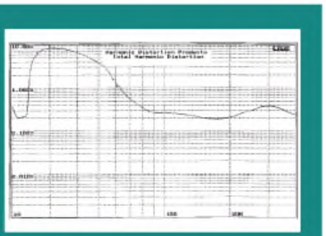


FIGURE 13: Parts Express SW250A distortion—200W.

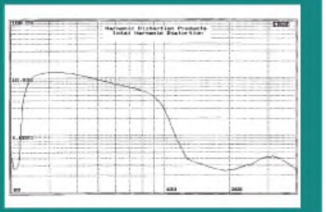


FIGURE 14: Parts Express SW250A distortion—250W.

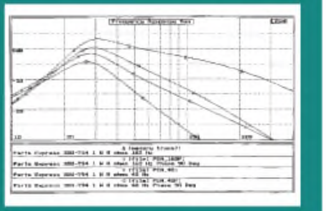


FIGURE 15: Parts Express 300-794 frequency response.

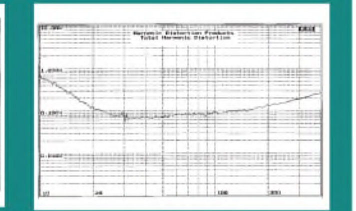


FIGURE 16: Parts Express 300-794 distortion—2W.

Telarc CD-80342
The Great Fantasy-Adventure Album—"Jurassic Lunch"
Elektra 9 61315-2
Sergio Mendes—*Brasileiro*
Tristar Music WK36852
Kodo—*Ibuki*
Columbia CK57424
Tony Bennett—*Steppin' Out*

I performed two complete sets of tests. The first used the ACI woofer in conjunction with all the amps and source material. The second repeated the test procedure with the HSU woofer. Before playing music with each amp, I listened for any audible noise at the listening position. Note that because of its late arrival on the scene, I did not perform any listening tests of the Parts Express 300-794.

A. PHASE ONE—THE ACI WOOFER

1. Apex Junior. There was no noticeable noise at the listening position.

- *The Wall*. This CD has a good balance of complex sounds including percussion and bass guitar. The bass was balanced with some limitation of the kick bass when levels went above around 98dB. In the mid-90s, the results were quite good overall.

- "Jurassic Lunch." This cut represents the scene in Jurassic Park where the lawyer gets eaten by the T-Rex. There are some 12Hz simulated foot steps that are low in intensity with other higher-level bass as well. With the Apex, there was some foot stomp. The overall bass was good, but there was not much floor shake.

- *Brasileiro*. The first cut, "Fanfarra," has really good percussion that becomes almost explosive. The result here was very good. The sound was clean with good transients. The second cut, "What Is This," has some very high-level low bass just past two minutes in. It was reproduced very well into the high 90s to 100dB.

- *Ibuki*. The drum work on the track, "The Hunted," is very dynamic and clean. The dynamics will crush a bad system. Here the drums sounded very good up to around 103dB, where they started to sound a little harsh.

- *Steppin' Out*. The Bosendorfer piano in this CD provides a good test of bass smoothness. Too much and it sounds bloated. Too little and it loses its warm character. In addition, the string bass is almost a little heavy to start with. Play it over a woofer with a lack of control and you have bass mush. The reference track was "Who Cares." The Apex/ACI combination did very well on the piano and Tony Bennett's voice. The bass was good, but just a little fatter than with my reference.

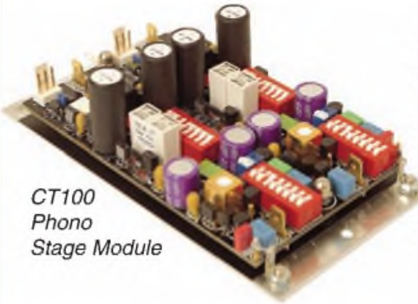
2. Parts Express SW250. There was just a very slight amount of noise at the listening position. It represented itself as a slight hum. Setting the level balance with this amp was a little difficult because of the high gain. I was working in the lower region of the volume control, and a slight change made a large volume change.

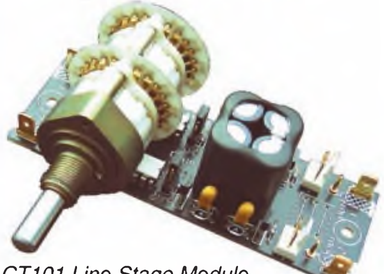
- *The Wall*. When the mid-bass level was brought to where it should be, the overall bass seemed a little heavy. The kick bass was good with not too much mid to upper bass.

- "Jurassic Lunch." There was not much foot stomp and not much floor shake.

- *Brasileiro*. "Fanfarra" had good initial percussion. The lowest notes were a little heavy but very clean. "What Is This" had good initial percussion, and the very low bass notes two minutes in were very strong and well defined up to around 105dB. Above that the sound became a little wooly. The very lowest bass was a little shy.

- *Ibuki*. The bass drum was very good. It limited around 105dB.





- *Steppin' Out*. The piano, bass, and voice all sounded good with this combination. It sounded better with a little more gain on the bass than with the other pieces.

3. Marchand PM31. This amp produced no audible noise at the listening position. Gain was a bit on the low side, requiring me to lower the drive to the monitors to get a proper balance even with its gain set to maximum.

- *The Wall*. Very natural bass, not at all bloated. Good transients on the kick bass. A little lighter on the deep bass, making it sound a little "faster." Clean at 105dB peaks.

- "Jurassic Lunch." Some 12Hz foot stomp, but not much floor shake.

- *Brasileiro*. "Fanfarra" had very tight and clean percussion. The peaks were clean to 108dB. Definition was very good. "What Is This" had good lead in percussion, but just a tad on the thin side. The low notes about two minutes in were very clean and tight, but not quite as deep. Output was good up to 105dB.

- *Ibuki*. This was very clean and dynamic up to 105dB.

- *Steppin' Out*. The piano had a warm and detailed sound. The voice was also very good, but the string bass was a little boomy.

4. AudioSource Amp Three. This amp produced no noise at the listening position.

- *The Wall*. The bass was very detailed. The kick bass was tight to around 105dB. A little thinner than the others.

- "Jurassic Lunch." Some 12Hz foot stomp, but not much floor shake.

- *Brasileiro*. "Fanfarra" had very clean, tight percussion to around 106dB. "What Is This" had very detailed bass in the introduction, but a little thinner than the others. Deep bass was clean and detailed to 106dB, but not quite as deep.

- *Ibuki*. Good, tight drum sounds to around 105dB.

- *Steppin' Out*. The piano and voices were very well defined, but a tad light. Strong bass, but a tad boomy.

B. PHASE TWO—HSU WOOFER

1. Apex Junior

- *The Wall*. Bass with kick drum good to around 102dB. Tight and punchy.

- "Jurassic Lunch." Not much floor shake, but good overall impact.

- *Brasileiro*. "Fanfarra" had good intro and follow on percussion to 106dB. "What Is This" had good opening percussion to 106dB. The later bass notes were tight and reached 106dB.

- *Ibuki*. The lead-in drum part was tight and punchy. The main part of the piece was good to around 105dB, getting a little harsh above that.

- *Steppin' Out*. The piano and voice were very good. The string bass was a little heavy.

2. Marchand PM31

- *The Wall*. The bass was very clean and natural sounding. The kick drum was clean to around 105dB.

- "Jurassic Lunch." Not much floor shake, but well defined.

- *Brasileiro*. "Fanfarra" had good opening percussion to 106dB. "What Is This" had both good opening percussion and deep bass to 106dB.

- *Ibuki*. Good impact on the drum to 106dB levels.

- *Steppin' Out*. The piano was very good with just a slight touch of leanness. The bass was very good with just a touch of fat.

3. Parts Express SW250

- *The Wall*. Good, full bass. The kick bass was good to about 103dB.

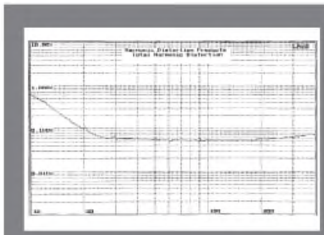


FIGURE 17: Parts Express 300-794 distortion—50W.

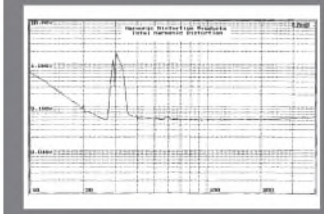


FIGURE 18: Parts Express 300-794 distortion—150W.

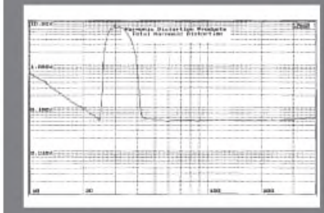


FIGURE 19: Parts Express 300-794 distortion—200W.

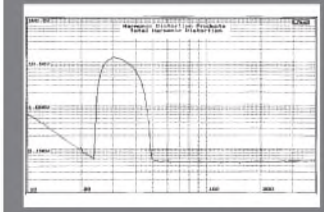


FIGURE 20: Parts Express 300-794 distortion—250W.

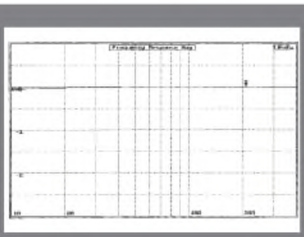


FIGURE 21: AudioSource Amp Three frequency response.

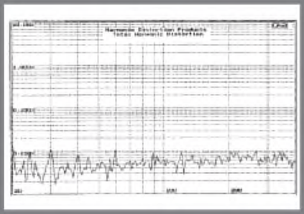


FIGURE 22: AudioSource Amp Three distortion—2W.

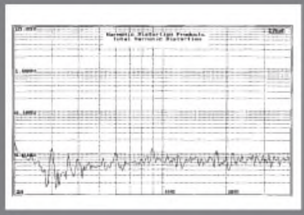


FIGURE 23: AudioSource Amp Three distortion—50W.

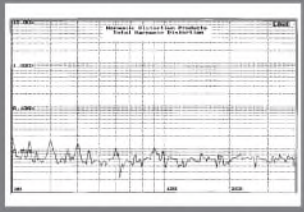


FIGURE 24: AudioSource Amp Three distortion—150W.

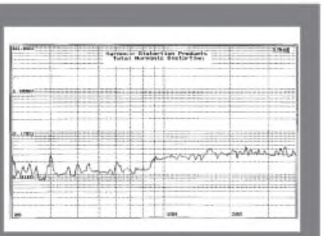


FIGURE 25: AudioSource Amp Three distortion—200W.

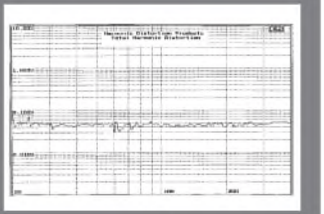


FIGURE 26: AudioSource Amp Three distortion—250W.

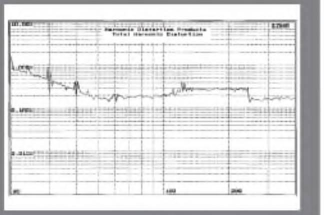


FIGURE 27: AudioSource Amp Three distortion—300W.

- "Jurassic Lunch." Almost no floor shake, but otherwise strong bass.

- *Brasileiro*. "Fanfarra's" opening percussion was good to 106dB. "What Is This" had good opening percussion to 106dB. The output on the low bass was good to around 105dB.

- *Ibuki*. The lead-in drum was full, but not hard. The main part of the cut was good until around 106dB.

4. AudioSource Amp Three

- *The Wall*. The bass was tight and clean. The kick drum sounded clean to around 105dB.

- "Jurassic Lunch." Some 12Hz foot stomps but not much floor shake.

- *Brasileiro*. "Fanfarra" had tight and detailed percussion in the intro to around 108dB. "What Is This" had very tight and detailed intro bass to around 107dB. The deep bass around two minutes in was clean and detailed to around 107dB, but not quite as deep as with the ACI woofer.

- *Ibuki*. The intro was very tight and detailed with the main section reaching levels in the range of 106dB while still clean.

- *Steppin' Out*. The piano and voice were very detailed, but a little light. The string bass was good, but a little on the fat side.

DECISIONS, DECISIONS, DECISIONS

What's the bottom line on these amps? Probably the most pertinent impression I can give is that the first piece I listened to with the first amp and woofer elicited the response: "This sounds much better than it has any right to." That impression was carried through the balance of the tests. The results were slightly different with the different amps and woofers. However, the overall impression of very good sound for not a lot of money survived all the listening tests.

I'm writing this conclusion right after coming home from a Gershwin concert at our local symphony orchestra. It was one of the recurring aural reality checks I perform to help me separate the wheat from the chaff. Did these amps produce the same kind of experience I had at the concert? You must be kidding me.

If anyone tells you that you can spend less than $1,000 and have an experience that is just like being at a live concert, button down your wallet. In fact, if anyone tells you that you can spend any amount of money and have the same experience as at a live concert, still button down your wallet. The simple fact is that sound is just part of the total experience. However, given the limitations of most listening rooms, it is amazing how good the sound can be today if you make the right choices.

A good subwoofer makes it much easier for any well-designed speaker to reach new heights of realism. The foundation that the subwoofer provides is essential to a sense of realism. Unloading the most demanding part of the volume-displacement equation from the main speakers allows for a cleaner, more defined sound.

INDIVIDUAL AMPS

The Apex Junior is probably the best value of the group. It doesn't have as much continuous power as some of the others, but in music tests that are transient in nature, the power supply was probably not stressed as much, helping to explain why it held its own in many of the tests. The response was fairly flat, leading to a balanced sound except for the very lowest notes.

The fact that it did not have a large boost in the bass allowed it to actually produce cleaner low bass than some of the others, although at relatively low levels. However, the lack of the boost made the balance a little light when used with woofers with an f_B above 40Hz. In my mind, that's not a problem as you can use external eq if needed.

What about the power level? Well, my preference in subwoofer design is to use, where possible, at least two drivers on opposite sides of the enclosure. That configuration, with cross enclosure bracing, can eliminate the majority of problem vibrations. In this case, I would simply use two Apex Juniors. That's right, if you build your sub this way you would have two—two—two amps in one. The price is certainly right and the amp has good performance.

The Marchand is a good example of a well-designed, broadband amp that would do well in almost any application. Words that come to mind when describing the construction include solid, heavy-duty, professional, and clean. This type of construction does not come cheaply, but it tends to pay off in consistent performance and durability. The performance was, in fact, very good, and this was the only amp that had a balanced input.

The flat response of this amp gives you the maximum flexibility to control the ultimate response of the sub by using your choice of external signal-level control. It costs more, but it also frees you from someone else's idea of what frequency contour should be used. If you prefer the performance usually associated with a stand-alone amp but with the convenience of a built-in, this is the one to look at.

The Parts Express SWS250 has lots of power and efficiency. The price was moderate. It did not meet its published power rating, but performed well in actual tests. The fixed boost at 30Hz worked to the advantage of these subs, which are typical of what you might expect in a home-built. However, on some material the low boost tended to make the sound a little slow.

This was a good amp, but its main claim to fame—high efficiency—does not seem to be a major factor when power levels of this magnitude are in play. Picking up efficiency in a multi-kilowatt amp that will be playing continuously in professional sound-reinforcement applications is very different than a home environment at lower power levels with occasional use.

The Parts Express 300-794 has lots going for it. The power output is high, the highest in this group. The price is right, build quality is good, and the distortion is low. Although I prefer to have frequency contouring controllable by the user, the fixed boost in this amp worked well in most situations with the subwoofers tested.

The caveats are a phase-control circuit that was tricky to use and a rather strange crossover contour. Both of these are more a problem with set-up and balancing, and I would not expect them to be a problem once things are settled.

How do I compare these to the separate amp and crossover such as the AudioSource and a custom-built crossover I used? For the majority of applications, I would say one of the sub amps on the market today would be a great choice for the novice or intermediate sub builder. Even if you are going to build a monster sub with multiple drivers, it might be cheaper to buy several of these amps rather than a multi-kilobuck, multi-kilowatt separate amp.

