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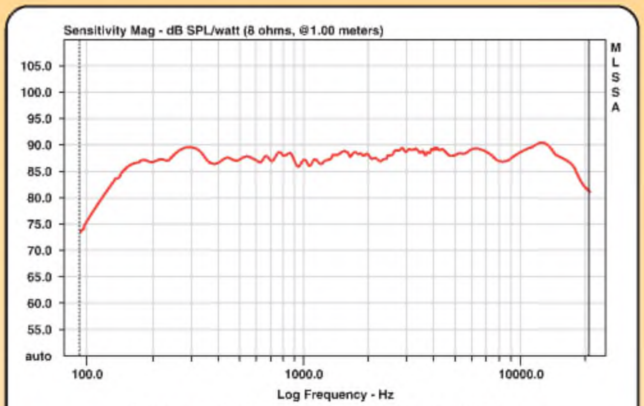

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264-720	Neo3FP	20W	2,000-20,000	5-1/8" diameter x 3/4" deep
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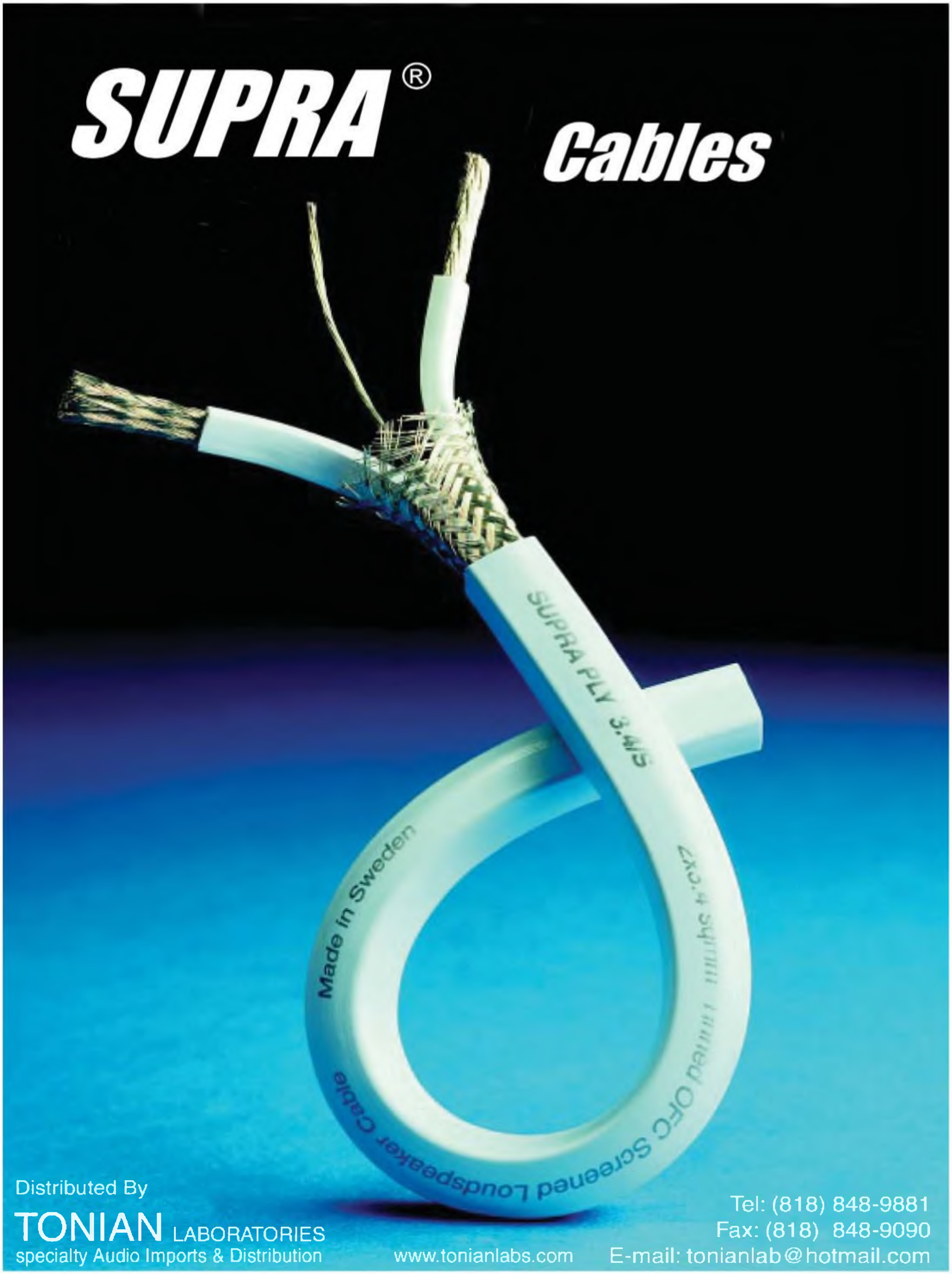
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Editorial

Interdependency

I am pleased to announce the appointment of Strategic Media Marketing of Gloucester, Massachusetts, as the new Advertising Representative for *audioXpress* magazine. SMM's owner is no stranger to our pages, however. Peter B. Wostrel was part of our advertising sales staff some years back. We are happy to welcome him and his organization to this relationship and are confident that his staff's sales skills can improve the advertising levels in our fledgling periodical.

If you have been aware of our advertising lineage over our first year and a half, you will see mostly a decline, which has been reflected in the total page count. Some of this has doubtless been an influence of the recession we are just now coming out of. But the rest of it is directly related to sales skills, or lack of them—in this publisher's opinion.

Most magazines must maintain at least 40 to 50% advertising levels in order to survive. For the last 12 months *audioXpress* has been nearer 25%. I have cut page counts only about 30 pages per issue because I wanted to give you as readers as much value as possible for your subscription dollars.

Your wonderful comments about the magazine tell me that you are generally happy with the periodical. So happy, in fact, that you are sending us many, many more good letters than we have the space to publish. The pages we cannot afford to include each month are not there because of the low advertising participation levels.

We believe we are delivering a wonderful audio technology market to advertisers. In English, the publication is unique. It is also growing in circulation each month. The quality of what authors are writing for these pages cannot be excelled anywhere. So we have great hopes that Wostrel and company can turn us around in the one department where *audioXpress* is vulnerable.

I wonder whether you as a reader know how much you depend on advertising, how much you influence the advertiser's decision to use our medium, or how much your choice of vendor for your audio purchases affects the size and scope of *audioXpress*.

Specialty vendors in audio are vital to your success in building and modifying equipment. You could say that there is at least a level of interdependence which, unless it flourishes, both readers and vendors lose. In nearly every article where construction is called for I encourage authors to include full information about their suppliers. After all, you come first with us, whether the businesses mentioned choose to advertise in our pages or not. But I hope you will become sensitized to those businesses who *are* helping to bring you this magazine. Without them, you would not have this publication.

On the other side, I hope you will note how many companies we mention as sources do *not* advertise with us. We have, in fact, listed some large multi-billion dollar corporations as sources regularly and often without the slightest show of interest or even respect when we approach them about advertising in our pages. One of the most popular of them, mentioned as a source for solid state and other hardware, told us scornfully that they were "...not interested in small orders." It is as if they are doing the hobbyist a favor by selling to them. This situation affects not only the amateur, however. It makes it quite difficult for authors, and even for those in the engineering departments of many companies, to get the devices they need for development.

The more specialized vendors who do advertise with us are often very responsive to your needs. Some have expanded their offerings just because you and this magazine show an interest in some particular audio idea. They are, if they are good at their jobs, very interested in what your buying choices are. Any interaction you choose to undertake with them will, I imagine, be welcome. Unless, of course, you are one of those mindless types who call a dealer and take up an hour of his time with chat about your latest project, and crow about having bought all your parts at a surplus shop.

I suppose this discussion all adds up to asking you to be thoughtful about what and where you purchase. If you saw the vendor's name and website in an article in *audioXpress*, it will benefit your magazine to tell the vendor so. If you buy from a company who offers you excellent parts and service, but who are not listed in our ad index, it would help if you ask them why they aren't showing their wares in *audioXpress*.