If ultimate control of frequency contouring and power are your goal, a separate amp and crossover may be your solution. Be aware though, that this type of power and control may not be the best value. Would I buy an integrated for my own designs? You bet! ❖

Manufacturer's Response:
I would like to thank Tom Perazella for including the Apex "Junior" in his review and for his favorable testing results. Apex Jr. is a Surplus Electronic company specializing in affordable electronics for both industry and do-it-yourselfers, offering both new and new surplus items.

For a limited time, Apex Jr. will offer a special price to readers of audioXpress. *The offer will end Friday, August 31^{st}, 2001. Mention Perazella's review of the Apex "Junior," and the price for one piece will be $85, two for $165. You can view more info and other items on my website at www.apexjr.com.*

Steve Slater
Owner

consumeR
PRODUCTS REVIEWED

Parasound HCA-1000A Power Amplifier

Reviewed by Charles Hansen and Duncan and Nancy MacArthur

PHOTO 1: **HCA-1000A front view.**

Parasound Products Inc., 950 Battery St., San Francisco, CA 94111, 415-397-7100, www.parasound.com. Suggested retail price: $650 US. Power consumption: 700W. Dimensions: 17¼″ W × 13¼″ D × 3½″ H (4 ⅛″ H with feet): net weight: 22 lbs. Limited two-year warranty.

The Parasound HCA-1000A is a two-channel high-current amplifier (HCA) with a rating of 125W per channel. The power amplifier is a product of Parasound's legendary designer, John Curl, whose designs have been featured in *TAA* and *AE*.[1,2] The HCA-1000A is certified by LucasFilm for use in Home THX® audio systems. The "A" suffix in the model number designates auto turn-on, a feature useful in home theaters and custom, multi-zone installations.

Photo 1 shows the front panel, which has the power switch and a number of LED indicators. Two LEDs are used for left and right current overload indicators. Just below them are the LEDs for AC line, standby, and normal power status indication.

The amplifier is extremely rugged, constructed of heavy gage steel. A two-rack space mounting adapter is available. The top cover engages a slot in the 0.187″-thick front panel over its full width. This produces a very tight cover fit, requiring only six screws to secure the cover to the chassis. The top cover features cooling slots above the power transformer and the transistor heatsinks, while the bottom cover is slotted below the heatsinks.

The rear panel (*Photo 2*) includes the IEC power receptacle with line fuse, a 12V DC trigger terminal block, four high-quality gold-plated Tiffany-style Teflon™-insulated RCA jacks, two 50k input level controls, a switch to select mono-bridged or stereo operation, and two pairs of high-quality gold-plated five-way speaker binding posts, which are on US 0.75″ spacings, so you can use dual banana plugs. Two of the phono jacks are used for inputs, and are hard-wired to the second pair so you can output-loop the inputs to another power amplifier for bi-amplification.

Photo 3 shows the amplifier with the cover removed. A large toroidal transformer sits front and center. The main PC board is machined to clear the transformer. Two large heatsinks occupy the left and right sides of the amplifier. Each heatsink has six output transistors, as well as additional driver transistors and thermal sensors. Four large filter caps and two power diode bridges are just behind the transformer.

A small PC board for the trigger circuit is located in the front-left corner. There are also small PC boards for the LEDs, the input jacks and gain controls, and the output Zobel networks. The relay and protection circuits on the main PC board are just in front of the input/output area.

The unit is furnished with a heavy custom-designed power cord. You can connect the power transformer primary for 115V or 230V mains by means of a line terminal block near the LEDs. A line fuse is located in a drawer in the IEC power receptacle. The third pin of the AC receptacle is connected to the chassis. There is adequate finger space under the unit to easily lift it.

PHOTO 2: **HCA-1000A rear view.**

TOPOLOGY

A schematic was not furnished with the unit. The 785VA toroidal power transformer has independent secondary windings for each channel, with independently rectified power supplies and a group of 10,000μF filter capacitors (40,000μF total). Three pairs of Sanken beta-matched 15A, 50MHz, 130W bipolar transistors are connected in high-bias Class A/AB complementary-symmetrical operation on each side. The circuitry is direct-coupled with no capacitors or inductors in the signal path, and AD711 op amps are used in a DC servo circuit to prevent any DC component in the output. For added safety, the speaker outputs have relay time-delay start-up and protection circuits.

The input stage uses hand-matched complementary JFETs. The input jacks are looped to the output jacks for connecting multiple amplifiers to a single source. These jacks appear to be two pairs of inputs, so you must be aware of their actual function when you make connections. Resistors are 1% metal-film and 5% metal-oxide types. High-quality film caps are used throughout, with 5% silver mica caps used in the input stages.

The 12V DC automatic turn-on circuitry allows you to use one master component—typically a control preamplifier—to activate the entire audio system. The power switch on the preamplifier sends a 12V DC trigger signal to activate every auto turn-on component attached to the system. Auto turn-on allows custom audio-video systems installers to place the amplifiers in out-of-the-way locations.

MEASUREMENTS

I operated the HCA-1000A with pink noise at 10W into 8Ω for one hour. After this run-in period, the chassis was only warm to the touch. The distortion was slightly higher in the left channel, so I present the test data here and summarize it in *Table 1*. There was a very low level of hiss and no hum with my ear against the speaker. Other than the time-delay relay, there was no noise during power-up or shutdown. While the HCA-1000A has input gain controls, I left them set at maximum during the testing.

The HCA-1000A does not invert polarity. Input impedance was 48k for both channels. The gain at 2.83V RMS output into 4Ω and 8Ω loads was 28.9dB and 29.0dB, respectively. The output impedance at 20Hz and 1kHz was a very low 0.06Ω, increasing slightly to 0.08Ω at 20kHz. DC offset at the speaker output was nonexistent.

The frequency response for the HCA-1000A was ruler flat from 5Hz to 50kHz at an output of 2.83V RMS into resistive loads of 4Ω and 8Ω and a complex load of 8Ω par-

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power output (RMS)	125W RMS, 8Ω × 2 ch 200W RMS, 4Ω × 2 ch 400W RMS, 8Ω bridged	148W, 8Ω, 1kHz; 130W, 8Ω, 20kHz; >200W, 4Ω, bridge 8Ω
Current capacity	45A peak per channel	
Dynamic headroom	>1.5dB	
Slew rate	>130V/µs	
Power bandwidth	5Hz–100kHz, +0/–3dB at 1W	5–100kHz, +0/–2dB, 1W
Total harmonic distortion	<0.03% at full power; <0.01% typical levels	0.01%, 125W, 8Ω 0.03%, 125W, 4Ω 0.0045%, 10W, 8Ω 1kHz
IMD—CCIF (19 + 20kHz)	<0.03%	0.012%, 12V p-p
MIM (9 + 10.05 + 20kHz)	N/S	0.008%, 12V p-p
TIM	Unmeasurable	
Interchannel crosstalk	>80dB at 1kHz >60dB at 20kHz	–90dB at 1kHz –68dB at 20kHz
Input sensitivity	1V for 28.28V (28.9dB) THX ref; 1.1V for full output	28.9dB
Input impedance	33kΩ	48kΩ
Signal to noise ratio	>116dB, full power >96dB THX ref level	
Power requirements	700W maximum	
Output impedance		0.06Ω 20Hz–1kHz 0.08Ω 20kHz
Damping factor	>800 at 20Hz	

PHOTO 3: HCA-1000A interior view.

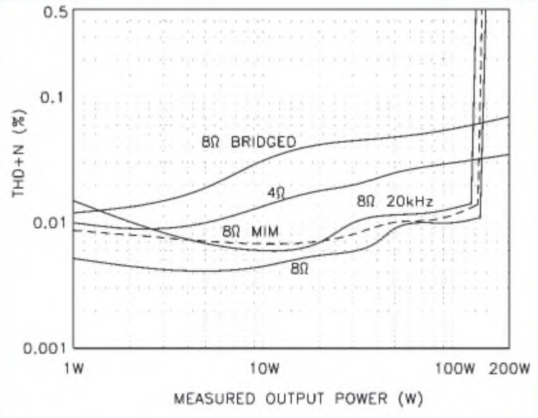


FIGURE 1: THD+N versus output.

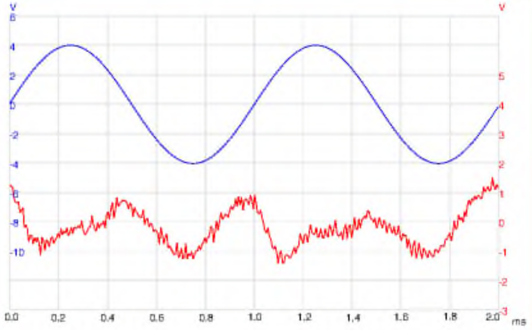


FIGURE 2: Residual distortion, 1kHz.

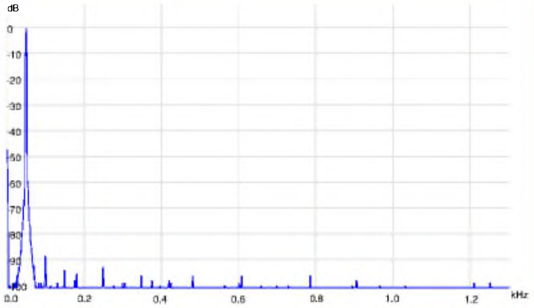


FIGURE 3: Spectrum of 50Hz sine wave.

■ Reviewed by Duncan and Nancy MacArthur

AESTHETICS AND OPERATION

The HCA1000A arrived in a strong cardboard box with custom-cut cardboard inserts to protect the amplifier during shipping. A major advantage of solid-state amplifiers over their tube brethren is their weight: the Parasound was easy to unpack and position.

The "fit and finish" of the HCA1000A is excellent and includes a number of nice touches, such as indicator LEDs in the grooves in the front panel. The Parasound is a stylish black box. Although black boxes are practical and easy to match with other black boxes, in the end they're still black boxes. If you're looking for a fashion statement, you may wish to look elsewhere.

On the plus side, the HCA1000A exemplifies the many positive features that a solid-state amplifier in this price range can include. Switching was flawless and (with the exception of a muted relay click) soundless. Everything worked well throughout our audition; the construction quality of this amplifier inspires confidence that it will continue to work for years to come.

The Parasound has various convenience features that could simplify setup of some systems, particularly in home-theater applications. One feature we're not used to seeing on power amplifiers is output RCA jacks. The output jacks (termed loop outputs by Parasound) are connected in parallel with the inputs to allow multiple amplifiers to be fed by a single long pair of shielded cables.

In addition, on the rear panel this amplifier has separate volume controls for each channel, a mono bridging switch, and a 12V DC trigger connection for turning power on and off remotely. These features would probably aid in integration of the HCA1000A into a video system, but they are unlikely to be used in audio applications.

IN A NUTSHELL

Overall, the Parasound HCA1000A is quite listenable. Its strong points are its bass response; clear, clean mids and highs; superb imaging; and awesome dynamics. Its major weaknesses are a slightly bright sound and lack of naturalness compared to similarly priced tube amplifiers.

DETAILED LISTENING RESULTS

All our auditioning was done using Genesis 400 speakers. These three-way floor-standing systems (not bi-amplified) are rated at 89dB/W efficiency but are 4W designs with a fairly difficult impedance curve. The source for all listening was a modified Philips DAC 960 with a variable output that was used to drive the HCA1000A directly. We compared the Parasound to a pair of VTL "Tiny Triode" tube amplifiers and a Manley Stingray tube integrated amplifier. For reference purposes, the VTL amplifiers would rate about six in all four categories in the ratings box.

The sound of the HCA1000A did not change noticeably during our month-long listening period. Either this amplifier does not require an extended break-in period or this sample had been broken in before it arrived here. Out of habit we turned on the Parasound for at least an hour before we did any serious listening.

As usual, we listened to the Hi-Fi News and Record Review disk III (track 2: Parry's "Jerusalem," track 4: Vivaldi's trumpet concerto, tracks 5 and 6: excerpts from Prokofiev's "Peter and the Wolf," track 7: Purcell's "Welcome, Welcome Glorious Morn," track 10: a Corkhill percussion piece, and track 14: Rio Napo RSS demo). We also played favorite tracks from a wide variety of musical genres ranging from jazz to classical.

The image the Parasound produced is exceptionally wide. The choral voices in "Jerusalem" extended smoothly across the stage. The Rio Napo demonstration piece gave an astonishing width of sound, extending well beyond the speakers. On some recordings, such as Blues on Bach by The Modern Jazz Quartet (Atlantic compact disk, 1652-2), this extreme width of sound produced an almost "surround sound" effect, even through a single pair of speakers.

Oddly, we disagreed as to how much image depth this amplifier offered. Duncan felt that the image depth, while excellent, fell slightly short of the astonishing standard set by the width. But on a number of tracks Nancy heard more depth, with the instruments apparently located as far back as eight or nine feet behind the plane of the speakers.

The Parasound pinpointed the location of each instrument precisely within the soundstage. The Vivaldi trumpet concerto from the test disk and "Misguided Angel" (Cowboy Junkies, The Trinity Sessions, BMG compact disk 8568-2-R) provided good demonstrations of this characteristic. Some amplifiers smear instruments and voices muddily across the soundstage, but not the Parasound.

Ambiance recovery was not one of the strengths of this amplifier. The space around the choir in "Jerusalem" and the acoustic space around the trumpets on the Vivaldi piece were not particularly extended or "spacious." This performance was acceptable and did not detract from enjoyment of the music but could have been better.

The Parasound produced an extremely strong and realistic bass. The bassoon on "Peter and the Wolf" sounded like a real bassoon being played in the listening room. The sound of the timpani was very deep and well defined. The bass response remained full and realistic even on large orchestral pieces such as Brahms Symphony #4 (The Royal Philharmonic, Fritz Reiner, Chesky compact disk CD-6).

The amplifier's midrange reproduction was very good. Each instrument in the "Peter and the Wolf" selection was well defined and easily separable from the other instruments. All the sounds on the Rio Napo track were reproduced realistically (if realistic is the right term for this recording). With our system, the upper midrange response sometimes tended towards brightness; examples included the trumpets in the Vivaldi, the winds in the Prokofiev, and the high soprano notes in the Purcell.

As illustrated in the Rio Napo demo, the Parasound's highs were extremely clean and clear. Again, in our system this resulted in unacceptable brightness on some recordings.

The HCA1000A reproduced transients quite realistically. The individual drumbeats in the Corkhill were crisp and well defined. This amplifier also handled the dynamic range of a full orchestra very well. The Parasound never sounded compressed, even when playing complex music at earsplitting levels.

This amplifier delivered a very clean and clear sound but ultimately failed to sound completely natural, at least with our system. In the recordings of "Peter and the Wolf," "Welcome, Welcome, Glorious Morn," and the Rio Napo demonstration, the individual instruments were clear and very well defined but did not sound as fully realistic as these recordings sometimes can. The massed strings and trumpets on the Vivaldi trumpet concerto were reproduced cleanly but again sounded unrealistic and slightly bright.

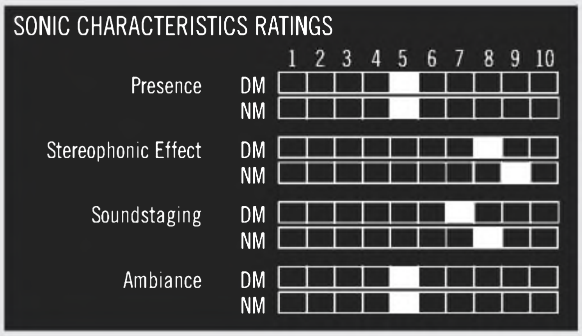
FINAL THOUGHTS

NM: Before considering this amplifier, ask yourself how your other components sound. If, like ours, the rest of your system is voiced to sound best with a tube amplifier, the Parasound may not be the amplifier for you. If, on the other hand, your other components produce a relatively mellow sound, the Parasound's superior imaging, depth, and dynamic range make it well worth a listen.

Also, keep in mind when reading this review that we both give natural sound very high priority. If other qualities, such as stereophonic effect, are especially important to you, you may draw different conclusions.

DM: I guess that I just like the sound of tube amplifiers. Although the HCA1000A has a number of exceptional audio features (I wish I could keep the imaging and the dynamic response) and it appears to be very well made, I just couldn't warm up (no pun intended) to its overall sound. A thorough listening session using well-known source material and your own speakers is indicated prior to purchase.

The audio and operational strengths of the HCA1000A seem directly aimed at the home-theater market. Although we didn't test it this way, this amplifier might be just the ticket for video listening. I wouldn't hesitate to recommend an audition if this is your intended application.



alleled with a 2μF cap. The response was down 1dB at 70kHz and only 2dB at 100kHz. Crosstalk performance was excellent at greater than −90dB below 1kHz, rising in a straight line to −68dB at 20kHz.

THD+N versus frequency with 2W into a 4Ω load did not exceed 0.027% from 20Hz to 20kHz. The complex 8Ω/2μF load produced similar results. It was even lower with 1W into 8Ω, not even rising to 0.01% until 12.5kHz. When I connected the HCA-1000A for bridged 8Ω 10W output, the highest distortion was only 0.03% at 20kHz. This is the best distortion versus frequency performance I have ever seen in a power amplifier.

Figure 1 shows THD+N versus output power (top to bottom at 10W) into 8Ω bridged load at 1kHz, 4Ω at 20Hz and 1kHz, 8Ω at 20kHz, and 8Ω at 20Hz and 1kHz. I engaged the test-set 80kHz lowpass filter to limit the out-of-band noise. Due to limitations in my load resistors, I could not achieve the maximum power output with the 4Ω or bridged 8Ω loads. I also plotted the 1kHz product of the multi-tone intermodulation (MIM) 9kHz + 10.05kHz + 20kHz test signal vs. output power (dashed line) into 8Ω. This gives a better indication of the HCA1000A's non-linear response, since it is a closer approximation to music than a sine wave. There was absolutely no strain right up to the point of maximum power.

With the HCA-1000A reproducing a combined 19kHz + 20kHz intermodulation distortion (IMD) signal at 12V p-p into 8Ω, the 1kHz IMD product was only 0.012%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz) resulted in an even lower 1kHz product of 0.008%.

Using a 1kHz signal, the HCA-1000A produced 148W into 8Ω at 1% THD+N. The 20kHz power was a bit lower at 130W into 8Ω at 1% THD+N. After this full-power testing, the amplifier was very hot to the touch, as you can imagine.

The distortion waveform for 10W into 8Ω at 1kHz is shown in *Fig. 2*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. This distortion residual signal shows mainly the third harmonic. The noise riding on the waveform is of no consequence, since the THD+N at this point is only 0.0045%.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 3*, from zero to 1.3kHz. The THD+N measures 0.0065%, and only the second harmonics poke above −90dB, without any visible artifacts of power-supply rectification.

The 2.5V p-p square wave into 8Ω at 40Hz showed negligible tilt. The 1kHz square wave was just about perfect. The leading edge of the 10kHz square wave showed one barely noticeable cycle of damped ringing.

All-in-all, the HCA-1000A showed outstanding performance in every area I measured.

Manufacturer Response:

Thank you for reviewing our HCA-1000A amplifier. We are very proud of our products here at Parasound, and I am glad to read that you have enjoyed your listening experience with the HCA-1000A. I only have one correction: In Charles Hansen's portion of the review, he mentions a "Limited two-year warranty." In fact, our amplifiers boast a ten-year parts, five-year labor warranty, a warranty rarely (if ever) seen in our industry. As I mentioned, we are very proud of our products, as you can tell by the warranty period.

Thank you again for your support, and if you have any questions or comments feel free to e-mail me. ❖

Ross Hulstein
Parasound Product Information Manager
ross@parasound.com

REFERENCES

1. J. Curl, "Classic Circuitry, JC-2," *TAA* 3/77, p. 48 (correction, *TAA* 2/78, p. 49).
2. J. Curl, "Classic Circuitry, JC-3," *TAA* 2/81, p. 56; *AE* 5/97, p. 53.

Easy Driver Flush-Mounting

Here are some helpful tips for flush-mounting odd-shaped drivers. **By Rodney Buike**

A perfect fit!

We all know the importance of flush-mounting drivers in our DIY speakers, right? Well, a while ago I started on a Focal-based home-theater speaker system. I was all ready to begin until I got to the drivers—a round woofer with square sides and square tweeter with rounded corners. How was I going to flush-mount these?

I scratched my head for a while and came up with this method. It is quick, easy, and makes perfect-fit flush mounts for drivers of any shape or size. I am by no means a master cabinet maker, so if you have any comments or suggestions, I—and other readers—would love to hear them.

SAFETY FIRST

Always wear safety glasses and follow the tool manufacturer's instructions. Most important, wear hearing protection! No use building speakers if you are going to be deaf!

First, cut five pieces of the ½″ MDF to a size approximately 2″ larger, on all sides, than the driver to be flush-mounted (*Photo 1*). This should give plenty of room for your fingers, which you will want to keep the length they are. Now mark the center of the MDF boards and drill a pilot hole through two pieces. Here is where a circle jig comes in handy. This indispensable tool for the DIY speaker builder yields perfect circles every time.

If you are countersinking tweeters, I recommend removing their faceplates to prevent tweeter damage. Cut a hole in one of the pieces of wood to the outside diameter of the surround of the driver (*Photo 2*). Do not throw this piece away yet!

Next, mount the driver inverted to this piece of wood using screws and two washers per screw to space the woofer off the wood (*Photo 3*). Here is where another "make it yourself tool" comes in handy—a router table. Mine is an old worktable. I cut a 2″ hole close to the end of it and made a bracket to hold my plunge router. Combined with a flush trim bit, this works wonders for cutting many identically shaped pieces. Router off all the excess wood, leaving a piece the exact shape you desire.

Remove the driver, blow off the dust,

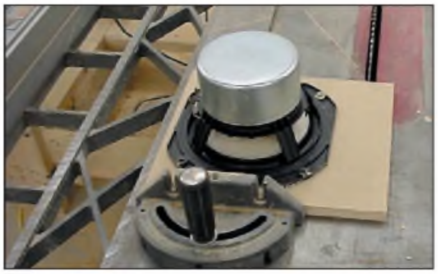


PHOTO 1: Cutting wood down to size.

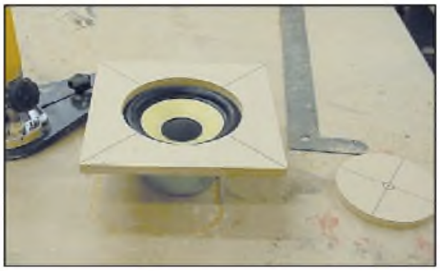


PHOTO 2: Cut the hole so the driver fits.

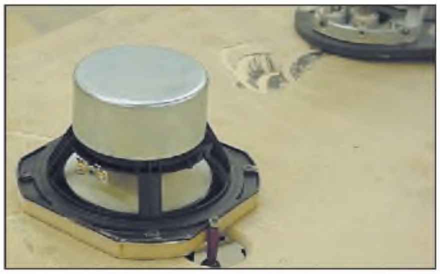


PHOTO 3: Trim excess with router table.

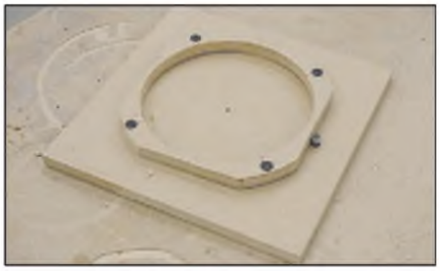


PHOTO 4: Attach to new piece and trim with router table.

and put it away. Using the pilot holes as guides, attach the circular piece you cut out of the first piece to the second piece of MDF. Cut a strip of the gasket tape into nine 1″ long pieces. Stick them three high at equal distances around the circular piece.

TEMPLATE

You can now mount the shaped piece together with a fresh piece and remove the center piece. Drill a ⅛″ hole through a new piece of MDF on the very edge of the shaped piece (*Photo 4*). Next, carefully trim the shape of the piece into the new MDF, leaving a hole in the MDF exactly ⅛″ larger than the driver (*Photo 5*). This is the pattern for your template so it must be *perfect*! I cannot stress this enough—better to do it right now than to mess up a baffle! Square any corners with a chisel if necessary (*Photo 6*).

Now that you have a perfect pattern, it is time to shrink it down to size! Short of going to your local electronics supply store and buying the parts necessary to build a DIY MDF shrinker, you'll need to use a router.

Attach the pattern to a fresh piece of MDF. Using a plunge router with a ¼″ straight cutting bit and a ⅛″ bushing guide, cut out the middle section. This will leave the original pattern and the new one, which is ⅛″ smaller than the original. Repeat this step to produce your template, which should be ⅛″ larger than the driver all the way around.

ABOUT THE AUTHOR

Rodney Buike is employed as an auto sound and security installer in Winnipeg, Manitoba. He runs a small DIY supply company on the Internet at www.hifiaudiolabs.com servicing the Canadian DIYer. He is currently enrolled in an electronics technician course.

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BKTE2 ....$44.95

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by Robert Harley

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BKAP3 ....$19.95

PHOTO 5: Here is the first pattern ⅜″ larger than driver. After two more repetitions, it will be ⅛″ larger.

PHOTO 6: Square corners if necessary.

easy. Cut the hole in the baffle for the driver and mount it. Using the ⅛″ foam tape, line the inside of the template. This acts as a spacer to guide the template so the driver is exactly centered. Attach the template using air nails or wood clamps or two-sided foam tape, and remove the driver and foam tape.

PHOTO 7: After setting router to depth, remove all excess within pattern.

Using the plunge router with ⅛″ bushing and ¼″ straight bit, set the depth you want the driver countersunk, remembering to account for veneer and gasket thickness. You can now remove all the excess wood inside the template (*Photo 7*). Remove the template and square out any corners if necessary, and you have a perfect countersunk hole.

TOOLS AND PARTS *Photo 8*

Handful of ¾″ screws
8 1⁄16″ thick washers
⅛″ thick foam gasket tape
½″ MDF—size dependent on driver size
Plunge router
Router table (if you have one)
Circle jig (again if you have)
Jig saw or table saw
¼″ straight cut router bit
⅜″ flush trim router bit
Router base attachment with ⅛″ guide
Drill

It is that easy! I hope that this will aid readers in the construction of speaker cabinets and add to their enjoyment of speaker building! ❖

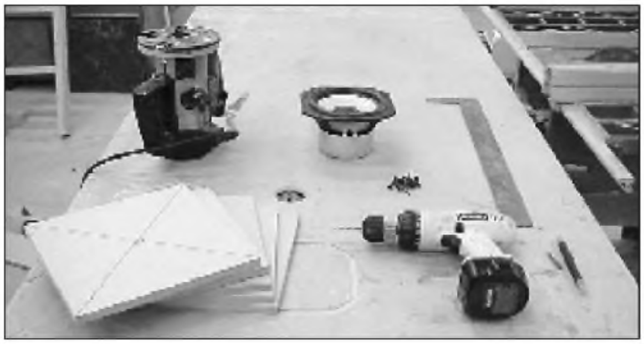

PHOTO 8: The tools required.

Linear AC Measurements Using a "Diode"

Once again, synchronous rectification solves a pesky problem caused by diodes. **By Michael Kornacker**

A couple of years ago I built an isolation transformer with a variable autotransformer, or Variac®, so that I could work safely on hot-chassis equipment. I also use it on my regular projects because the Variac allows me to power them up slowly so I can check voltages and find shorts before something blows up. I hate being surprised by a puff of smoke or a burned finger.

ABOUT THE AUTHOR

Mike Kornacker just earned his BSEET from Roosevelt University and is a brand new 44-year-old engineer at Northrop Grumman, a major defense contractor. Prior to that, he was an electronic technician there for 20 years. Kornacker is still in school working on a Master's degree in computer science. He got started in electronics because of his dad, who was also an electrical engineer at the same company. Kornacker also has two beautiful daughters.

HUGE TRANSFORMERS

The idea for this project came from *Electronics Now* magazine (Dec. 1997, p. 18), which suggested using two identical high-voltage transformers hooked up secondary-to-secondary. I found two huge surplus 380V 500VA transformers. At 120V they could supply 4A of current, which was perfect.

The Variac I used was a 3A model connected to the primary output of the second transformer. I also used two DC milliammeters to measure the Variac output voltage and the current flow of any device I plugged into it. All this I installed in a large aluminum box along with a small fan, an on-off switch, a fuse, an outlet socket, and other miscellaneous requirements.

The metering circuit (*Fig. 1*) gave

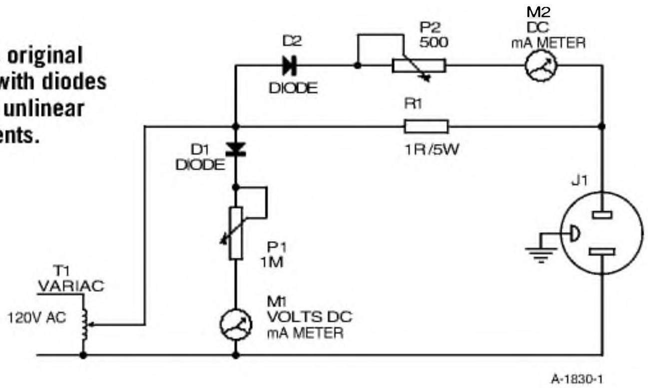

FIGURE 1: The original meter circuit with diodes that produced unlinear meter movements.

me some problems. Since the meters were DC, I used diodes to half-wave rectify the voltages so they would work properly. The problem is that PN junction diodes such as these are nonlinear devices. The diode resistance is not constant, but varies with the amount of current passing through it. If I doubled the voltage of the Variac or the load current, the meters would not show a corresponding linear increase. I could easily have re-marked the meter faces to reflect this, but I am an idealist.

I then tried using Schottky diodes because they work on a different electrical principle than the PN junction types. These worked better in the circuit, but still were not good enough. Germaniums that I tried were also unlinear. What I desired was the perfect diode, a linear device that would let me

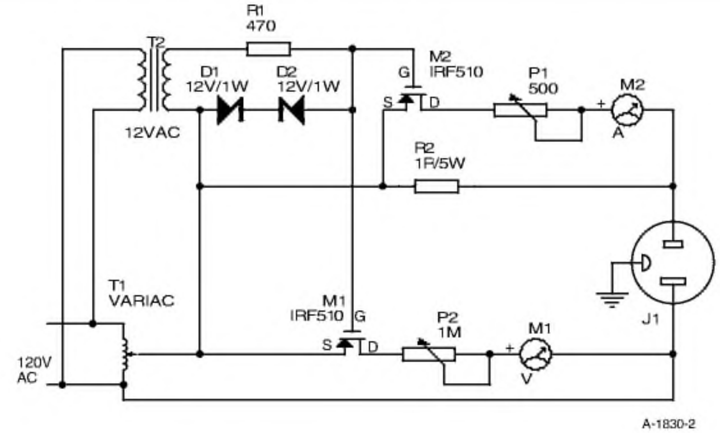

FIGURE 2: Using a synchronous rectifier circuit makes a perfect diode and linear meter movements.

accurately measure AC with a DC meter, no matter what the level.

One way to measure AC current only is to use a current transformer with the primary winding in series with the main line and the secondary winding connected to a diode rectifier and DC meter. This method works by converting the current to a proportional voltage and then stepping it up so the diode works in its linear region. A toroid transformer would be perfect for this, but would require me to wind my own, so I chose not to go that route. If you would like more information, see Eric Lowdon's work[1], which contains an excellent chapter on current transformers, with all the details.

SYNCHRONOUS RECTIFIER

A simpler way to solve my dilemma of measuring AC voltage and current with DC meters is with a synchronous rectifier using a power MOSFET transistor and a 12V transformer (see my article, "Try Synchronous Rectification," *AE* 5/00, p. 26). It makes the perfect non-diode diode rectifier. One of the beauties of a MOSFET is that it does not have a PN junction voltage drop like a bipolar junction transistor. Its voltage drop acts more like that across a standard resistor—an IR voltage drop. Actually, it is exactly like a resistor, albeit a silicon one, and resistors, we know, are for the most part linear devices. It is this characteristic that I wished to take advantage of.