We live in an incredibly interdependent world. We don't always realize the extent to which what we choose affects others—and how what others we will never know personally choose affects us. *audioXpress* exists primarily to support the audio technology of its readers. As a group we like building stuff which enables us to hear the most beautifully and accurately reproduced sound possible. Your sensitivity about how authors, readers, vendors, and this publication all interdepend on each other will give us a better avocation, and a better magazine.—**E.T.D.**

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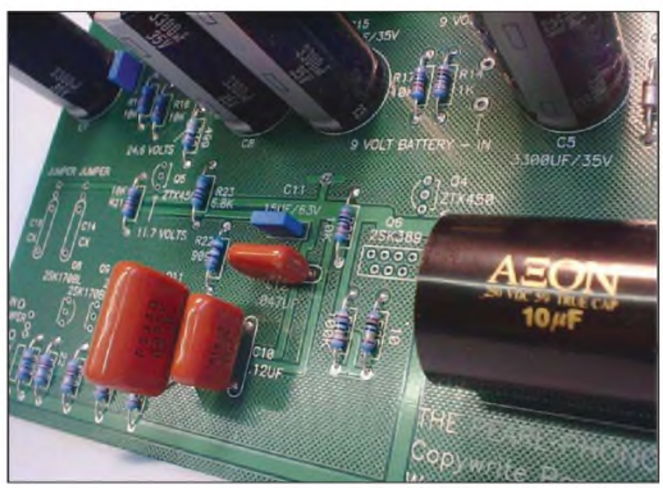

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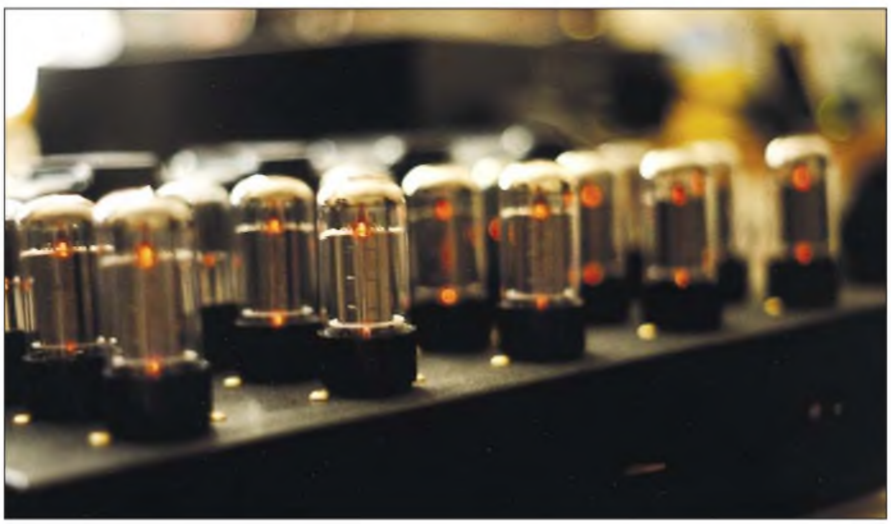

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The peculiar evil of silencing the expression of an opinion is, that it is robbing the human race; posterity as well as the existing generation; those who dissent from the opinion, still more than those who hold it.

JOHN STUART MILL

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A Valve Preamplifier, Part 1

This article (reprinted with permission from *Elektor Electronics*, June 2000, pp. 41–47) presents a valve preamplifier that unquestionably belongs to the high end. **By G. Haas**

In the high-end region, valve amplifiers enjoy uninterrupted popularity. Although a lot of things can be done with modern semiconductors, the old-fashioned valve is still very much in the news in elevated audio circles. In both domestic and studio equipment, valves are being used increasingly often in equipment such as compressors, equalizers, simple amplifiers, filters, and the like.

The objective is to achieve a warmer, more attractive sound than what you can obtain using only sterile semiconductor technology. The digital era, in particular, gives many recordings an unpleasant sharpness, which can be moderated by the knowledgeable and consistent use of valve technology. The valve preamplifier described here usually makes CDs sound more pleasant than they otherwise would.

This preamplifier is a no-compromise design. Only valves are used in the signal path, while semiconductors are used for the auxiliary functions. In this way, the two technologies complement each other. To be consistent, we have also avoided using semiconductors for switching in the signal path.

THE VALVE DETERMINES THE DESIGN

The choice of the amplification elements—the valves—largely determines the topology of the circuit. Given the limited selection of suitable valves, a few basic designs have come to predominate. However, all these have certain significant disadvantages.

A typical preamplifier, for example, is fitted with ECC81, ECC82, ECC83, ECC88, or similar valves. The ECC83 has a high no-load gain, but only a small working current (1 to 1.5mA). The ECC81 and ECC82 have lower gain, but they can be operated at currents up to 10mA. The ECC88 (or the equivalent type PCC88) is most commonly used in television sets, but it is also popular for low-frequency applications, since it can work at currents of up to 15mA, with an operating voltage of only 90V.

Connecting two double triodes in series, however, provides far more gain than is needed nowadays, while still not providing an adequate amount of current. In order to reduce the output resistance, a cathode-follower circuit is frequently used. This significantly diminishes the dynamic output resistance, although it does not completely eliminate it.

A typical cathode-follower circuit using an ECC83 is shown in *Fig. 1*. The dynamic output resistance R_d is given by the formula:

$$R_d = \frac{R_i \times R_k}{R_i + R_k \times (\mu + 1)}$$

With the following typical values for the ECC83, this yields the following value for R_d:

no-load gain:	$\mu = 100$
internal resistance:	$R_i = 62.5\text{k}\Omega$
anode current:	$I_a = 1\text{mA}$
cathode resistance:	$R_k = (47 + 1.5)\ \text{k}\Omega$

$$R_d = \frac{64.5 \times 48.5}{62.5 + 48.5 \times (100 + 1)}$$

This appears to yield a low output impedance. However, in actual fact the total cathode resistance (48.5kΩ) or the internal resistance of the valve will determine the effective output resistance of this arrangement (strictly speaking, only if it is overdriven).

If you assume that the net capacitance of the circuit and the output cable is only 500pF—which is easily possible with cables that are a few meters long—the attenuation at 20kHz is 14dB. This is why you often see recommendations to limit the cable length to 1.5m and to use the lowest-capacitance cable that is available. This certainly helps to reduce the problem, but it does not eliminate its source. The sound of the system will be audibly different when different types of cable are used, and this is a natural consequence of the construction of the amplifier.

PHOTO 1: **Valve preamplifier featuring the ECL86 triode-pentode.**

Using a single valve for both stereo channels is a mistake that has been inherited from the early days of stereo hi-fi technology, and which has proven to be almost impossible to eradicate. The channel separation suffers, due to capacitive crosstalk within the valve and in wiring of the valve socket, and this considerably impairs the spaciousness and detail resolution of the sound.

Valve amplifiers are often operated without negative feedback. This may not have caused any problems in the days of monophonic sound, but the only way to guarantee proper stereo reproduction is to use strong feedback to achieve equal performance in the two channels of a stereo system, regardless of tolerance variations in the characteristics of the valves. This also causes the distortion factor to remain very low, even when the amplifier is driven hard, and it flattens the frequency response. This also satisfies the demand for the lowest possible distortion in the preamplifier.

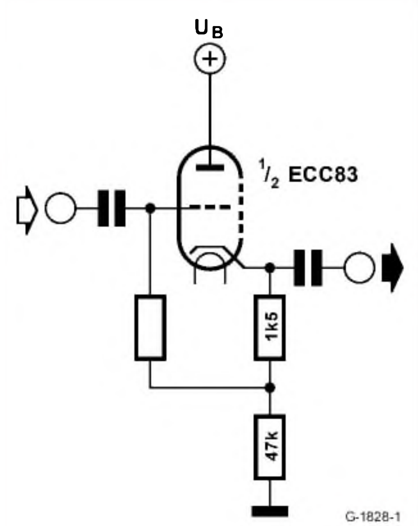


FIGURE 1: A standard valve preamplifier configuration using one ECC83 for the two channels.