Figure 2 shows the synchronous rectifier circuit in a half-wave configuration. The basic circuit uses a small 12.6V AC 300mA single secondary transformer and two IRF510 n-channel MOSFETs. The peak unloaded voltage of this auxiliary transformer was about 23V, information that is important because the MOSFET gate with respect to source bias, for saturation, must be greater than 10V peak but less than 20V peak to prevent transistor destruction.

In other words, when the source goes positive, the gate must go more positive by the aforementioned amount, but not any more than that. Since the gate voltage was over the allowable, I incorporated as a limiter two 12V, 1W zener diodes (1N4742A) connected in a cathode-to-cathode connection and a 470Ω, ½W dropping resistor.

The secondary of this transformer, you will notice, is connected to the wiper of the Variac. The phasing is such that the gates of the transistors are always switching at the line rate, regardless of where the wiper is set. If the wiper is just off 0V, for example, the transistors are ready to rectify the voltage for each of the meters.

DEDICATED TRANSISTORS

Each meter has its own transistor. I tried a single transistor for both, but for some reason I have not yet fathomed the current meter would pin the needle when the Variac was set to a certain point. Apparently there was an unexpected voltage-divider-path problem between the 1Ω shunt resistor and the ½Ω MOSFET on-resistance that fortunately was cured by using dedicated transistors for each meter.

The zener-diode circuit across the auxiliary transformer's secondary also

helps protect the gates from power-line high-voltage spikes and transients that could destroy them. I also installed a 200V metal oxide varistor (MOV) across the primary of the transformer for some added safety. All the parts used, the 12V transformer, MOSFETs, zeners, resistor, and MOV can be found at your local parts store. Finally, no heatsinking is required for the transistors, since they do not draw any appreciable current.

Now my problem is solved. When I increase the Variac voltage or when the load draws more current, the meters read in a linear and predictable fashion. Using synchronous rectification in this manner solved an annoying measurement problem. It allowed me to use DC meters to measure AC without the bad side effects caused by the nonlinear nature of diodes. ❖

REFERENCES

1. Lowdon, Eric, *Practical Transformer Design Handbook*, Tab Books, 2nd Edition, Blue Ridge Summit, Pa, 1989, ch. 13.

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Secondary Inductance	719 H	1600 H
Effective Sec. Leakage Induct.	15 mH	22 mH
Primary DC Resistance	0.1 ohms	0.1 ohms
Secondary DC Resistance	190 ohms	273 ohms
Eff. Sec. Internal Capacitance	700 pF	800 pF
-3dB Power Bandwidth, Start	35.35 Hz	35.35 Hz
w/ Rep in-series	1.051 Hz	0.515 Hz
Pri. Imped. W/Rep. 10Hz	18.26 ohms	18.10 ohms
Electrostatic Speaker Cap	1 nF	1 nF
Resonance Freq., 2nd order	31.52 kHz	25.29 kHz
Q factor	0.601	0.642
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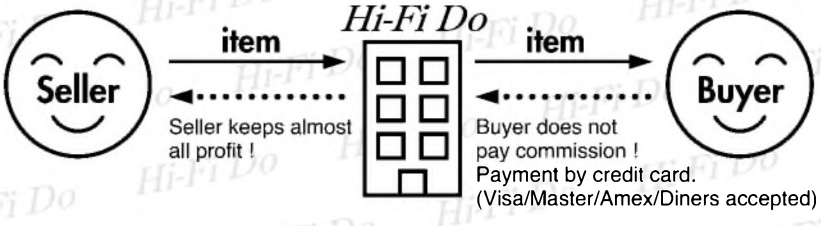

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Bayonet Sockets

To accommodate different tube configurations, you may need to modify your supply of sockets. This crafty veteran shows you how.

By Chris Wolf

One of the real pleasures of producing your own audio gear is setting it up the way you choose. Too many over-the-counter sound systems look great from the outside, but are crammed full of cheap hardware and plastic. It makes you wonder what other sacrifices they made to lower the price of the units.

Custom-built equipment often includes a variety of parts that may be overkill for their actual application, but so what? Take bayonet sockets, for example. These were originally used to keep tubes in close contact with their mounts, and to make them stay put. Transmitting cabinets or panel mounting also come to mind as practical uses for bayonet sockets. Another application is with a tube that's going to spend a bunch of time bumping and crunching around in a cockpit or Jeep.

There's no other way of mounting a 211, but bayonet sockets are usually overkill for typical 811, 45, and 300B type tubes. Like I said, some folks dig overkill parts. When you're building your own system, why not use them?

OVERKILL BLUES

Well, I had the overkill socket blues the other day. Let me preface this sad story by confirming the fact that not all 300Bs out there are pretty. I have a couple with downright scuffed logos, which would look just nifty hidden in a nice bayonet socket.

My collection of bayonet sockets work just fine when I am using them on virtually any 4D-based triode in the Known Universe. Unfortunately, West-

PHOTO 1: WE 300 B-style socket after finishing and plating.

ern Electric threw us a curve when they modified the 300A by rotating the pin 45°, and thus created the 300B. Bernard Mager's book[1] says this was to accommodate the 300B in old 205D-equipped amplifiers. These are great triodes—but it's impossible to find matching bayonet sockets.

"Not so," said the hired help at a couple of tube houses I contacted. "The 300Bs that we have fit the old E.F. Johnsons and the new Chinese versions just fine."

I wondered whether they were using an original 300B, but my buddy Alex at Darcell Electronics straightened me out by pulling out a set of new-production WE 300Bs for inspection. "The Chinese ones work OK, but man, I see what you mean," said Alex, "the real ones don't fit."

Well folks, here's the skinny. All those thousands of repop 300Bs floating around out there using 811-style bases are actually 300As, the way I see it. The 300B base has a different bayonet pin location, and there is no possible method to use the standard E.F. Johnson bayonet socket on the original. I'm sure Western Electric made a few, but call me (in about ten years) after you find a set.

VENERABLE JOHNSONS

It is said that Henry Ford's Model T could be fixed by anyone with just about anything at hand. The same is true for the venerable Johnson bayonet socket. Since the base pinout is the same on the 300B and others in the 4D family, all you need to make a usable bayonet sleeve is a bit of 0.0625″ wall steel pipe (1½″ outside diameter, 1⅛″ inside diameter), some tools, a work bench, and—if you're really picky like I am—a vertical mill or drill press milling bed (*Photos 1-4*).

Start out by hacksawing the pipe into rough dimensions. Use some masking tape to transfer the dimensions of the original bayonet collar onto the steel pipe. Leave enough metal to produce the mounting tabs later. A single-cut file will take care of the burrs.

Jig a supporting piece into the pipe and bend the tabs. I used a welding torch to soften the tabs during the bend, but if you're a real animal, tap them over cold. Form the tabs with the

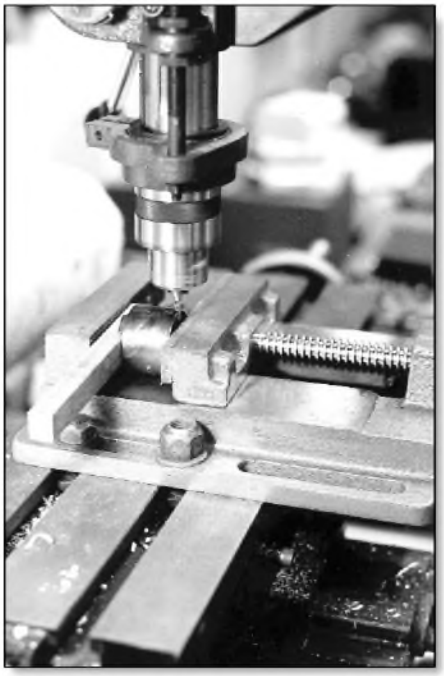

PHOTO 2: Initial cut. Use vertical milling bit to make vertical guide slot and right degree bend.

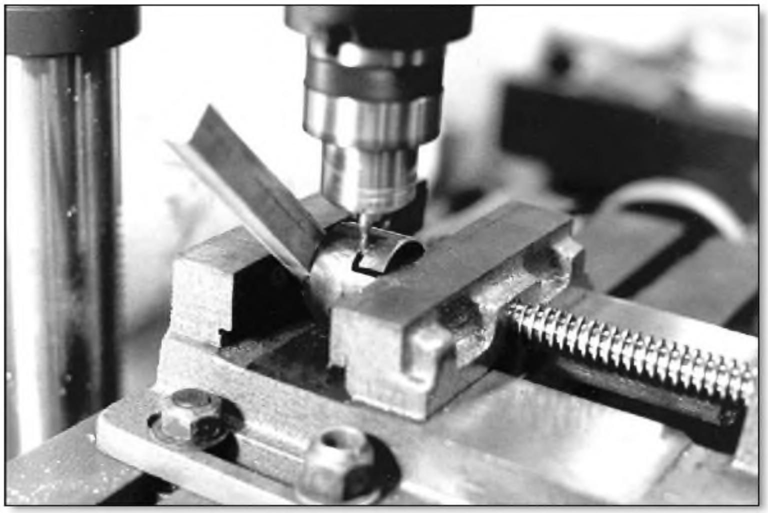

PHOTO 3: Re-adjust sleeve in table vise and mill remaining horizontal section of slot.

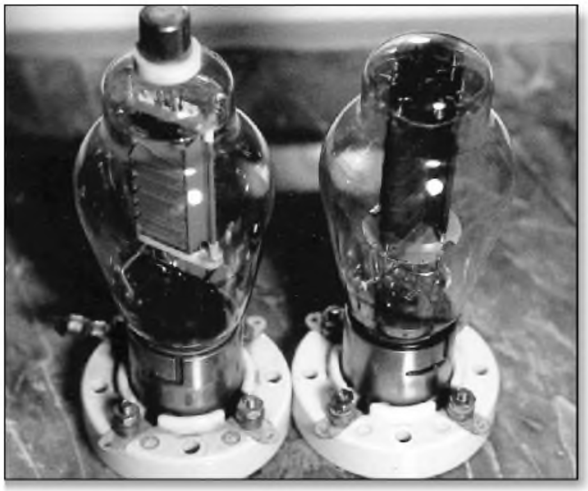

PHOTO 4: Rear view of modified sleeve readily shows difference in pin locations. At left, 812A triode; at right, 300B triode with rotated bayonet pin.

file to fit the original ceramic base and you're in business.

Trial-fit the new bayonet sleeve and file it for a nice, snug fit. Mark the mounting hole location from the underside, drill, and thread the sleeves with a 4-40 tap. The fun part is setting up the milling table to cut the bayonet slot.

Clamp the piece securely and squarely in the vise. Go out and spend six dollars to purchase a 5⁄32″ end mill to cut the slot. Transfer the dimensions from the original sleeve, only rotate the pin location 45° for the proper engagement of the real 300B. Slowly cut the slot with lots of lubricant and then deburr it.

DRESSED TO KILL

I always try to dress up my pieces so they look the part. Carefully sand the inside and outside rims of the sleeve. I hate cutting a groove onto a tube base with a sharp bayonet sleeve, so if you have a rotating sanding drum attachment for your hand drill, a quick clean-up is in order.

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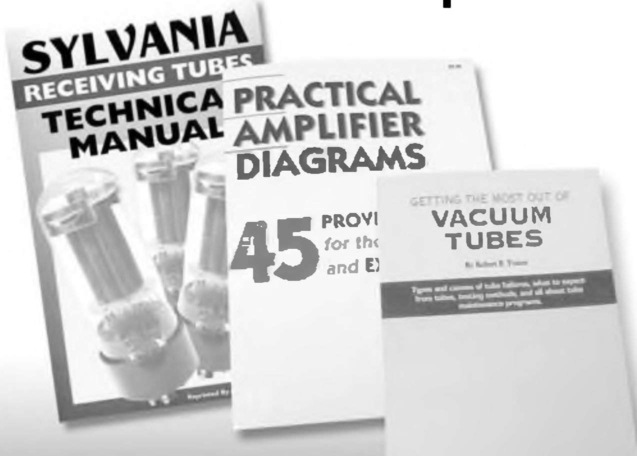

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I prefer to cut a slight taper or bevel at the inside top of any bayonet socket to avoid any potential marring of the tube. File all cut edges true, and sand them smooth with emery paper. Just to bother my friends, I punched W.E.CO. into the sleeves prior to applying 24K gold plating.

Sonic benefits of bayonet sockets over ceramic plate types are nil, but I'm sure some readers will disagree. It is sure nice having bayonet sockets for my beat-up 300Bs, however. ❖

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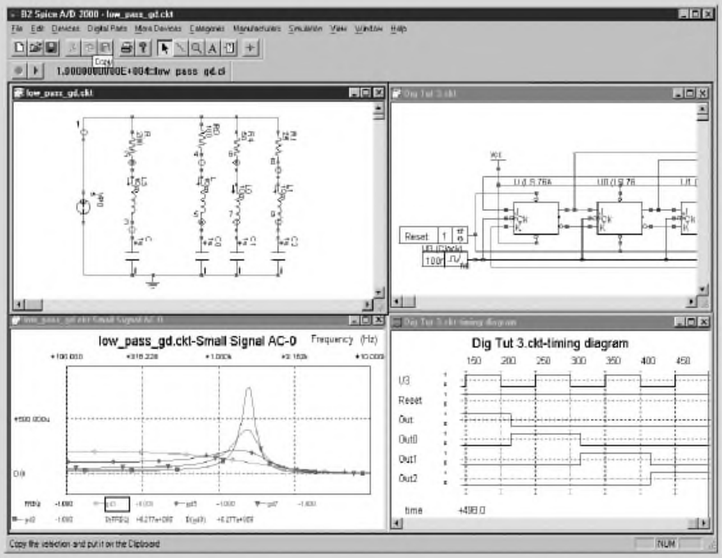

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PROS AND CONS

audioXpress looks good, and is a logical successor to the prior separate magazines. However, it does appear that tubes and speakers have the edge in content. I'd like to see more solid-state analog and digital articles. Solid-state amps, preamps, and DACs are my main interests now.

As for advertising, I enjoy appropriate advertising. We all need sources for the components that go into our projects, and it's very useful to find them through the magazine. Locating sources for parts is without a doubt one of the most challenging aspects of this hobby.

Expanded coverage of sources, perhaps including a monthly supplier or advertiser profile might be interesting. Kits are also of interest if they are unusual and/or of high quality. My current favorites are the digital kits from Assemblage. Audio test equipment kits are also a high priority for me, including PC-based oscilloscopes, spectrum analyzers, and so on.

Jeff Chan
jeff@Supranet.net

Allow me to congratulate you on the new format, incorporating the three magazines into one. I personally prefer this new format, especially the increased frequency. I think, and hope, you have a winner.

Justus Verhagen
Oxford, UK

What a disappointment! The speaker builder part is most definitely only about 20% and far less than *Speaker Builder* was. You can be assured that if there are no changes for the rest of the year, I won't be renewing. Then to raise the price of this new beast is unconscionable!

Guess you could say I am not impressed or amused. My worst fears were realized in your first issue.

I guess you still don't realize that the "glass" people, the "silicon" people, and the "speaker" people don't have enough in common to bring them together in one magazine. I will not waste my time reading the "glass" articles nor "silicon" articles. I have absolutely no interest in them.

A failed experiment. Whoever thought of this concept should be shot and horsewhipped every morning before breakfast at 6:30 AM (little bit of humor)!! What were you thinking of? Just because you had the three magazines doing fairly well doesn't mean you can combine them and still be successful; it meant the individual magazines were successful in their own space.

Suggestion: Go back to the way you were or go to an e-zine with only speaker articles and ads. I'll access it over the internet and print it myself. Bring down the cost as well.

Clarence A. Zacharias CBCP
Winnipeg, MB

I looked at the first issue of *audioXpress* at the newsstand. Frankly, I was disappointed. The issue is very heavily biased towards tubes, and, as a solid-state person, there was little in the issue for me. I had planned on subscribing to the new magazine, but will wait and see whether the coverage becomes more balanced between tubes and solid-state.

Claude Dickson
cdickson@mhfa.com

I was impressed by Perazella's system ("On Angel's Wings," *aX* Jan. '01). I was considerably less impressed with the new magazine. It appears to be less "*audioXpress*, incorporating *Speaker Builder*, *Audio Electronics*, and *Glass Audio*" than "The new, expanded *Glass Audio*, incorporating *Speaker Builder* and *Audio Electronics*."

I occasionally bought all three of the precursor magazines, but *Glass Audio* much less often than the other two. The "new" magazine is mostly *Glass Audio* topics with one (count them, one!) speaker-building project and no solid-state gear in sight—even in the ads! To me, trying to save your publishing business by combining three money-losing magazines into one is fruitless if the resulting magazine primarily targets only one of the core audiences of the previous three.

Since I've still got a year left on my original *Speaker Builder* subscription, I'll give them the time to get it right, but for the time being, I'm really disappointed.

Bob Stout
rbs@snippets.org

Ed Dell responds:

Mr. Stout might want to look again more carefully at the contents of aX issue 1, 2001. There are three speaker articles and, unless I am missing something, Norm Thagard's preamp is solid state, and Chuck Hansen's feature reports on some very solid-state chips. As too often with internet comment, the information is usually pure hearsay, distorted, inaccurate, and bad tempered. Not one of the three periodicals we published was losing money.-E.T.D.

Thank you for an exciting new magazine. Continued success to you and your staff. I do have one minor complaint. I was looking forward to reading the second installment of "On Angel's Wings" by Tom Perazella (*aX* Jan '01), but after reading it, I was somewhat disappointed. In future DIY projects, could you provide an e-mail address for the author, a materials list, and possible sources of supply? Also, information for obtaining blueprints for odd-sized shapes. Thank you.

Michael Anderson
Mikehound@webtv.net

Thomas Perazella responds:

Personal contact information on the authors is usually not included to ensure that they do not become overwhelmed with requests for information. Inquiries are forwarded to them by the magazine for response, as in this case with your e-mail. Many of the authors write as part of their hobby and have day jobs. This limits their time available for personal responses to readers. Even when I wish to contact another author, I go through audioXpress, *since it provides a buffer for the authors and keeps the magazine informed on things that could lead to other articles.*

The article did provide a list of supplies with costs (on page 69 of Part 2). Web addresses were given for AudioXstream on page 42 and Bohlender Graebener on page 46 in Part 1, although I have learned that AudioXstream is no longer distributing the drivers to the DIY market. You should contact BG to find a source of the drivers.

I did not provide sources of supply for the wood and other construction materials, as you can usually purchase them locally. The only unusual material is Kerfkore, which I purchased from Parkwood Chicago, Inc., 2100 South Foster Ave., Wheeling, IL 60090, 847-577-8001.

However, unless you are in the Chicago area, I suggest trying to purchase it locally, since it is heavy and the shipping costs would probably be prohibitive. This material is made by Interior Products, (800) 637-3539, www.interiorproducts.com.

The latex-backed carpet is available from Parts Express at www.parts-express.com, and the terminal binding posts from Vampire at www.vampirewire.com.

There are no blueprints for these baffles. They were never designed to be production pieces, or even the basis for a kit, but rather as a custom exercise for one pair. The drawings I used were literally pencil drawn on a large sheet of paper. This is a common practice with many home-design projects. In this case, the DIY extends all the way to doing your own drawings.

As mentioned in the article, inspiration for this design came from the work that Rudi Blondia and John Whittaker had done. However, in good DIY fashion, I designed the configuration myself.

I hope this information helps you. This project was certainly not for a first-time builder. I don't know your level, but if you do produce a pair of your own baffles for this fine driver, you will have something that will provide you with much satisfaction and listening pleasure.

Congratulations on an outstanding publication. It is really a captivating magazine, and the wide mix of articles riveted my attention from cover-to-cover. The format is strikingly similar to *Audio Anthology* with the addition of Glass Shards, New Chips on the Block, Audio Aids, Book Reviews, Xpress Mail, and Consumers Report. The publication is of great value for the reader, and I'm sure it will develop a wide following.

I especially enjoyed Norman Crowhurst's article ("Audio Classroom," p. 86). I'm sure he would have thoroughly appreciated your new endeavor. I regard him as the greatest teacher/writer ever. His main interest was conveying knowledge, and I have never detected one word of his masterpieces dedicated to "ego," only the dissemination of knowledge. In this, your "love" and "passion" for dissemination of knowledge through your publication also conveys this same dedication. I congratulate you on the outstanding formatting of your magazine; it will indeed get increased circulation.

Joseph Norwood Still
Bel Air, MD

I've just scanned the mag, but I'm not as disappointed as some of the others whose comments have appeared on the 'net. Some of this may be because I seem to be in the minority here, in being interested in tube amps (I have the luxury of two separate systems, one vintage tube, and one modern solid-state). And some may be because I have found very little of interest in any of Ed Dell's magazines recently (and I was subscribing to all 3). There may be little new in the tube audio world these days, but there wasn't much more new in *SB* or *AE* the last couple of years.

Certainly nothing nearly as important as Bill Waslo's original IMP series, Borbely's work on solid-state amplifiers, or the POOGE articles on modifying CD players. Perazella's article on his RD-75s was more useful to me than anything AAP published in all of 2000. That's enough to make the first issue of *aX* a success.

Paul Zyxtan
zyxtan@yahoo.com

I have been a subscriber to at least one or several of your magazines for many years, since, I believe, the '70s. A few weeks ago this new magazine showed up in my mail that I had never seen before. I put it aside for a quiet moment to study it at leisure.

I would like to congratulate you on this first issue of *audioXpress*. I sure hope that you can maintain the caliber and quality of this first issue. I believe it is just outstanding. I like the presentation, the style, and the content. There is nothing better on the market that I am aware of that covers the subject matter. Keep up the good work.

I hope that my subscription is fully intact to ensure that I will receive future issues without interruption.

Philip Hauser
Montclair, N.J.

A40 SUBS

After 22 years, it appears that quite a few DIY enthusiasts are still eyeing the A40 Class A amplifier design, but are discovering that not a single transistor from this design is still available. The following information is for those scouting out substitute parts.

The Lambda output devices are no longer available, and they weren't that common to begin with. Lambda appears to still be in business, but not in the business of power transistors. Common, everyday complementary power Darlington transistors will substitute for these. They need to be rated at 75W or higher, 5A or higher, and 80V or higher. Good examples of these are the TIP142 (NPN) and TIP147 (PNP), which are currently available from Digi-Key.

Other fine examples, although harder to get, are the Motorola MJ11012, MJ11014, and MJ11016 (NPN) and MJ11013 and MJ11015 (PNP). You can replace the remaining bipolar transistors, MPSL01 and MPSL51, with ordinary TO-92 devices rated at voltages of 100V or more and currents of 25mA or more. For best results, the beta (current gain) of the devices should be 100 or greater. Q3, Q4, and Q5 should be fitted with heatsinks.

It is possible that the bias might be at a slightly different value with these substitutions; slightly altering the values of R11 or R12 will allow some adjustment. A slight increase of R11 or decrease of R12 will increase the bias, and vice versa.

The JFET in the circuit, Q11, is used as a constant-current source to bias a few milliamps through D1 and D2, which provides a clean voltage reference for the constant-current sources Q3 and Q4. First, note that the source pin of the JFET is attached to R9, and there has been some confusion about that, with some different pinouts on the 2N5248.

You can replace this part with a convention N channel JFET that has been tested to act as a current source in the circuit with R9. The value of R9 has been chosen so that the JFET passes about 3mA or so (not critical). You can also use a monolithic constant-current source if you can find one, rated at about this current and 40V.

If you can't find any JFET for the current source, or are impatient while waiting for one to arrive, you can simply use a resistor as a poor man's current source by putting a 10kΩ resistor where the JFET goes, connecting between R9 and ground. This will work, but is noisier than a constant-current source. You can improve this with an electrolytic capacitor (say 100µF @ 50V) between the negative rail and the junction between R9 and the 10kΩ resistor, with the positive pin of the capacitor between R9 and the 10k resistor.

So that's it. For those who keep wishing to know what the value of C4 is, read the text. C4 is an optional capacitor in the event of high-frequency instability, and usually is 20 to 100pF.

Nelson Pass
Nelson@passlabs.com

POWER-SUPPLY SOLUTIONS

I really appreciate the level of detail Gary Galo went into when exploring the insides of the NAD T550 box (*aX* Feb. '01, p. 49). You would think that it would be fairly simple to retrofit a more robust regulated power supply for the analog supply. My question is whether or not you could safely raise the supply voltage (to, say, ±12 or 15V).

Dave Dlugos
Planet10@pinc.com

Gary Galo responds:

Thanks for your kind words on my NAD T550 review. If you wish to improve the analog power supplies, there are a couple of possibilities. Raising the supply rail voltages is really quite unnecessary.

The 0dB analog output level of the T550 is 2V RMS. Raising the rail voltages will not increase headroom or provide any other benefits. On the other hand, doing this shouldn't cause any problems either.

Here are two relatively simple ways to fix the power supply:

1. Install jumpers across the series inductors, as I did to verify the regulator dropout situation. The ±8V regulators will now regulate, and the rejection of noise from the switching supply will be about the same as it was before.

2. Replace the 7808 and 7908 regulators with 7806 and 7906 types, which will give ±6V rails. The regulators should now regulate, even with the inductors left in the circuit. Now, you will get the noise rejection benefits of both the series inductors and the regulators' own line rejection. Rails of ±6V will still be more than sufficient to allow the op amps to deliver the required 2V RMS analog output. Be extremely careful not to damage the plated-through holes on the PC board when you change the regulators.

Any other solutions will invariably be more complex. If you intend to build a super supply, such as Walt Jung's Improved Regulator (AE 4/00), I also recommend building a real raw supply, rather than simply using the ±12V from the noisy switching power supply. Bear in mind that any mods will void your warranty and, without a service manual, you may find it difficult to troubleshoot if you run into problems.

PHONO PREAMP TWEAKS

I was fascinated by Norman Thagard's two-part article (*aX* 1/01 and 2/01), as I was trying to work out a differential input gain of one-half balanced input buffer for my active crossovers. As I have long leads, I need a good CMRR. I was playing around on paper with the very devices he used, and believe that I could modify his design for my needs.

In the first stage, I thought that if I replaced R3 and R4 with a resistor of 28k (C2 omitted, of course) and R1 and R2 with a resistor of 56k, I would have a buffer with a gain of one-half and could place the negative input at the junction of R2/R3. I also believed that with reduced gain the amplifier might have greater DC stability, so that the servo could be omitted. Am I right? What are the required characteristics of the current mirrors?

I have been looking at the UK Farnell catalog, which has low-cost mirrors (BCV 61/62C NPN/PNP) with an ft of 250MHz (which seemed OK), hfe of 110/125, (which is lower than discrete devices), and a Vce sat of 0.6–0.65V at 100mA. Would this be a suitable alternative to using dual transistors?

David Field
fields@clara.net

Dr. Norman Thagard responds:

I am pleased that you found interest in my article. The folded cascode topology certainly makes op-amp realization from discrete components practical.

You can make a difference amp from this topology. You can replace R3 and R4 with a single resistor of 28kΩ and change R1 and R2 to 56kΩ. You cannot, however, place the input at the junction of R2/R3 unless the input is a current. This junction is a "virtual ground" due to negative feedback.

For good CMRR, apply your inverted voltage input to this node through R2. Referring to the schematic of Figure 1 of Part 1 of the article, this means that rather than ground the left side of R2, you would make it the inverting input terminal of your difference amplifier. Also, you need to insert a second 28kΩ resistor in series with the noninverting input by replacing L1 with this new resistor. If you wanted to get fancy, you could make

R1 variable and adjust it for maximum CMRR.

Also, you will probably need to keep some capacitor in the C2 position for compensation of the feedback amplifier. PSpice simulation of the modified circuit that you propose does not show an oscillation, but stage two of the phono preamp required compensation with less feedback, so there's every reason to believe that even more compensation will be required with your circuit. Due to the nature of the folded cascode, you can alternatively compensate by omitting C2 and shunting the output to ground through some value of capacitance. You will need to experiment to determine the appropriate value in either scheme. Use the minimum required value, as you are "narrow-banding" the amplifier; that is, you are trading frequency response for stability.

As you surmise, DC performance of the circuit will be improved due to the increased DC feedback. Whether or not it is improved to the point of omission of the servo, I cannot say. I shall find out in the near future, as I intend to use two such difference amps to form the input stage of an instrumentation amplifier in a differential-in, differential-out configuration. This will be the basis of a control preamp's line amplifier. Perhaps with a single gain-of-ten stage, DC drift will be sufficiently low as to permit elimination of the servo.

The phono preamp described in the article needed a servo because DC gain around two stages was over 1,000. You must always be mindful that DC drift in the preamplifier will be amplified by the DC gain of the amplifier that it drives. Of course, AC-coupled amplifiers will be little bothered by this unless drift produces a problem in the preamp itself. My amps are not only DC-coupled, but are DC amplifiers, so I cannot tolerate much DC drift from my preamp.

Difference amps are described in almost every electronics textbook. You might find it useful to read chapter two of "Design with Operational Amplifiers and Analog Integrated Circuits" by Dr. Sergio Franco (McGraw-Hill). Good luck and thanks for your interest.

TUBE DIFFERENCES

I wish to respond to the letter entitled "Tube Identity" regarding information on the differences between the PCL805 and PCL85 (*aX* 1/01, p. 94). According to my data, the pentode sections of each are significantly different. The 85 is specified as having 170V anode voltage, whereas the 805 is 100V. Grid voltage is –1 for the 805, but –15 for the 85. Screen voltage is the same for both: 170V.

Other specs are listed below. However, the triode sections of each are identical.

My source is *Radio Valve and Semiconductor Data*, 10th ed., Butterworth. The book is obsolete, as are the valves listed in it. Just out of curiosity, why hasn't anyone tried putting a MOSFET in a glass envelope incorporating a little lamp? That way, you'd get the superior performance of a MOSFET combined with all the marketing angles of a valve.

Martin Eccles
Newcastle-under-Lyme, England

	V_{HEATER}	I_{HEATER}	I_{ANODE}	I_{SCREEN}	R_{ANODE}	GM
PCL805	17.5V	0.3A	200mA	35mA	11k	5.5
PCL85	18V	0.3A	41mA	7mA	25k	7.25

WARM-UP TIME

I'm referring to Satoru Kobayashi's "A 30W Push-Pull 3CX300A1 Monoblock" (*aX* Jan. '01). It disturbs me at age 71 to see yet another fine amplifier powered so recklessly. The necessity for heater warm-up of at least three minutes is clearly demonstrated before the HV is applied (on p. 18 under the heading Other Information). At all military and research installations, a warm-up time of no less than 20 minutes was generally observed before HV was switched on and calibration began.

In my view, considering the relative scarcity and price of valves, a separate heater and switching tranny is no luxury for today's amps. Choosing 12V windings we now can put heaters in series and use passive delay components to avoid such appalling start-ups as Kobayashi describes.

In a letter in the same issue ("Buyer Beware," p. 100), we read about the fate of many good ol' good-ones that could well work today if the heaters had not been paralleled. Therefore, working out a safe and gentle power supply for such a precious musical instrument as a hi-end valve amp is time and effort well spent.

Mike Panymo
Eindhoven, The Netherlands

THEATER SOUND

The article entitled "Building an Altec 816 System" (*SB* 8/00, p. 30) wakes up some remembrances. First, let me say the cabinets are gorgeous!

The author states that the "Voice of the Theater" had an industrial look. Maybe this is a polite way of describing the cabinet making for theaters in vogue at Altec from the '40s to the '60s. The wood used was ¾″ bare plywood, poorly painted in matte black, with minimal bracing and polishing, using nails.

The cabinet appearance was the exact opposite of the drivers, which were absolutely beautiful, with huge Alnico magnets, a molded aluminum finely enameled frame, and a quality of construction which was well above the usual. And the compression drivers were a charm to repair. Replacing the voice coil took five to ten minutes and needed absolutely no adjustments; you placed it, screwed back the unit in place and that was it!

The crossover was a huge box, screwed anywhere on the cabinet, with a three- or five-position jumper or a potentiometer to adjust the level of the high-frequency driver.