For all of these reasons, the preamplifier described here does not follow the conventional path. An ideal amplifier has a high input impedance, a high no-load gain, and a low output impedance. These conditions are easily satisfied by operational amplifiers using semiconductor technology. With valves, the situation is far more difficult.

In order to avoid these design shortcomings, an ECL86 double valve is used here. The triode section of this valve is exactly the same as that of an ECC83. The pentode section can be used as a power amplifier that can deliver 4W with an anode current of 36mA and a distortion factor of 10%. If you properly combine the triode and pentode sections, you can obtain a sort of valve operational amplifier with good characteristics that are similar to those of a modern semiconductor op amp.

MODULAR DESIGN

The block diagram of one channel of the preamplifier is shown in *Fig. 2*. A modular approach is used in the design, with each functional group located on a separate printed circuit board. The amplification chain consists of four elements, which are the input selector circuit, the volume/balance adjustments, the actual preamplifier, and a changeover switch for line or headphone outputs. A protective circuit with a time delay switches the outputs to ground when a fault is detected in the output signal. An external power supply is used.

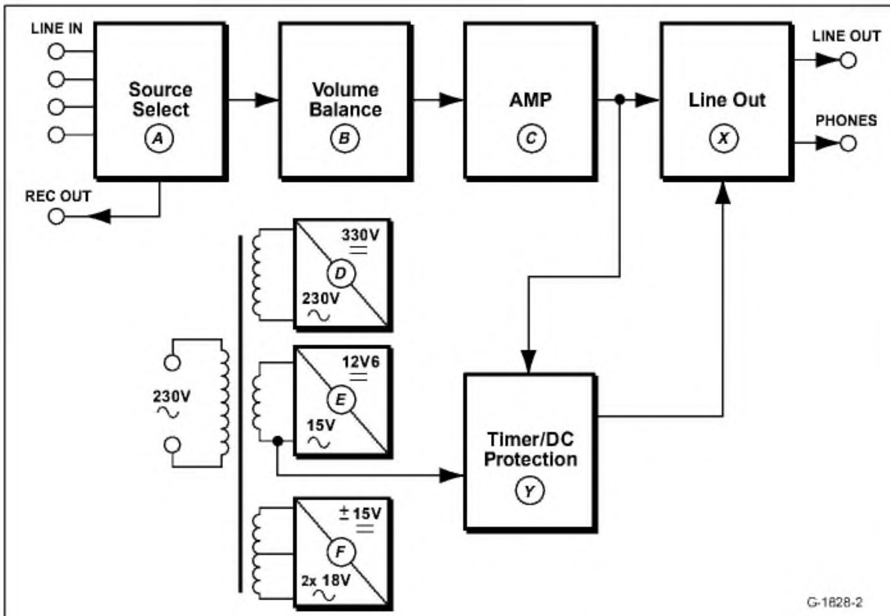


FIGURE 2: Block diagram of the preamplifier, showing its modular design.

The modular construction not only allows the circuit to be modified relatively easily, it also promises very good values for channel separation and the signal-to-noise ratio. This makes up for the increased cost and effort for the wiring.

In the following descriptions of the individual modules, the component numbers correspond to the labels on the circuit boards. The components are not numbered sequentially for the complete circuit, but instead separately for each module. Only one channel of the amplification chain is shown; the component numbers for the second channel are either shown in parentheses or are distinguished by a "prime" mark (').

INPUT SELECTION WITH RELAYS

The input selection module shown in *Fig. 3* is laid out for four signal sources K1 through K4 (K6 through K9) that are connected to a single common line via the single-in-line reed relays Re1 through Re4 (Re5 through Re8). The consistent use of screening and the split layout of the printed circuit board provide a high degree of channel separation from the very start.

The 100kΩ resistors R1 through R4 (R5 through R8) terminate the input sockets and conduct any static charges to ground. They prevent clicks when a different input source is selected or the input cables are reconnected.

The resistors labelled Rx can be used with signal sources that have different output levels; their values can be chosen as necessary. However, you should make sure that the series resistance for K5 (K7) matches the source resistance for the tape recorder output. If the resistance is too large, high frequencies will be lost.

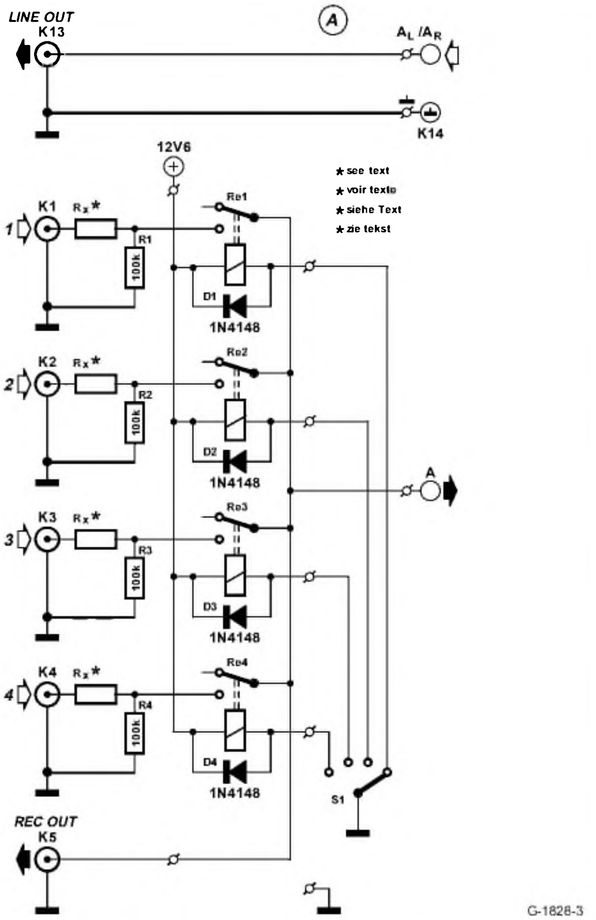


FIGURE 3: The external connections: the schematic diagram of the relay board with the inputs and outputs.

The 12V relays, which are powered by the +12.6V DC filament supply voltage, are switched by the rotary switch S1. This is connected to the circuit board via solder posts.

The line out sockets K13 (K15) and the ground terminals K14 (K16) are located on the same circuit board, but they are fully electrically isolated from the rest of the board. The tape recorder output sockets must be mounted externally, and then connected to solder posts K5 and K7 (ground to K17).

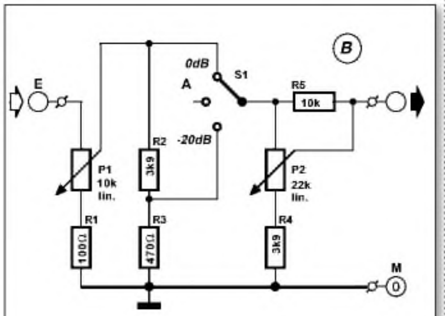


FIGURE 4: Here is where the volume and balance are adjusted.

VOLUME ADJUSTMENT

From connector K5 (K7) of the relay board, the signal is conducted to the circuit shown in *Fig. 4* for adjusting the volume, balance, and muting. The potentiometers that are used, with 41 detent positions, have good reproducibility. The balance adjustment also has a fixed midpoint setting.

The mute switch S1 spreads the adjustment range, so that even low volume levels can be sensitively adjusted. If you find the 20dB step unsuitable, you can modify the values of the resistors. The sum of R2 (3.9kΩ) and Re (470Ω) should remain approximately the same, but you can modify the individual values as desired. When the mute switch is in the middle position, the signal path is interrupted, and the output is muted.

The characteristic curve of the linear volume adjustment potentiometer P1 (which has good tracking due to its linearity) is slightly modified by inserting R1 between its lower terminal and ground. This compensates for the slight imbalance between the two channels in the initial region of the volume adjustment, which arises from the construction of the preamplifier. Of course, with such an arrangement it is not possible to set the volume level fully to zero.