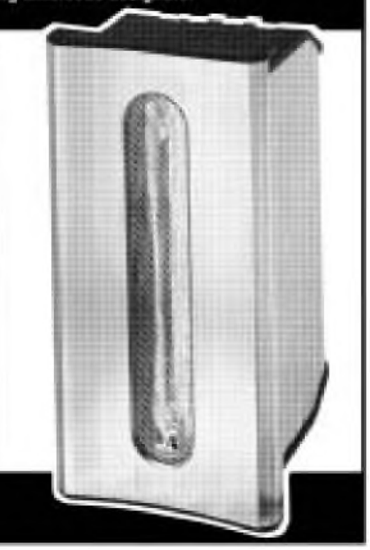
I still have one 815 15″ driver. The speaker fell off a scaffold: the box fell apart into pieces, the horn was bent a little, but the drivers were intact in perfect shape!

Since those speakers were hidden behind a screen where nobody would see them, the cabinet was unimportant. But they had to work day after day in almost any condition, and, since the show must go on, they were built like a battleship!

A little history: the A-7 was the baby of the family and was made for small theaters—100–200 patrons. The usual large theaters—1000–2000 seats—used the A-5, which was quite huge and used two 15″ drivers à la D'Appolito, in a vertical line one above the other. The high-frequency driver was always the same, but the horn was chosen according to the dimensions of the room. The catalog was quite huge, with dispersion characteristics covering almost every case.

Phasing was based on experience; to be polite, it was "cut and try." And the "sled" shown in Photo 24 in the *SB* article was the way those drivers were definitely installed! In mono, the speaker was in the center of the screen, the horn situated at about two-thirds of the height. In cinemascope theaters, with five speakers behind the screen, those speakers were installed following the curvature of the screen at around 1′ behind it. But the horns were targeted at a point situated at around two-thirds of the room in the center.

The usual way to do it was to put a big wheel of wire or a cardboard box in the chosen seat, removing the driver and orienting the horn by watching through the hole. As the author noted, phasing done with pink noise was quite



inconclusive. You needed to find the proper position according to voice coils, holding the drivers in place with C-clamps, and play a well-known soundtrack. You then walked through the theater and made any adjustments until you were satisfied. When you were, you usually tried out the system before a director you knew was a "difficult" customer. If he was pleased, you screwed everything in place! Part of that job had to be done with the screen in place because it changed the sound!

Equalizing "Voice of the Theaters" was not really an option. Flat frequency response was not really part of the characteristics of those speakers. They did not have very low frequency response, not much below 80cps, (obsolete unit of measurement replaced by hertz—Eds.). The crossover was around 500Hz, the low frequency unit had a peak around 300Hz, and the high at around 2-3kHz. There were not really any high frequencies above 8-10kHz, mainly due to the limitations of the optical soundtrack. And there was almost always a hole at the crossover frequency, but this was better for the ear than a peak at the same frequency!

But the efficiency was phenomenal. Most of those speakers were driven by a push-pull of 6L6s, an amplifier from 10W for the small room to 25-30W for huge theaters. Although used at only a fraction of their power, those amps made noise! They were installed in the projection booth, and were connected to the speaker through very long lines of AWG 14 to 16 wire. For the length, you simply doubled the size of those old theaters—300 hundred feet was not uncommon. Quite often when we worked in a theater, we connected a small 5-transistor radio—those with a ¼W output with 10% of distortion—directly to the speaker and played the radio, with the volume control far from being fully turned up.

In the theater of the '40s and '50s, the motion picture soundtracks were quite different from those of today. The most important thing was the dialog, followed closely by the music. And musicians knew very well the limitations of the optical soundtrack and the peculiarities of the "Voice of the Theaters." You can clearly hear this in scores such as those of Dimitri Tiomkin, with lots of strings or winds, but keeping the percussive sounds at a minimum and at a fairly low level; the optical would not "take it." But many of those picture sounds heard in one of the huge theaters was a real pleasure, an act of love. And those old enough to have seen "Sound of Music," "My Fair Lady," or "West Side Story" played in a 70mm-equipped theater with the five tracks behind the screen would really appreciate this.

And then in the '80s came Dolby stereo, with soundtracks featuring a lot of effects. The first—or one of the first—was "Apocalypse Now." The most noticed impact of the soundtrack was probably the sound of the helicopters. But this is another story, showing bass reflex design, JBLs, THX, and the rest.

Philippe Trolliet
Montréal, Québec

20W AMP BALANCE CONTROL

Regarding "A 20W $260 Amplifier" (*GA* 5/00), with the balance potentiometer in mid-position the input resistance of the amp is 33k. With today's low impedance sources, this balance control will not work unless it almost shortens out one channel. There are special stereo balance potentiometers that have zero resistance for each channel at mid-position and add, say, 50k in full clockwise or counterclockwise position for either the left or the right channel and should be connected in series with the potentiometer R1A.

A.J. van Doorn
Amersfoort, The Netherlands

Joseph Norwood Still responds:

My answer to your inquiry serves two purposes. The first is to respond to the questions you raised, while the second is to address any apprehension it may have caused among the readers/builders of the 20W stereo amplifier.

Your first statement suggests that today's consumer-electronics-manufactured stereo equipment has an output impedance in the 1k to 3k domain. This is not true. CDs, *DVDs, and AM/FM tuners have stated output impedances greater than 10kΩ, many have 50kΩ output impedances, and some have fixed output impedances stated as greater than 10kΩ with a variable output impedance of 50kΩ.*

Also many mixer consoles (for the non-professional market) have high impedance outputs of 27kΩ. When used with present-day CDs, DVDs, AM/FM tuners, and so on, the balance control of the 20W amplifier when adjusted to accommodate the most out-of-balance right-to-left channel signals is only rotated 10° to 15° off-center. (So your statement that the balance control must be rotated to an extreme right or left position is not true, as is your statement that CDs, DVDs, and AM/FM tuners have a very low output impedance.)

As you stated, the input resistance of the amplifier is 33kΩ, which is a compatible match with the greater than 10kΩ or 47kΩ output impedance of CDs, DVDs, AM/FM tuners, and so on in use today. I could have used a 200k balance control, but it had a somewhat wider correction span of 15° to 20° for out-of-balance right- and left-channel signal amplitudes.

In regards to the second part of your statement, "there are special stereo balance potentiometers," I know there are special stereo balance controls specifically designed to be used as balance controls! When I designed the 20W stereo amplifier, I wished to include this type of special balance control in my amplifier. None of the parts catalogs I received carry these controls. Apparently, they are special-ordered by the manufacturers of power amplifiers, and so on, from the manufacturers of potentiometers.

It is my policy not to include any item or component in my amplifiers that is not readily available, off-the-shelf, from parts-supply houses. It is very frustrating for the reader or potential builder of amplifiers, and so on, to discover an item is no longer available or available only through some obscure source. I would never subject the reader to such an inconvenience.

The balance control of the 20W amplifier has performed very well with no more than 15° of rotation off-center with severely unbalanced stereo signals. I operate the unit using a Kenwood AM/FM tuner model KT8300, Sony CD (5-disc) CA9ES, Sony CD (200 disc) model CDP-CX235, or Sony CD Walkman (line out). There is, however, a negative—the only negative pertaining to the balance control that the builder of the 20W amplifier will encounter— you failed to mention.

The simple balance control has cross-channel spillover (distortion). It is, for all practical purposes, so far down as not to be of primary concern; however, it is present. The bottom line is this circuit permits a low-band signal to be transferred from the right to the left channels and from the left to right channels. These transferred signals are very low in amplitude in relation to the main R- and L-channel audio signals. In the real world this crossover distortion was accepted in the amplifier for the convenience of employing a traditional system of adjusting volume and balance of the incoming stereo signal.

There is another approach the builder of the 20W stereo amplifier may take to eliminate this negative feature of crossover distortion. It is simply to remove the balance control (R) and use just a single 100k audio taper potentiometer for the R-channel and a single 100k audio taper potentiometer for the L-channel. These potentiometers are available from Radio Shack, part number 271-1722, cost $1.49.

This somewhat unconventional channel concept of having separate R and L volume controls takes about three seconds to adjust. So if you really want to make this change, don't hesitate. For potential builders of the 20W stereo amplifier, select either system, both work great!

CD REVIEW

I purchased the HHB recorder based upon your review (*AE* 2/00, p. 32) and have been very pleased with it. I am a home rather than a studio user and have wondered whether I can upgrade the sampling rate through the use of an external digital processor. One unit that has received rave reviews in *Stereophile* is the MSB Link DAC 111 D/A Converter. I'd like to use the AES/EBU output, but the HHB is unbalanced while the MSB's AES/EBU input is balanced. After re-reading your review regarding the HHB's internal DAC, I was wondering whether this would simply be "gilding the lily." Do you have any suggestions?

Paul Blake
Pblake44@frontiernet.net

Gary Galo responds:

Mr. Blake is correct when he points out that the HHB CDR-800 CD does not have a balanced AES/EBU digital output, nor does its successor, the CDR-850. Each is equipped with an S/PDIF coax digital output. Quite honestly, I have not been particularly enamored of the AES/EBU interface, unless the cable runs are long. With the short cable lengths normally found in consumer applications, the S/PDIF interface with a high quality interconnect cable will perform just as well sonically. I have actually found some instances in which the S/PDIF actually sounded better.

Also note that the basic MSB Link III you mention does not have an AES/EBU input—only S/PDIF. For AES/EBU, you will need to purchase the Nelson Link III or Gold Link III versions.

The HHB CD recorders can certainly benefit from a top-notch external D/A converter for playback, even without the upsampling feature available with many current D/A converters. The HHB CDR-800 Stable-Platter mechanism provides excellent support for the CD. This, combined with the robust construction of the HHB, makes the unit an ideal transport for a first-class digital playback system.

SE AMP UPDATE

I have a question regarding the power-supply circuit in "SVB300B SE Stereo Amp" (*GA* 6/00, Fig. 12, p. 20). Is there a misprint in the upgrade? The author stated that he has upgraded the supply to the filament by using a switching power supply, whereas the upgrade uses only a bridge rectifier.

Toh Eng Cheong
Eng_Cheong_Toh@notes.ttsh.gov.sg

Satoru Kobayashi responds:

1) I have corrected "6V × 2 ZD" to "15V × 2 ZD." This will secure the operation of MOSFET ripple filter better than ever.

2) The name of an IC regulator pin was missing. It is TL783C.

Finally, I think I chose an incorrect word to explain the power-supply diagram. This upgrade describes a power transformer rather than the rectifier circuit. At the time of design and assembly, I had only the old model of the power toroidal transformer, though I knew that Plitron had developed a new model dedicated to the 300B amplifier. After submitting my manuscript to audioXpress, *I added the new power-supply diagram to replace the original circuit.*

The reason I deleted switching power supplies from the original version was that some audiophiles do not prefer this type of new component, believing that a switching power supply affects the sound quality. I do not think so. Also, the new power transformer provides a couple of 5V taps to heat up 300Bs, whereas Plitron #754709 doesn't, requiring 5V extra.

Thus, you can decide which you prefer. If you choose the new one, then you will see the color matching among all transformers lined up over the chassis, though the old one does not match the other output transformers shown on the cover of the Glass Audio *issue.*

BALANCED AMP

I've built Grayson King's Valkyrie Preamplifier (*TAA* 1/94, p. 10), and I'm currently building the Thagard/Pass A75 Amplifier (*TAA* 4/92 and 1/93). The

A75 uses balanced inputs (XLR connectors), but the preamplifier outputs are RCA phono connectors. I don't see how these two different components can be connected together and deliver balanced signals to the differential pairs of the A75. The output from the preamp is (2) wires—one the signal, the other ground. The XLR connectors are (3) wires—two are the signal, the other ground.

Is there any modification for the preamp that would remedy the situation?

Jeff Reeves
Ontario, Canada

(See also Charles Hansen's "An Audio Balun," AE *6/00, pp. 16–23.)*

Norman Thagard responds:

My A75 amp has both RCA and XLR input jacks. A DPDT switch, designated S2 on the schematic of Fig. 9, allows you to select either balanced or unbalanced inputs. The center contact of the RCA jack should be attached to the (+) input, that is, to the input side of the 475Ω resistor, R4. Since this is the same point to which the #2 contact on the XLR connector attaches, and since the RCA and XLR connectors are probably located adjacent to one another, you may simply wire the RCA center contact to pin #2 on the XLR connector.

Actually, unless you anticipate using a balanced source in the future, the best approach is to omit the XLR connector. I don't know how the ground contact of the RCA jack is connected, and I would need to take my amp apart to determine that at this late date. It is certainly easy to insulate the outer (ground) contact of the RCA jack from the chassis and connect this contact to the same ground point to which pin #1 of the XLR jack would have been attached.

If you refer to Fig. 17 of the article, you might note that, if you have already built the amp with only the XLR connector in place, you can still use an unbalanced source. You will need to buy or construct an RCA-plug-to-XLR-plug adaptor. You can clip one end off an ordinary RCA interconnect and wire the center conductor to pin #2 on the XLR plug. Wire the interconnect shield to pin #1 of the XLR plug. No connection need be made to pin #3 of the XLR connector. The DPDT switch on the A75 should be in the "unbalanced" position any-time you are using an unbalanced source.

The input grounding technique for my own designs is to connect the outer RCA jack contact to an immediately adjacent chassis point through a 10Ω, 1W resistor. The reason for the 1W power rating is that resistors have voltage ratings that can be lower than their power rating might suggest. I have had to replace 0.25W resistors used for this purpose because voltage spikes caused them to open. The reason for grounding through a 10Ω resistor is to avoid a possible ground loop that can produce audible hum at the amplifier's output.

I hope that this description allows you to modify or construct your A75 to accept unbalanced sources. Like you, I have no balanced source and can state unequivocally that the A75 works very well with the much more common unbalanced-output preamplifiers.

HELP WANTED

I'm looking for plans or good dimensions for speaker plans for sound-reinforcement applications. I typically run double 18" JBL speakers that are rated at 1200W at 4Ω, 20Hz to about 200Hz and then 15", and so on.

If you have some input in this regard, let me know. ❖

Mark Himberger
mthimberger@Equilon.com

New Chips on the Block

Analog Devices AD629 Diff Amp

By Charles Hansen

Now you can accurately measure differential signals in the presence of high common-mode voltages. The AD629 is a difference amplifier with a very high input common-mode voltage range up to ±250V. A large resistive on-chip voltage divider attenuates common-mode signals by 20:1.

The AD629 provides cost-effective isolation and can replace costly isolation amplifiers in applications that do not require galvanic isolation. The device will operate over a ±250V common-mode voltage range and has inputs that are protected from common-mode or differential voltage transients up to ±500V.

The AD629 has low offset and gain drift specifications. This keeps errors low while operating over a wide temperature range. Additionally, the device provides high common-mode rejection. Excellent CMR is essential to reduce measurement errors in the presence of high common-mode input voltages.

Specifications include 77dB minimum CMRR, 20μV/°C offset drift, and 0.001% maximum nonlinearity. Input impedance is 200k common-mode and 800k differential. Typical full-power bandwidth is 28kHz, with 1.7V/μsec minimum slew rate. The device is pin-compatible with the INA117.

The AD629 is available in low-cost plastic 8-pin DIP and SOIC packages. For all packages and grades, performance is guaranteed over the entire industrial temperature range from −40°C to 85°. Prices start at $2.25 (1k pieces).

Visit www.analog.com/diff-amp or call 1-800-ANALOGD. ❖

Burr-Brown DVD Audio Playback Chip

By Charles Hansen

The Burr-Brown 192kHz DVD Audio Playback chip set consists of the PCM 1737/1739 audio DAC, the PCM1604/1605 audio DAC, and the PLL1700 multi-clock generator. These ICs are designed for use with the OPA134 series of FET-input ultra-low distortion, low-noise op amps (fully specified for audio applications) to provide superior sound quality.

Applications include A/V receivers, DVD movie and audio players, DVD add-on cards for high-end PCs, HDTV receivers, car audio systems, and other applications requiring 24-bit audio.

PCM1737/1739 AUDIO DAC

The PCM1737/PCM1739 SoundPlus™ 24-bit, 192kHz sampling enhanced multi-level, delta-sigma, audio digital-to-analog converters are CMOS, monolithic, integrated circuits, including stereo digital-to-analog converters and support circuitry in a small 28-lead SSOP package. The data converters utilize Burr-Brown's enhanced multi-level delta-sigma architecture, which employs fourth-order noise shaping and eight-level amplitude quantization to achieve excellent dynamic performance and improved tolerance to clock jitter.

The PCM1737/1739 accepts industry-standard audio data formats with 16- to 24-bit data, providing easy interfacing to audio DSP and decoder chips. Sampling rates up to 192kHz are supported. A full set of user-programmable functions are accessible through a four-wire serial control port that supports register write and read back functions. The PCM1737 is software-controlled, while the PCM1739 is hardware-controlled.

PCM1604/1605 AUDIO DAC

The PCM1604/PCM1605 SoundPlus™ 24-bit, 192kHz sampling enhanced multi-level, delta-sigma, audio digital-to-analog converters are monolithic, integrated circuits that include six 24-bit digital-to-analog converters and support circuitry in a 48-lead LQFP (PCM1604) or MQFP (PCM1605) package. The data converters utilize Burr-Brown's enhanced multi-level delta-sigma architecture, which employs fourth-order noise

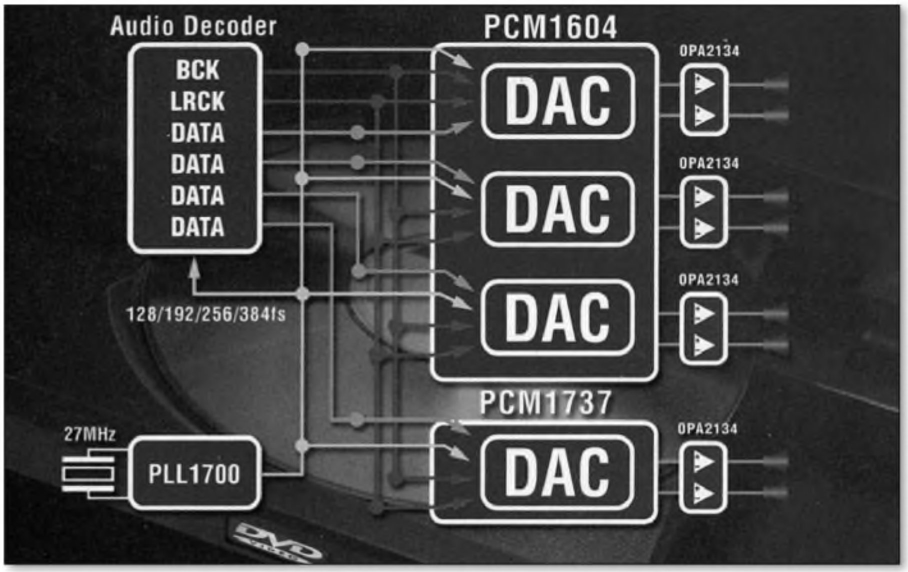


FIGURE 1: Burr-Brown DVD chips.

PCM1737/1739 FEATURES

24-bit resolution
Analog performance (Vcc = +5V, Vdd = +3.3V):
Dynamic range: 106dB typ
SNR: 106dB typ
THD+N: 0.0015% typ (–96dB)
Full-scale output: 3.1V p-p typ
4×/8× oversampling digital filter:
Passband: 0.454fS
Stopband: 0.546fS
Stopband attenuation: –82dB
Passband ripple: ±0.002dB
Sampling frequency: 10kHz to 192kHz
System clock: 128, 192, 256, 384, 512, or 768fS with auto detect
Accepts 16-, 18-, 20-, and 24-bit audio data
Data formats: Standard, I^{2}S, and left-justified
User-programmable mode controls:
Digital attenuation: 0dB to –63dB, 0.5dB/step
Digital de-emphasis
Digital filter rolloff: sharp or slow
Soft mute
Variable oversampling for DS DACs
Zero detect mute
Zero flags for each output
Dual supply operation:
+5V analog, +3.3V digital
5V tolerant digital logic inputs
Small 28-lead SSOP package, recommended US resale: $4.15 (1k)

PCM1604/1605 FEATURES

24-bit resolution
Analog performance (Vcc = +5V, Vdd = +3.3V):
Dynamic range: 105dB typ
SNR: 104dB typ
THD+N: 0.0018% typ (–95dB)
Full-scale output: 3.1V p-p typ
8× oversampling digital filter:
Passband: 0.454fS
Stopband: 0.546fS
Stopband attenuation: –82dB
Passband ripple: ±0.002dB
Sampling frequency: 10kHz to 200kHz
System clock: 128, 192, 256, 384, 512, or 768fS with auto detect
Accepts 16-, 18-, 20-, and 24-bit audio data
Data formats: Standard, I^{2}S, and left-justified
User-programmable mode controls:
Digital attenuation: 0dB to –63dB, 0.5dB/step
Digital de-emphasis
Digital filter rolloff: sharp or slow
Soft mute
Zero detect mute
Zero flags for each output
Dual supply operation:
+5V analog, +3.3V digital
5V tolerant digital logic inputs
Small 48-lead QFP package, recommended US resale: $6.95 (1k)

shaping and eight-level amplitude quantization to achieve excellent dynamic performance and improved tolerance to clock jitter.

The PCM1604/1605 accepts industry-standard audio data formats with 16- to 24-bit data, providing easy interfacing to audio DSP and decoder chips. Sampling rates up to 192kHz are supported. On-chip enhancements include an 8× oversampling interpolation filter, digital attenuation, mute and zero flags. The PCM1604 is software-controlled, while the PCM1605 is hardware-controlled.

PLL1700 MULTI-CLOCK GENERATOR

The PLL1700 is a multi-clock generator phase-locked loop (PLL) designed for DVD systems. The device provides both cost and space savings by eliminating external components, and achieves the very low jitter performance needed for high-performance audio digital-to-analog converters (DAC) and/or analog-to-digital converters (ADC).

The PLL1700 is ideal for MPEG-2 applications that use a 27MHz master clock such as DVD players, DVD add-on cards for multimedia PCs, digital HDTV systems, and set-top boxes. The device accepts a 27MHz reference clock and generates the following zero-PPM error output clocks audio system clocks:

SCK01—33.8688MHz fixed for CD/DA and DVD DSP
SCK02—256fS for a Dolby Digital/MPEG Audio Decoder
SCK03—348fS for the DAC(s) in the system
SCK04—768fS for a karaoke DSP

The device has low clock jitter (150ps at SCK03) with multiple sampling frequencies of f_S = 32kHz, 44.1kHz, 48kHz, 64kHz, 88.2kHz, and 96kHz. It has dual power supply capability at +5V and +3.3V, with +3.3V CMOS logic interface.

Small 20-lead SSOP package, recommended US resale: $1.95 (1k), (800) 548-6132, www.burr-brown.com

❖

Ohmite Heatsink Resistors

By Charles Hansen

Ohmite Manufacturing has introduced two series of noninductive TO-220-style thick-film power resistors, good candidates for use in high-frequency applications. The resistors are manufactured in an ISO-9000-registered facility.

TAH20

The new TAH20 series 20W is rated for 20W at 25°C case temperature. The TAH20 carries a high joule rating and is, therefore, more reliable in pulse applications. The noninductive characteristic reduces voltage spikes for protection of other sensitive components in snubber applications. The molded symmetrical package outline is designed for use with a snap-in-style heatsink, which means that assembly time and effort are greatly reduced (a thermal heatsink compound should be used).

FEATURES

20W power rating at 25°C case temperature
High pulse and transient tolerant design
TO-220 package configuration
Quick-snap molded package
Resistor element environmentally protected
Noninductive design
Resistor package electrically isolated from heatsink
Low thermal resistance to heatsink at RTH<6.25°C/W

SPECIFICATIONS

Electrical
Resistance range: 0.05Ω to 10kΩ; other values available upon request
Tolerance: ±5% stock, ±0.5%, ±10% available
Temperature coefficient: referenced to 25°C, ΔR taken at +105°C
1 to 10Ω: ±(100ppm + 0.002Ω)/°C
10Ω and up: ±50ppm/°C
Maximum operating voltage: 350V
Dielectric strength: 1,800V AC
Power rating: 20W at 25°C case temperature, derating linearly from 50°C to 150°C
Insulation resistance: 10GΩ minimum
Momentary overload: 2× rated power for five seconds where applied voltage is ≤1.5 times operating voltage
ΔR ±(0.3% + 0.001Ω) maximum
Lead material: tinned copper
Mounting: requires the use of a snap-on style heatsink—a thermal compound should be applied.

TCH35

Ohmite's new TCH35 TO-220 package resistor, designed for intermediate power applications, provides 35W of continuous power using the specified heatsink.

These noninductive resistors give you effective control over voltage spikes in snubber applications, which make them ideal for switching power supplies, high-frequency applications, and pulse loading. These resistors are built under proprietary processes that Ohmite claims provides 75% more power-handling capability than other TO-220 package resistors of similar size. Standard lead forms are provided for easy assembly, and the molded case provides environmental protection for the resistor element.

A single screw-mounting tab helps simplify assembly. Generally, heatsink systems operating below 150°C rely on conduction and convection as the primary means of heat removal. The TCH35 series offers a low thermal resistance to the heatsink of <4.28°C/W to support conduction efficiency, and helps define heatsink cooling requirements. A thermal compound should be applied.

SPECIFICATIONS

Electrical
Resistance range: 1Ω to 10kΩ (higher values available)
Resistance tolerance: ±1% to ±10%, 5% standard (0.5% available on request)
Temperature coefficient: referenced to 25°C, ΔR taken at +105°C
1Ω to 10Ω: ±(100ppm + 0.002Ω)/°C
10Ω and above: ±50ppm°C
Maximum operating voltage: 350V
Dielectric strength: 1800V AC
Insulation resistance: 10GΩ minimum
Momentary overload: 2× rated power for five seconds as long as the applied voltage is ≤1.5 the continuous operating voltage, where ΔR ±(0.3% + 0.01Ω) maximum
Lead material: tinned copper
Power rating: 35W at 25°C, case temperature, derating linearly from 50°C to 175°C

Ohmite Mfg. Co., 3601 Howard St., Skokie, IL 60076, (847) 675-2600, Fax: (847) 675-1505, ohmite@wwa.com, www.ohmite.com.

❖

OhmCraft Audio Resistor

By Charles Hansen

OhmCraft claims that in high-end audio applications their extremely low-noise FineFilm resistors are transparent in both low-signal applications, such as microphones, and high-wattage applications such as power amplifiers. Made by a patented process, each OhmCraft FineFilm resistor is written in a long serpentine pattern on a ceramic substrate using a thin-film ink. The process is extremely versatile. By controlling the thickness, width, and length of the drawn line as well as the ink conductivity, OhmCraft can manufacture resistors of outstanding performance over an extremely wide range of values.

There are no hard-tooling costs in the FineFilm process. New resistor models are easily produced using the OhmCraft CAD/CAM program together with the correct pen orifice and decade-value ink. Since they are made the same, production parts perform identically to prototypes.

After the resistors are drawn, conventional manufacturing techniques are used to form a broad spectrum of surface-mount and through-hole parts—including networks and dividers. OhmCraft says that their FineFilm resistors combine the close tracking, low current noise, high-current-voltage linearity, and close resistance tolerance of thin films together with the high durability, wide resistance ranges, and low cost of thick films.

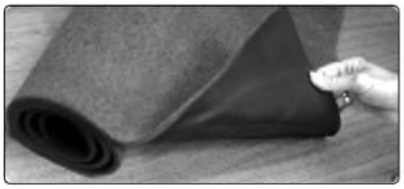
PERFORMANCE

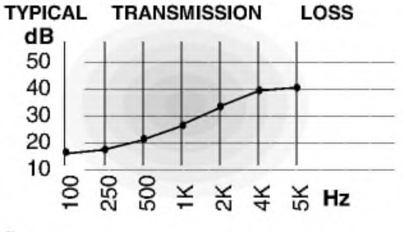
Unequaled peak pulse tolerance: The longer trace in FineFilm resistors leads to greatly reduced internal voltage gradients, which allow it to handle intermittent voltage spikes and ESD much better than conventional designs.

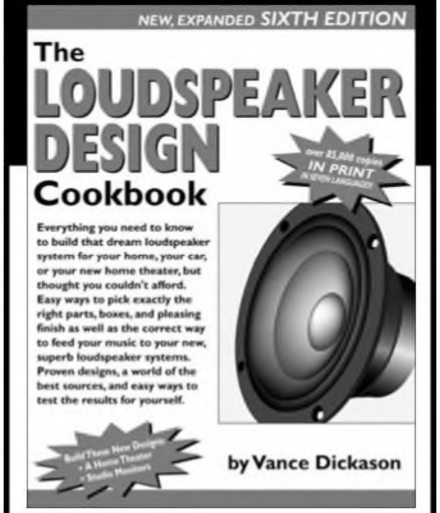
Low noise: FineFilm resistor noise characteristics are similar to thin-film resistors.

Far higher operating voltages: FineFilm chip resistors offer voltage ratings at least an order of magnitude higher than conventional resistors of the same size.

Superior voltage linearity: The voltage linearity of a typical FineFilm network exceeds that of a competitive product by an order of magnitude.







Exceptional stability: Thanks to their low-resistivity composition, the stability of FineFilm resistors is significantly better than conventional designs, as indicated by their low thermal coefficient of resistivity and virtually flat voltage coefficient of resistivity across a wide range of values. Ratio is also outstanding.

Corrosion immunity: FineFilm resistor traces are made of glass and conducting oxide constituents, which are totally free of corrosion effects.

Non-inductive design: FineFilm designs—both planar and cylindrical—maintain a constant impedance at high frequencies by utilizing serpentine patterns in which adjacent traces carry current in opposite directions.

Close resistance tolerances: The long-path configuration of FineFilm resistors permits laser trimming of such features as ladders, loops, or top hats, which provide separate regions of high and low trim sensitivity—unlike conventional thick-film trimming, which is limited to destabilizing and damage-prone single notch or plunge cuts.

PRECISION DISCRETE RESISTORS

Values to teraohms
Fractional percentage tolerances
Very low voltage dependence
Low noise and TCR
Low inductance
Enhanced stability
Conformal encapsulation
Spade or wire leads
Immediate, tool-less prototyping
Quick-change ink cartridges
Easy design changes
Rapid, low-cost setup
High-fidelity line quality

Typical Performance:
Overall tolerance: to 0.1%
Absolute TCR: 25 and 50ppm/°C
VCR: 0.05ppm/V
Load life: 0.15% maximum ratio change
Over voltage: 1.5× rating (5 sec)
Derating: to zero at 125°C

Standard Values:
Tolerances to 0.1%
Power ratings to 6W
Voltage ratings to 40kV
Resistances to GΩ

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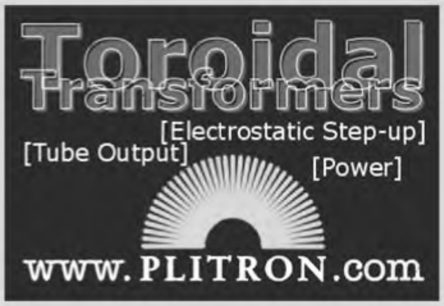

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Book Review

Radio Tubes and Boxes of the 1920s

Reviewed by Scott Frankland

Radio Tubes and Boxes of the 1920s, *by George A. Fathauer, Old Colony Sound Lab, 305 Union St., PO Box 876, Peterborough, NH 03458-0876, 603-924-6371, FAX 603-924-9467, e-mail custserv@audioXpress.com, $26.99, 105pp., softbound, 8½ × 11, $26.95. BKV4*

Most books about vintage radios feature photographs of luminous tuning sets peeking seductively out of curvaceous wood cabinets. *Radio Tubes and Boxes of the 1920s*, on the other hand, presents the radio boom from the inside out. Rather than consoles, what you will find pictured here are hundreds of radio tubes, posed beside their original cartons. In addition, there are brochures and ads, plus announcements about tubes of special significance, all of which provide a unique "insider's view" of the radio boom.