The balance control P2, in combination with R4 and R5, has an adjustment range from around +3dB to –4dB. This is fully adequate for correcting the midpoint. With the indicated component values and the 100kΩ resistors on the relay board, at the middle position of

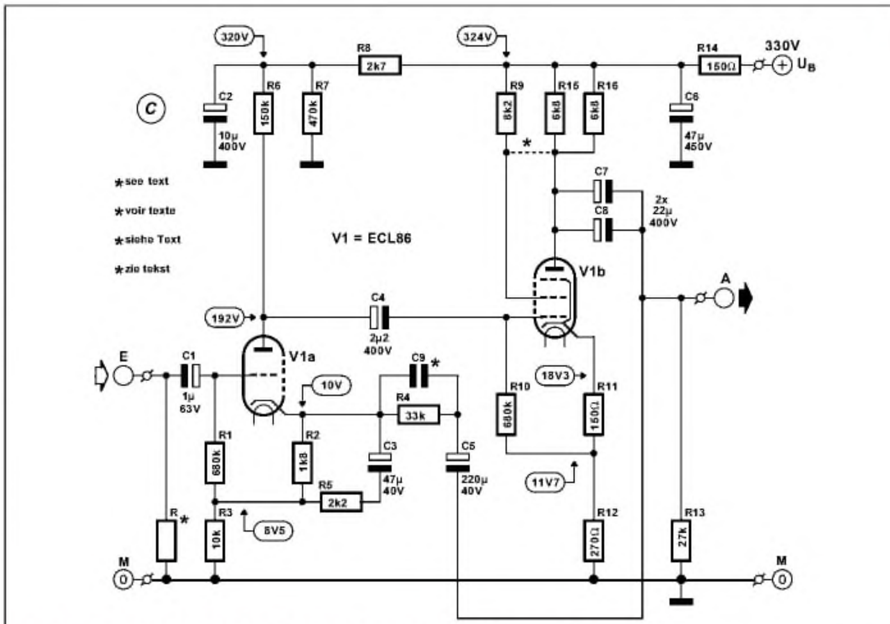


FIGURE 5: The highlight of the design is the valve amplifier stage.

the balance control the input resistance is approximately 2.2kΩ. This is a suitable value for all modern signal sources, and it is low enough to keep the disturbance sensitivity and the total noise at low levels.

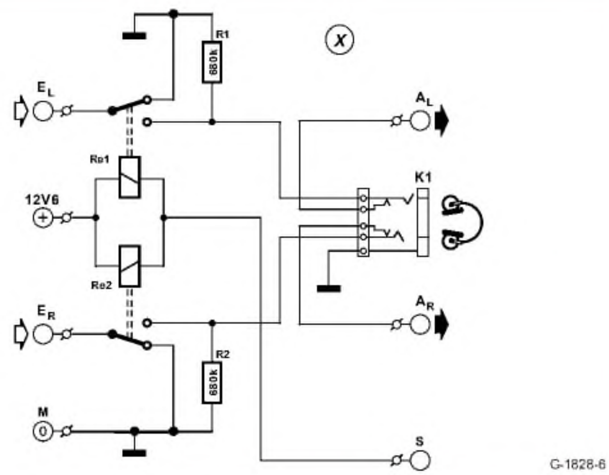
THE AMPLIFIER

The amplifier module shown in *Fig. 5* makes a familiar impression. Valve V1a is the triode section of the ECL86, equivalent to half of an ECC83. The DC operating point is set to just below 1mA by R2. The amplified signal reaches the pentode section V1b via capacitor C4.

Neither section of the valve is wired in the conventional manner. The cathode and the grid bleeder resistor are not connected directly to ground, as is usual, but instead via resistors R12 and R3, respectively. This provides local negative feedback for each valve section. This constrains the valves to work at well-defined operating points, so it is not necessary to use selected valves.

The load resistance of V1b consists of R15 and R16 in parallel (each 6.8kΩ, 4.5W). This divides the not insignificant power dissipation equally over two component packages. The signal is coupled out via two parallel 22µF capacitors (C7 and C8). Using two capacitors instead of a single 47µF capacitor halves the ESR (effective series-resonant resistance) of the output capacitor, which benefits the high-frequency transfer characteristic of the amplifier. Resistor R13 bleeds off static charges.

FIGURE 6: The headphone and line output connections.



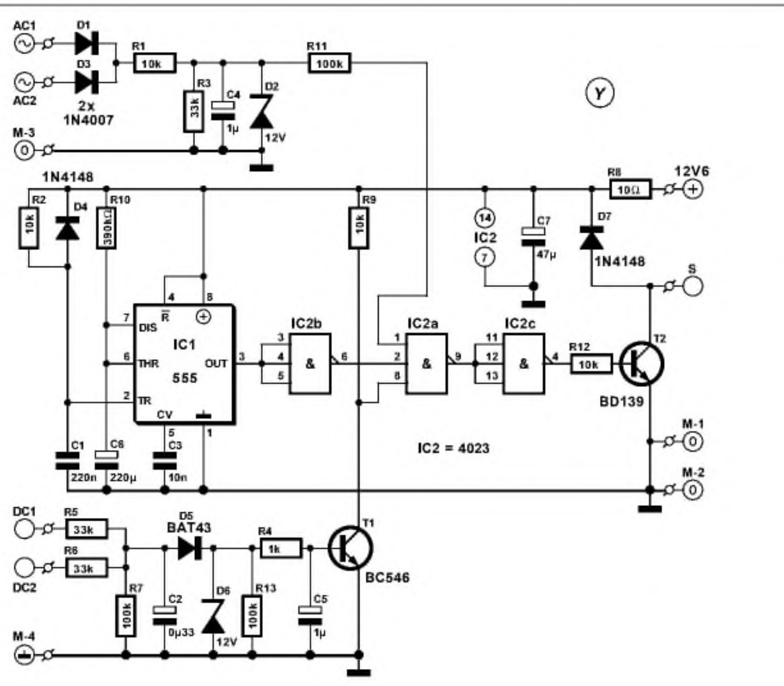


FIGURE 7: The protective circuitry monitors the state of the preamplifier and switches the output relays.

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To ensure that the circuit has a well-defined behavior, the output signal is coupled back via C5 and R4 in negative feedback. The basic amplification of V1a depends on the load resistance of R6, the operating voltage (the voltage across C2), and the cathode circuitry. Resistors R2 and R5 are connected in parallel for AC signals. The amplification is thus determined by the relationship of the parallel combination of R2 and R5 to the resistance of R4.

If you wish to modify the basic amplification, you can change the values of R5 or R4. The smaller the resistance of R4, or the larger the resistance of R5, the higher the gain. Capacitor C5 separates the AC and DC currents, so that the DC operating point of V1a does not shift. Resistor R2 may not be modified to change the gain, because this would shift the DC operating point in an unacceptable manner.

The supply voltage, U_B, is 330V. It is decoupled and filtered by R14 and C6, in order to avoid channel crosstalk via the supply voltage. The supply voltage is further decoupled and filtered for V1a by R8 and C2. Resistor R7 is a bleeder resistor that discharges the high-voltage electrolytic capacitors after the supply voltage has been switched off.

OPTIONS

The input resistor R can be omitted, since the volume and balance adjustments also shunt static charges. However, if the amplifier module is used in some other application, any desired value can be used as necessary. The indicated connection between R9 and the anode of R1b should only be used if you want to operate the pentode in quasi-triode mode. In this case the combination of R15 and R16 is shorted out, and R9 is naturally redundant.

LINE OR HEADPHONES

Using the ECL86 as a preamplifier valve has the additional advantage that the pentode section can deliver 4W of output power. This capability can be used to advantage by connecting a switched phone socket (K1) between the amplifier output and the line output.

When a headphone plug is inserted into this socket, it automatically disables the line outputs. The pentode section of the ECL86, which can deliver a relatively large current, can easily drive headphones with an impedance greater than 300Ω—even over a relatively long screened cable—without degradation of the sound quality. Normal screened cables are fully adequate for this purpose.

The portion of the circuit shown in *Fig. 6* also includes two relays, which are driven by the protective circuitry. The two resistors bleed static charges to ground.

UNCONDITIONAL PROTECTION

The protective circuitry shown in *Fig. 7* performs several tasks at the same time. Three conditions must be satisfied before the preamplifier is switched "online."

When the amplifier is first switched on, the outputs are initially shorted out via the relay outputs. The timer output of IC1 (pin 3) goes low after approximately 100s, at which time the valves should be properly warmed up and all charging processes completed. This delay prevents humming, burbling, or other upsetting sounds from being generated by the speakers while the preamplifier is heating up. The output of inverter IC2b then sets input 2 of NAND gate IC2a high.

If a voltage is then present at pins AC1 and AC2, which are connected directly to the filament winding—diodes D1 and D3 monitor both half cycles of the AC waveform—then C1 is charged via R1 to 12V (limited by D2). A high level is thus applied to input 1 of IC2a.

The DC components of the two output signals are monitored via connections DC1 and DC2. A DC voltage can, for example, be present if an output coupling capacitor breaks down. The AC components are shorted out by C2, which is discharged by R7. This combination also determines the time constant of the monitoring circuit.

If a DC voltage higher than around 1.3V is present, transistor T1 is driven via D5 and R4. This pulls input 8 of gate IC2a low. A Schottky diode with a very low forward voltage drop is used to ensure that the monitor circuit will respond, even with very low values of the DC component. Transistor T1 is cut off only when no significant DC component is present, so that R9 can pull input 8 of gate IC2a high.

Resistor R13 ensures that T1 is securely cut off in this situation. Diode D6 limits the maximum voltage applied to the base of T1, while R4 and C5 provide a short time delay, so that the circuit does not immediately respond to every tiny disturbance.

Only when all three of these conditions have been satisfied—so that all three inputs of the AND gate are high—will the protective circuit enable the output relays via inverter IC2c and the driver transistor T2.

When the mains transformer is switched off, the level at input 1 of the gate goes low almost immediately. This causes the relays to be immediately disabled, so that no disturbance signals can be passed on to the following equipment while the capacitors in the preamplifier are discharging. The reset circuit consisting of R2, D4, and C1 also resets timer IC1 after a brief interruption of the mains power, so that the full delay time must expire each time.

The delay time can be changed by modifying the values of R10 and C6. The time is equal to 1.1 × RC. The protective circuitry is operated from the filament voltage power supply, with a voltage of 12.6V.

ONE TRANSFORMER, THREE POWER SUPPLIES

A good power supply is essential for the proper operation of a preamplifier. Since the amplifier draws relatively little current, you can build a cleanly filtered and stabilized power supply economically using modern semiconductors. *Figure 8* shows the schematic diagram of the entire power-supply circuit. This is also built using several separate circuit boards, which are indicated by dashed outlines.

The preamplifier needs a high voltage for the valves, as well as a low DC voltage for the filaments, the relays, and the protective circuitry. These voltages, as well as some others, are provided by a single mains transformer (NTR-10B) that is protected by a 0.4A slow-blow fuse on the primary side. This transformer is made using grain-oriented 0.35mm steel laminations with especially low dispersion and losses, which are usually used for high-quality low-frequency coupling transformers.

A very orderly winding technique and vacuum impregnation, which is not possible with toroidal transformers, provide long-term stability and corrosion resistance. The impregnating resin penetrates into the coils and permanently anchors each individual winding. Electrical safety is provided by a 4000V breakdown test between the primary and secondary windings and a static screen that is connected to the protective ground lead. Obviously, you will not find this sort of high-end transformer in any electronics supermarket. It is only available from Experience Electronics, my firm at experience.electronics@t-online.de.

You can see the high-voltage section at the top of the schematic diagram. This circuit filters the hum voltage to a level that is below the intrinsic noise level. Resistor R1 and diodes D1 through D3 generate a good reference voltage with a value of around 330V.

Series regulator transistor T1 is a high-power V-FET (type BUZ92) in a TO-220 package. This allows you to construct the high-voltage power supply in a compact manner, using a small heatsink. You mount the heatsink on the printed circuit board, which makes for short leads in the high-voltage path. Resistors R4 and R5 and transistor T2 provide current limiting.

With the indicated component values, this is set to around 90mA. Resistor R6 discharges the electrolytic ca-

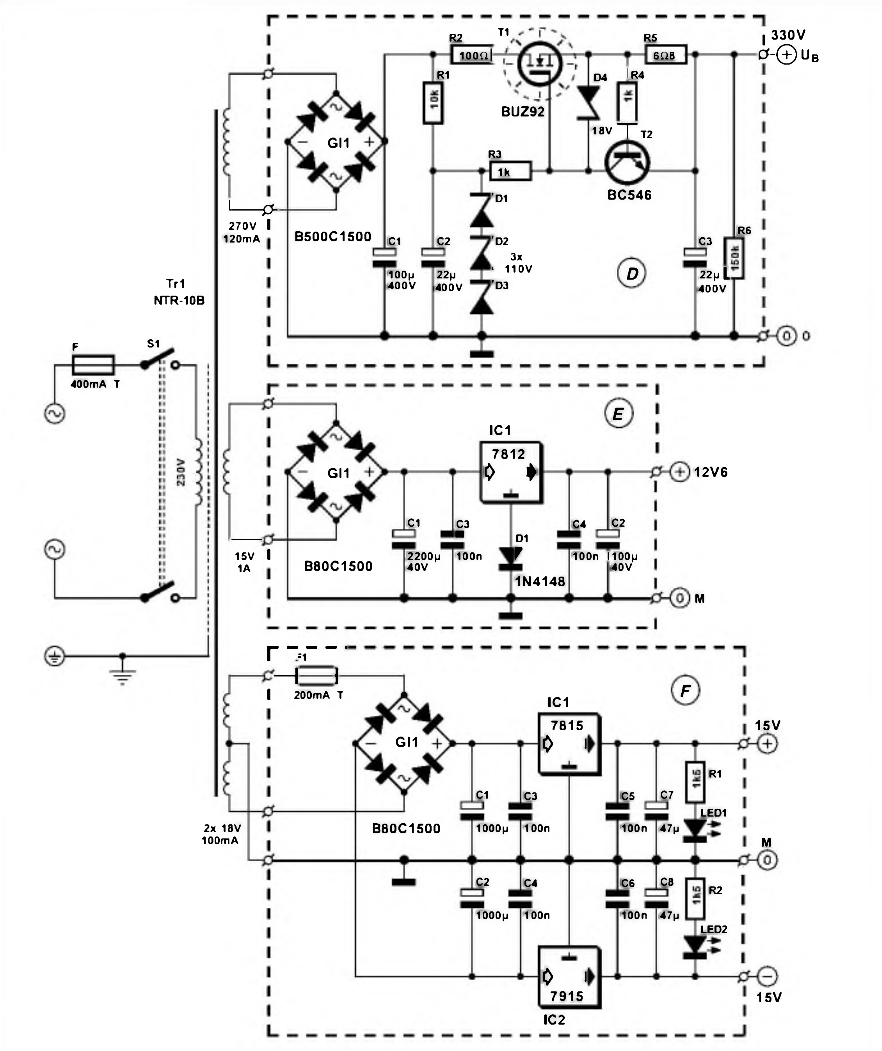

FIGURE 8: The power supply is a combination of a high voltage supply, a low voltage supply, and an optional third supply.

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CIRCUIT ADDENDUM

The dashed connection between R9 and the anode of valve V1b is not an optional wire link. As indicated by the PCB component mounting plan, resistor R9 may be fitted in one of two positions marked R9 and R9*. In pentode mode (default), the resistor is fitted in position R9, i.e., connecting the suppressor grid (g2) to the +324V anode voltage. This configuration is shown in the circuit diagram, *Fig. 5.* The dashed connection has no meaning.