THE TUBE THAT LAUNCHED A BOOM

Although a great many '20s tube types are shown here, the 201A triode is by far the most prevalent. The reason is not hard to fathom: The 201A was the first "acceptable" tube in America, in the sense that it was cheap, reliable, and efficient in use.

This commercial and technical breakthrough was the brainchild of Irving Langmuir at General Electric. Technically, the 201A was a tube of many firsts. It was among the first commercial tubes to include a getter, a thoriated filament, and a hard vacuum. These three techniques combined their strengths to provide a tube preeminent in its practicality.

The 201A's influence was widespread: all told, there were some 500 brands offered during the '20s! While many of these tubes were produced in small batches (and some were rebrands), today you can only marvel at the variety. The typical 201A had a mu of 8, an "output resistance" of 12,000, and a plate current of 3mA. Puny, perhaps (by today's standards), but nonetheless sufficient to power the radio boom.

TUBES ON THE WILD FRONTIER

Throughout the '20s, radio manufacturers came and went, some years by the hundred. The one constant throughout the changing fortunes of the radio boom was the demand for tubes. Since tubes were the softest commodity of the radio industry, there was enormous interest in cracking the market. In an effort to stand out amid the ensuing mad scramble for market share, tube marketeers of the '20s raised hyperbole-in-advertising to an art form.

Take the Myers Company, for example, whose slogan was "Myers tubes, practically unbreakable." And, yes, they were glass tubes (but with Bakelite endcaps)*. Sovereign's ads were a step more brazen, however, claiming "No hum, no microphonics, no noise." What I want to know is, did they turn the damn things on?

Televocal tubes were "quick heating" and, I take it, designed for persons of dubious patience. The "ever-dependable" 222 from Diatron would "increase volume" and give "the finest quality of tone." The ostensibly rugged tubes from Triad promised to "reduce service calls" while the apparently invulnerable Milo Radio tube was "the tube that never fails."

The extramundane tubes from the Van Horne Company would "make a world of difference in your reception." The Gold Seal Company employed a yet more celestial approach, exhorting its customers to "enjoy perfection." And the Crosely Corporation anticipated the modern demand for realism with their slogan: "You're There With a Crosely Radio Tube."

Then again, there was Magnavox. I will let their ad copy stand without comment: "The most noticeable feature of the new Magnavox Radio Tube consists in eliminating the grid."

THE NAME GAME

In addition to the rampant sloganeering, many manufacturers banked on romantically tinged brand names to spruce up their profile—names such as Air-King, Blue Streak, Mello-Tron, Music Master, Royal Blue, Sky-Sweeper, Songbird, and Sunlight Crusader. Then there were the tonalists: Belltone, Cleartone, Goldentone, and O-T Silvertone ("Clear as a Silver Bell"). And the one-upper 'tones: Live-Tone, Real-Tone, Tru Tone, Wonder-Tone, and of course, Perfectone.

Like the 'tones, the 'trons came in for more than a few cloying acronyms. Amplitron, Cleartron ("The Master of Space"), Duratron, Dynatron, Musiktron, Radiotron, Supertron, Teletron, and, of course, Thermatron. These are not so bad, but there were some truly vacuous names as well: Blazing Train, Blo-Pruf, Echotron, Good Luck, Goode Tube, Magictron, Kazoo, Ozarka, Unitone No Bee, and my personal favorite Vacobub (a beautiful bulb in spite of its goofy name).

Not all tube companies relied on hyperbole, however; many were quite sober and straightforward in their approach to advertising. Moreover, there existed a panoply of comforting blue-chip name brands: Cunningham, General Electric, Hytron, Ken-Rad, National Union, Philips, Raytheon, RCA, Sylvania, Tung-Sol, and Westinghouse, among others. All of the foregoing are revealed in color, many for the first time, in the more than 400 color photos of this compact encyclopedia of radio tubes. The brands are conveniently arranged in alphabetical order.

THE PHOTOS

As a dedicated tube man, I found that many of these photos took my breath away. Of particular note are the "blues." Arcturus, for example, produced a lovely series of tinted blue bulbs. I must admit, there is something alluring about a blue tube (yes, even the gassy ones).

Another beatific blue tube of note is the Beacon Blue, which had a curvaceous pear shape and smooth, tipless crown. And, of course, there is the fabulous Brightson True Blue, which came packed in a blue, velvet-lined case. It was accordingly touted as "The Finest Radio Tube in the World!" Maybe not, but at least you knew it would arrive safely.

Sodion, on the other hand, doggedly pursued authentic technical innovations, not all of which were uniformly successful. The D-21 employed an elongated—and frosted—glass envelope of elegantly smooth proportions. Gorgeous, I think, is not too strong a word here.

Their boxes and cartons, not just the tubes, could in many cases be striking. The De Forest D-series tubes, for example, came packed in metal canisters, complete with the Master's visage conspicuous on the can. The metal packaging was evidently intended to project an aura of rugged dependability.

Other manufacturers found different ways to project quality in their presentation. Arcturus boxes, with their celestial scene and observatory, seemed to gaze heavenward. More down-to-earth, but nonetheless appealing in its own right, was the Leader box—a plain-as-Jane cardboard cylinder carton à la Quaker Oats.

Lightning bolts were a common styling motif, and Continental Tube Labs chose to use a myriad of bolts. The tube packed within Continental's lightning-laden box was evidently lightning-proof, being fully ensconced within a copper sheath of—shall we say—masculine proportions. The "copper condom" would not be amiss here.

Most of the tubes from this period exhibit the classic pear shape ("S" envelope) with clear glass, although there are countless variations among the brands. Quite a number of tubes had attractive logos affixed to the glass. The bases could be interesting, too, as they were often of ceramic or metal—with some brass, some nickel, and some brightly polished.

THE PUBLISHER

This eye-popping picture book originates from Sonoran Publishing of Chandler, Ariz. Sonoran is the dreamchild of George A. Fathauer (son of George H. Fathauer) of the famous father-son team who built up Antique Electronic Supply from a gleam-in-the-eye beginning in 1982. Having built AES into a high-profile purveyor of radio parts and receiving tubes, the two Georges then sold the company in 1994, only to come full circle. Later that same year, the younger George fired-up Sonoran Publishing with the aim of producing world-class books about his original interest—antique radios and their innards. George Sr., too, has returned to his roots; having become a specialty dealer of collectable radio tubes.

Sonoran Publishing is committed to preserving the heritage of the radio age in all its complex dimensions. This is the first book by the publisher himself, and it is an auspicious debut. The book's physical execution and graphical presentation are first-rate—reminiscent of the superb *70 Years of Radio Tubes and Valves* (also from Sonoran Publishing). The photos and the paper in both books are of equal high quality. Information on Sonoran's books is available at their website sonoranpublishing.com.

IN RETRO

Mr. Fathauer has fashioned a dazzling panorama of thermionic incunabula for the tube world. The emotions evoked by the various photographs, brand names, and slogans range from the sublime to the ridiculous; but throughout it all there is an abiding sense of wonder. *Radio Tubes and Boxes of the 1920s* will surely rocket to the top of every tube collector's wish list, and tube lovers of every lineage will no doubt find something of interest here.

The basic feel of the book is wall-to-wall tubes and their containers. That, plus the many period ads and announcements, all combine to instill a palpable feeling of "you-are-there" (retubing, it may be, the elegant receivers of yesteryear). Anyone interested in vacuum tubes should consider owning this book, as it rounds out an important stable of tube classics by Tyne, Stokes, Thrower, and Mager. These latter books highlight the historical and technical aspects of tubes. In contrast, *Radio Tubes* emphasizes their aesthetical side—their color, their diaphanous beauty, and their sensuous curves—revealed here as in no other book I have yet seen. ❖

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* The colorful career of Elman B. Myers is detailed by Gerald Tyne in *Saga of the Vacuum Tube*, wherein Myers is drawn as the archetypal "bootlegger."

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Showcase

Magnetic Recycle

I thought this might make an interesting, silly, or humorous submission to *audioXpress*. This is a stack that I have collected from various blown or damaged woofers and tweeters. Included are some from Audax, Dynaudio, Focal, Philips, and Vifa. I estimate the value of the blown drivers to be in the neighborhood of $600! ❖

Alan Ersen
Sacramento, Calif.

PHOTO 1: Is it the Leaning Tower of Pisa?

PHOTO 2: It's a stack of ferrite magnets salvaged from dead woofers and tweeters.

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Many used computer grade-capacitors and AC line filters up to 250V AC @ 60A, other miscellaneous electronic components too. For a complete list, e-mail perizzo@enter.net.

26 Eminent Technology LFT-8 planar midrange panel drivers, new, in factory-sealed wrappings, 6″ × 44″ × 1½″, 5Ω, $175 each—will sell all or part. Ken, daytime: (800) 426-1828, evenings and weekends: (804) 794-4107.

Vandersteen 2As, original owner, need grille and foam surrounds, $150 pr; Magnum Dynalab FT101 tuner, mint, $325. Contact David: (562) 924-2666, FAX (562) 924-9626.

Magnaquest SE output transformers, TFA 204, 3000W primary, 4, 8, 16W secondary, 8W–two pairs. One pair brand new, $175; one pair slightly used (leads not cut), $150. Mark Heinlein, PO Box 1764, Bend, OR 97709, (541) 388-7584.

WANTED

Shure M33 or M77 cartridge in working condition—stylus irrelevant, as I intend to use with stylus I already have. James Wilkinson, 197 Rutledge Ave., Charleston, SC 29403, (843) 723-9856, jimmy@cs.cofc.edu.

Schematics for Harman Kardon Citation A preamplifier. Will pay for P&P. Reply to maristranslations@telinco.co.uk.

Schematic for a Maas-Rowe Carillon amplifier Model 500A built by Bell Air Corp., Glendale, CA. Tube comp.: 2-6550, 2-5V4, 2-6SN7, 6SJ7, 6SC7, 6J7—1960s vintage. E-mail Bob, ab4hq@hotmail.com.

Front panel for HK Citation I stereo preamp. Is anyone making a replica? Phil Taylor, 3 Silver Lane, Billingshurst, W. Sussex RH14 9RP, England, phone/FAX +44 1403 786250, 10AM–10PM UK time.

Focal 7K011DBL woofer and T120k or TI replacement coils, surround processor preamp. Contact David, (562) 924-2666, FAX (562) 924-9626.

Test Tracks

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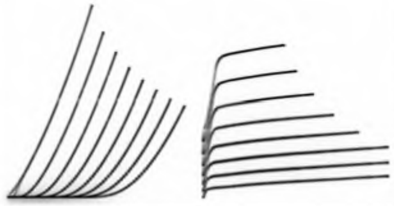
When we audiophiles sanctify "the absolute sound," we need to remember that there aren't many venues where we get to hear it. My wife, a piano fan, often prefers some of our recordings to the sounds we get in many halls (although Vienna's *Musikverein* certainly provided an exception).

At stores that don't feature phono hookups, I replace the LPs in this list with CDs: a good concert band CD (the break strains in "Semper Fidelis" and "Footlifters" can turn most systems to mush), a complex orchestral album (Falla's *The Three-Cornered Hat*, London 414039-2/10), and a jazz combo (Dave Brubeck's *Time Out*). ❖

Desmond Grier
Minneapolis, Minn.

Let's hear from you. Simply describe your seven favorite pieces (not to exceed 1000 words); include the names of the music, composer, manufacturer, and manufacturer's number; and send to "Test Tracks," Audio Amatuer, Inc., Box 876, Peterborough, NH 03458. We will pay a modest stipend to readers whose submissions are chosen for publication.



Test Tracks

Some technically fine recordings with mediocre performances make some systems sound better than they are by presenting rarely heard sound quality without the distraction of seductive music. Some technically weak albums (e.g., "spiked" treble or bass) may enhance the sound of some cheaper systems and sound execrable on a good system, making the better system seem at fault. Test albums need to span a range of ensemble sizes and complexity while being enjoyable—garage doors and synthesized pianos won't do it.

I employ three auditioning systems—one relatively expensive; one decent, inexpensive system; and a boom-box. If an album pleases me musically and sounds progressively better at each level, it becomes a candidate.

1. *Music for Two Guitars* (Oscar Caceres and Turibo Santos, Musical Heritage Society, MHS 1944 LP).
From its opening Scarlatti sonata, this relaxing set, mostly of duo guitar transcriptions, spans five centuries of composition. The Scarlatti and Cimarosa sonatas are particularly intriguing, but the entire album is a perfect "lights out" escape into other times and places. High-end systems, panels, and electrostatic speakers really shine with this recording, while lesser systems tend to constrict its lifelike image. [MHS is still very active in the CD era. Their address is 1710 Highway 35, Oakhurst, NJ 07755.]

Look into MHS, which has been releasing gems like this and the sadly ignored *Hummel Mandolin Concerto* (MHS 1701, LP) for years. They seem able to find performances of great early literature that orchestral programmers have overlooked. I've had much luck finding MHS LPs at estate sales and used record shops.

2. Poulenc, *Concerto pour 2 pianos et orchestre* (Katia and Marielle LaBeque, Seiji Ozawa with Boston Symphony Orchestra, Philips D 105623).
You need some piano. If one piano is good, then two pianos....The dynamic range on this album is wide, the interplay between the two pianos and orchestra reveals a system's resolving power, and the music is downright fun. Besides the Poulenc concerto, there are four more of his works for two pianos and Milhaud's *Scaramouche*.

3. Berlioz, *Symphonie Fantastique* (Ataulfo Argenta with the Paris Conservatory Orchestra, London CS 6025).
This is one of the classic interpretations of the *Fantastique* and has been reissued on archive labels. I prefer its rhythm and balance across the movements to any of the many recorded and live performances I've heard. It offers excellent tests of full orchestral dynamics and very subtle, realistic explorations of the bass range. Only the best systems reveal everything on this recording.

Although this is my favorite, there are many fine CDs and LPs of this work. The *Fantastique* makes a fine system tester in terms of varied orchestration and nuance.

4. *West Side Story* (Original Broadway Cast, Columbia CK 32603).
I've owned the LP since 1960 but purchased the CD on J. Gordon Holt's recommendation and could not believe how much better it is. An inferior system will not handle the chorale "Tonight," making this a great recording for exposing purportedly "high end" systems that lack cohesiveness. Don't mistake the movie soundtrack for the highly musical original cast album.

5. Stravinsky, *L'Histoire du Soldat* (Robert Mandel with Ars Nova, Westminster WST-1401, LP).
This elegant interpretation of a thorny, yet accessible, 20th century work provides an almost microscopic audio study of the chamber ensemble. This is Stravinsky's version of the Faustian theme, and a demonic work it is, as well as a surprising musical adventure. Each playing reveals something new. Most live and recorded versions that I've heard seem disjointed by comparison.

6. Mozart, *Concerto for Clarinet and Orchestra* (Robert Marcellus with George Szell, Cleveland Orchestra, Columbia LP MS 6968, reissued on CD).
Szell was one of the great orchestral accompanists for soloists, and this exemplifies his best. Marcellus, whose career tragically ended early due to a crippling disease, blends with Szell in a performance that is mostly duet for clarinet and orchestra. One passage in the first movement is so strikingly coordinated that hairs on the back of my neck literally rise whenever I hear it played on a good system.

I've owned many versions of this work and have often heard it live. This stands out as my favorite interpretation of my favorite work. The CD does not sound quite as good as the LP, but don't let that stop you. Thea King's Hyperion release of this concerto has the advantage of the richer sound of the basset clarinet, but Marcellus's performance, for me, rules.

7. John Prine, *In Spite of Ourselves* (Oh Boy Records OBR-019).
Auditioning audio systems can become tedious by about the fifth store, so it's best to carry one album that can loosen you up while still giving you some perspective on the systems. Prine's "Bruised Orange" offers a wider range of sounds and somewhat higher fidelity, but he did audiophiles a favor by setting up duets with a variety of female voices and then highlighting his own and their voices quite nicely.

This album also brought to mind the question of live versus recorded performances. I recently heard Prine sing a number of duets with Iris DeMent at a live concert. Her voice sounded awful—scratchy, sibilant, and distorted. When I got home, I put on the album, and there she was in all her glorious clarity and twang. The hall acoustics and horn tweeters at the concert had ruined the work of a fine performer.

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