The quasi-pentode configuration requires the resistor to be fitted in position R9*; i.e., between the suppressor grid and the anode.

Figure 6 does not show two back-emf suppression diodes across the relay coils. These diodes are, however, present on the board, as well as included in the parts list.

pacitors after the power is switched off. Zener diode D4 limits the gate voltage of T1, which must not exceed 20V.

In order to avoid hum voltages, the filaments are powered by a DC supply. The frequently heard assertion that using a DC current for the filaments damages the valves is simply nonsense. All that is necessary is to bring the filaments to a particular temperature, so that the cathode can emit enough electrons. Whether this temperature is achieved using an AC current or a DC current does not matter.

The "odd" value of 6.3V for the filament voltage, by the way, comes from the early days of valve technology, when the filaments were powered by four carbon-zinc batteries. Since fresh batteries have a terminal voltage of more than 1.5V, a margin of 0.3V was chosen to prevent the filaments of the (at that time) very expensive valves from burning out prematurely.

Each of the two ECL86 valve filaments draws 0.66A at 6.3V. To minimize the losses in the filament circuit, the two filaments are connected in series. A 12V voltage regulator connected to ground via a 1N4148 forms a simple but good 12.6V power supply for the filaments. The relays for the input selector are also powered by this supply. The aluminum mounting plate is used as a heatsink for the voltage regulator, which must be mounted using an insulator.

At the bottom of the circuit diagram, you can see a third power supply that provides symmetrical ±15V. This supply is optional; you can use it to power external devices, such as an equalization preamplifier, from a non-grounded supply.

If such a supply is already present in the preamplifier, you save the cost of a mains adapter and the associated cabling, and you can switch the external equipment on and off at the same time as the preamplifier. The aluminum mounting bracket is also used as a heatsink for this supply, which must be isolated. The supply voltage has additional protection in the form of two fuses in the transformer leads.

The second part of this article will include the printed circuit board and component layouts, the components list, precise construction information and, of course, performance data. ❖

The DR10a

Another in a series of horns designed to deliver the most bang for your buck. **By Bill Fitzmaurice**

When I designed the original DR10 folded horn (January 2001), my intent was to squeeze the most performance I could from a mid-price 10″ driver. That project succeeded beyond my expectations, delivering unheard-of power and bandwidth from a 7ft^3 cabinet, and doing it at a bargain-basement price.

My next project, the DR12 (July 2001), defined a whole new set of standards for folded horns, giving wideband concert-level 140dB output from a single 12″ driver. Small club needs were covered with the DR8 (November 2001), the most powerful 2ft^3 cabinet ever built. When I finished that project—my third design within a two-month period—I had earned myself a rest. I took two days off before I started in on my next task.

I designed the first three cabinets of the DR series primarily with PA in mind, to cover the gamut of small-, medium-, and large-venue requirements. The full-range DR10 and DR12 also work exceedingly well on bass and keyboards, but are capable of output levels considerably higher than most bass or key players need from a backline (onstage) cabinet.

I designed the next three cabinets of the series as backline bass or key cabs. Made as small as possible for the drivers they use, they are still more powerful than any commercial cabs of similar size. Their model designations are the DR10a, DR12a, and DR15a, using 10, 12, and 15″ drivers, respectively. The first to come off my backyard assembly line was the DR10a (*Photo 1*).

PHOTO 1: The DR 10a.

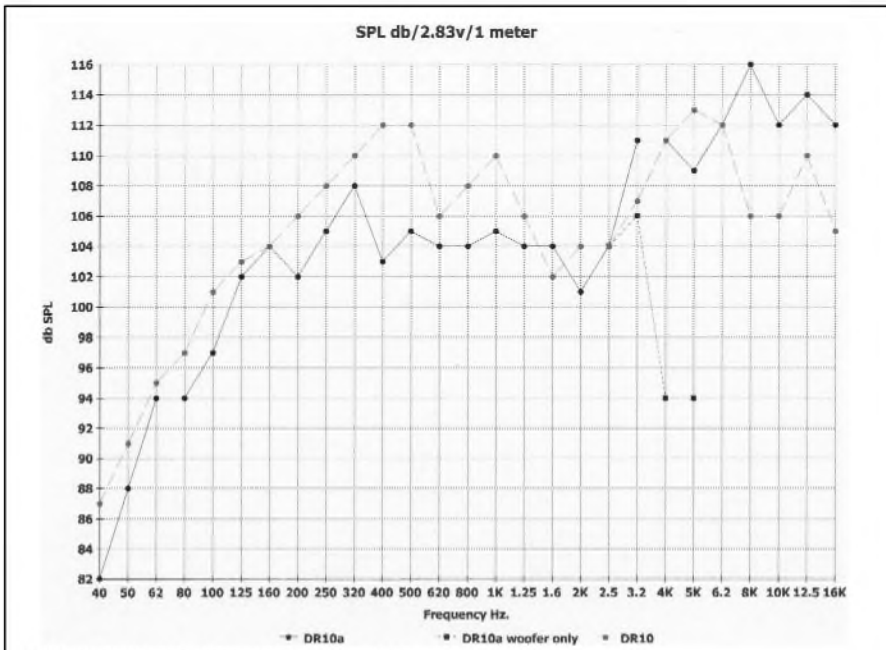

FIGURE 1: DR10/DR10a SPL comparison.

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FEATURES

Small size was not my only requirement for this design. I had been using a "MidRanger" as my main bass cabinet for three years, and while its size was not excessive, its weight was. The main culprit was the driver, an 18 lb EVM-12L™. [For details on the MidRanger and other Fitzmaurice designs, see his book entitled *Loudspeakers for Musicians*, available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-6371, Fax 603-924-9467, E-mail custserv@audioXpress.com.]

For the DR10a I decided to switch to a mid-line ten (a Carvin PS-10™, available at www.carvin.com for about $60), which lopped off 11 lbs. Further weight reductions were achieved by using the thinnest possible materials, far lighter than traditional cabinet designs allow.

The DR10a is an expanded version of the DR8, 30% smaller than the DR10. With a horn both shorter and having a smaller mouth, it consequently does not have the large club main PA capabilities of its bigger brother. On the other hand, it is perfect for backline bass and keyboard for bands who run at moderate stage levels, and will handle PA requirements for up to 300 seat rooms, encompassing 95% of the rooms the average garage band will play in.

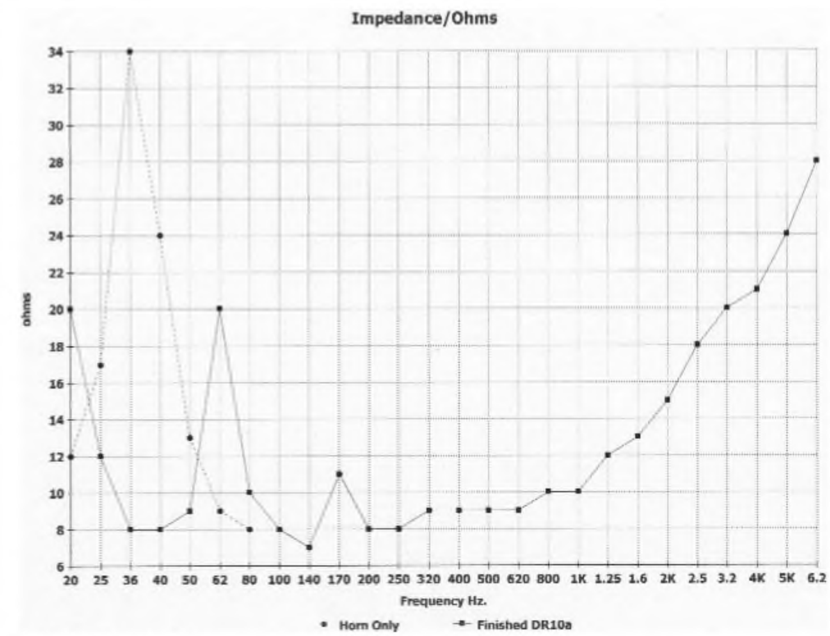

FIGURE 2: Impedance.

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For a comparison with the DR10, look at *Fig. 1*. The DR10 averages 4dB higher SPL than the 10a; 2dB of that differential can be attributed to the larger radiating area of the DR10, while the other 2dB is the result of different driver efficiencies (the Eminence™ woofer in the DR10 has an SPL rating of 98dB/W, while the Carvin™ has a 96dB rating). You can get more power from the DR10a by using a more efficient driver, but don't try to use a premium driver with a 10 lb magnet, such as an EVM10™ or JBL-E110™, because it won't fit. To be safe, use a driver with a magnet weight between 40 and 80 ounces.

Spec wise, the PS-10 has an f_S of 50Hz, which is actually a bit lower than optimal. If you substitute drivers you will do better with an f_S between 60 and 70Hz. Other drivers I would try are the Selenium 10PW3™, with an f_S of 69Hz, and the EV Force 10™, with an f_S of 65Hz. Both have 98dB ratings, and the 10PW3 is reasonably priced at about $60 (available from www.partsexpress.com).

When loaded with the PS-10, the DR10a doesn't suffer woofer rolloff until above 3.2kHz, allowing smaller tweeters running at a higher crossover point than on the DR10. The tweeters on the prototype—two Motorola™ Twin-Bullets model KSN1177—total four elements, giving higher SPL than the three elements on the DR10. (Note: while the KSN1177 is rated nominally at 99dB SPL, this is when wired in series with a 20Ω resistor as recommended by the manufacturer. I have found today's piezos to be very stable without the resistor, and nearly 6dB more sensitive when so wired.) This gives a very strong high-end—definitely not hi-fi, but great for cutting through the muddy acoustics of, say, a gymnasium.

If you want flatter response, use two elements, via either one dual-element driver or two singles. I prefer the higher SPL of four elements, because you can always trim off unwanted high end with EQ, gaining amp headroom in the process.

PERFORMANCE

The throat of the DR10a at about 46 in^2 is larger than that of the DR10, to compensate for the higher f_h (horn flare frequency) of the smaller box. The best combination of low-frequency power and flat response from a folded horn occurs when the ratio of f_h to $f_{S(h)}$ is no more than 3:1. ($f_{S(h)}$ is the resonance of the driver/horn combination.) Thus, the lowest desirable $f_{S(h)}$ for the DR10a (with an f_h of 140Hz) is 46Hz.

This leads to less-than-optimum performance with the PS-10, whose 50Hz f_S results in an $f_{S(h)}$ of 36Hz. A higher $f_{S(h)}$ could have been achieved with this driver via an even larger throat, but at the expense of high-frequency loading.

The net effect of this lower-than-optimum $f_{S(h)}$ is seen on the SPL chart as a "flattening" of the curve at 80Hz. Note how the DR10 curve has no flat spot, as its $f_{S(h)}$ of 40Hz matches nearly perfectly its 125Hz f_h. I would have used a driver other than the PS-10 but for one factor: I already owned one. Figuring that many readers would share this quandary, I decided to proceed with the prototype, and ultimately found the DR10a—even when operating at less than optimal—is still better than any other box of the same size.

To understand the effect of the low f_S of the PS-10, look at *Fig. 2*. On the dotted trace the $f_{S(h)}$ peak is seen at 36Hz. As

PHOTO 2: Assembling the throat horn divider and sides.

PHOTO 3: Attaching throat horn sheathing using clamps and cauls.

PHOTO 4: Completed throat horn. Note how cauls are now braces.

$f_{S(h)}$ determines f_B, the box is tuned to 36Hz, seen on the solid trace as a low 36Hz. Two octaves higher, the beginning of the horn passband, the f_h, is seen as another impedance low at 140Hz. This wide spread between f_B and f_h causes the loss of response at the halfway point between those frequencies.

If the driver had an f_S of 60Hz, the $f_{S(h)}$ would occur close to 48Hz, giving at least an additional 2dB SPL at 80Hz. This would also rob 2dB at 32Hz, but since there is negligible program power there it is a worthwhile trade.

Conversely, a driver f_S of 80Hz would result in $f_{S(h)}$ and f_B of about 70Hz. This would increase SPL at 100Hz, but response below 70Hz would suffer. Thus, an f_S between 55 and 70Hz is best for this application.

CONSTRUCTION

Construction methods depend on your choice of materials. Where the parts being joined are ½″ (or thicker) plywood, fasten with drywall screws, piloted and countersunk, and construction adhesive, preferably urethane based. Fasten parts too thin to accept screws with hot-melt glue. You should not use hot melt where joints are subject to high stress, but it works well for fast setting and gap filling. Be sure to chamfer well the edges of any part that is to be hot-melt glued, to serve as a trough for the glue.

PHOTO 5: Attaching throat horn support; note guideboard and clamps.

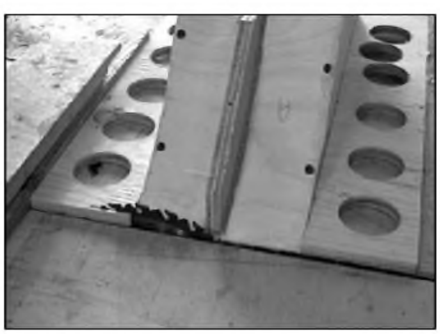

PHOTO 6: Trimming the throat horn and supports on the table saw with panel cutting jig.

All part thicknesses are optional. I used the thinnest plywood possible that would still be structurally sound. If you are not a skilled woodworker you may opt to use thicker materials to make joinery easier. However, the integrated bracing of the design does not require heavy panels.

Note that all listed material thickness is nominal, because no plywood measures its advertised thickness. You must determine the actual size of all parts based upon the materials you are using. Where ⅛″ plywood is specified, Baltic birch is preferable.

For other sizes pine or spruce works best. You may use Luan where ¼″ plywood is specified, but, more than other species, this tends to splinter when cut, and with two thin veneer sheets over a thick core it does not bend as well as plywood with three equally thick laminations.

PHOTO 7: Attaching throat horn assembly to cabinet top/bottom.

PHOTO 8: Ready for inserting the baffle.

When laying out bending parts, always flex the sheet to determine its easier bending axis. When cutting out two identical parts, it is best to rough-cut them oversize, screw the pieces together, and then finish-cut the pieces simultaneously for a perfect match. Remember to fill all the screw holes later.

You cannot accurately cut panels without a panel-cutting jig and a table saw, so make sure you have (or have access to) both. A straight guideboard is also a necessity for joinery; you can make one by laminating a 1½″ square piece of plywood. To secure a joint, first drill pilot holes through the joint line.

PHOTO 9: Determining the baffle hole location.

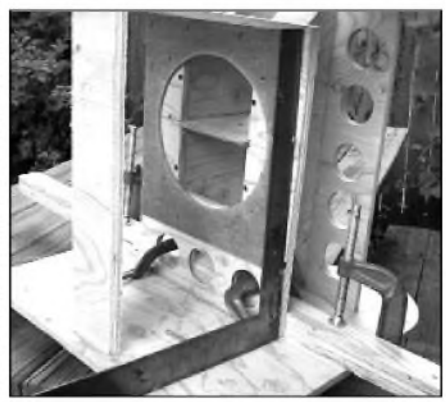

PHOTO 10: Checking parts alignment before assembly.

PHOTO 11: Assembly after installing horn braces.

Then clamp the guideboard to the outside of the joint line. Clamp the mating part to the guideboard.

Once you've confirmed that all is perfect, take the mating port off, apply adhesive, and clamp it back in place. Now use a pilot/counter bore bit to drill through the previously drilled holes and finally drive the screws. For screwing through ½″ plywood use 1¼″ screws; for thinner materials use 1″.

COVER-UP

Critical to the best performance from this, or any cabinet, is a thorough job of stuffing and/or lining the cabinet. Any sound waves reflected back to the driver cone from any surface within the rear chamber, even its own frame and magnet, will disrupt frequency response. The best method for the elimination of reflections is to cover all interior surfaces, including the surfaces of the ducts and the drivers, with absorbent pads such as those used in auto sound. That option is both costly and difficult, but if you want the best performance it's the way to go.

Option two is to do the same with egg crate foam—not so costly, but again, a time-consuming and difficult job. My preference is to stuff the enclosure with poly-fill. Stuff it loosely into every reachable void within the cabinet, while being sure to leave both the rear of the driver cone and the entrances to the ducts unimpeded. The last option is to use a combination of all three methods—using pads on driver parts, foam where you can easily glue it in place, and poly-fill everywhere else.

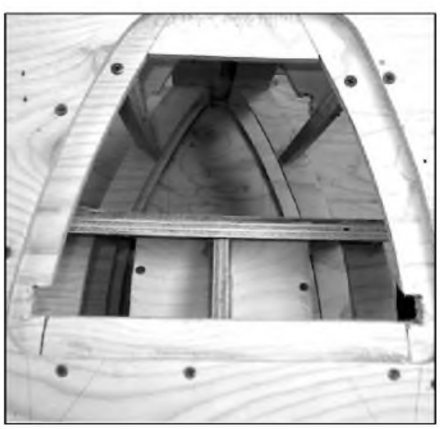

PHOTO 12: The flanged porthole opening. Note trimming to accommodate woofer frame.

Construction starts by cutting the throat horn's sides from ⅛″ plywood (*Fig. 3*), chamfering heavily the curved edges. Save the scraps for later use. Cut the throat horn divider from ¼″, ⅜″, or ½″ plywood, chamfering all edges. Hot-melt it to the first one and then the other throat horn side, offsetting it to one side of the centerline, so the halves of the throat horn are slightly different in size (and reflections within the two halves will occur at different frequencies to minimize aberrations).

The divider should extend ½″ past the sides on the driver end. Use a 5″ wide piece of scrap and clamps to hold all parts square and true while the glue sets (*Photo 2*).

PHOTO 13: Attaching the mouth horn sheathing.

Use the scraps from the horn sides to build clamping cauls that perfectly match the curvature of the horn. Cut out the throat horn sheaths from ⅛″ plywood; when installed these should overhang either end at least ½″. Using a caul, clamp a sheath to the assembly (*Photo 3*).

ASSEMBLY

When you have made sure things are square and true, hot-melt the sheath to the side, inside and out, being careful not to glue the caul. Repeat the process on the other sheath. You may drive a

PHOTO 14: Checking duct hole for proper fit.

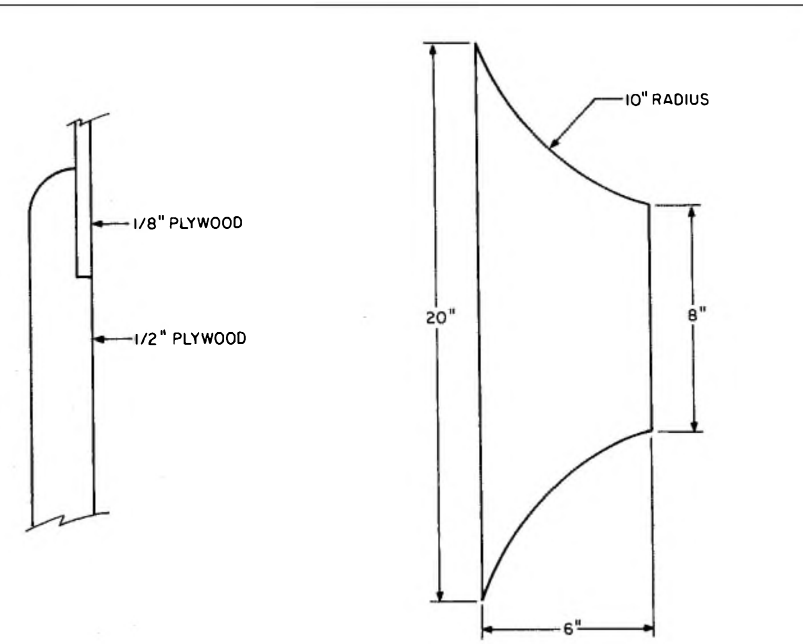

FIGURE 3: Throat horn sides, side view.

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screw or two through the sheaths to stabilize things (very precisely into the center of the sides). Disassemble the gluing cauls, using two of the scrap pieces as stiffeners glued to the outside of the sheaths (*Photo 4*).

Cut the horn supports from ½″ plywood, making them a bit longer than required. Use a hole saw to drill random holes through them, reducing reflection area within the cabinet. Attach the supports to the horn assembly, clamping the parts together to hold them in alignment until the screws are in place (*Photo 5*). Run the completed assembly through the table saw, atop a panel-cutting jig, to both trim the assembly to finished size and true all parts in one step (*Photo 6*). At the throat end use a belt sander to sand the sheaths flush.

Cut the top and bottom (*Fig. 4*) from ½″ plywood. Draw on them the joint lines for all intersecting parts; on the bottom draw the cut line for the porthole. Use a saber saw to cut the porthole out of the bottom, starting with a plunge cut. Attach the throat horn assembly to the top (*Photo 7*).

Cut the tweeter baffle from ½″ plywood, with the edges cut at a 40° angle. Cut it from the center of a piece of plywood about 6″ wide so that you have left over two 1½″ slivers square on one edge, 40° on the other. Attach the tweeter baffle to the top, and then the bottom to the entire assembly (*Photo 8*).

Cut the baffle from ½″ plywood, fashioning a notch in it so that it can be held flush to the throat horn without interference from the protruding throat divider (*Photo 9*). Trace the throat hole location on the baffle and cut it out. Make a spacer from ⅛″ or ¼″ plywood to fit between the driver and the baffle, to prevent cone slap in long excursions.

Attach the spacer to the baffle. Drill mounting holes through the baffle and insert them with T-nuts; four bolts will be sufficient, aligned so that the bolts are away from the baffle edge. Drill holes through the baffle, below the driver-mounting zone, to minimize internal reflecting surface. Attach the baffle to the assembly, making sure to true the cabinet (*Photo 10*).

Cut the horn braces from ½″ plywood or ¾″ stock. The braces attached to the bottom are installed whole, forming the majority of the porthole flange. Use scrap materials to complete the flange. The remaining braces are bisected to accommodate the baffle, and bracing is not extended from the baffle to the tweeter baffle near the porthole.

Temporarily place the driver in position when attaching the braces to be sure that they do not interfere either with the driver frame or accessing the mounting bolts. Use the scraps from

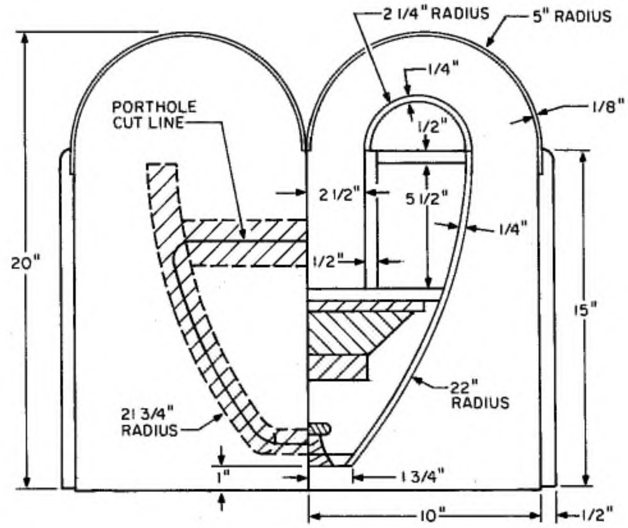

FIGURE 4: DR10a top/bottom view. Right half of view shows location of horn braces, port hole flanges, and port hole cut line. Left half of view shows interior parts. Note, not all parts are shown, see text for details.

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PHOTO 15: Ready to install side braces.

PHOTO 16: Holding back braces in place while hot-melt glue sets.

PHOTO 17: Using a jig to secure PVC pipe for cutting.

PHOTO 18: PVC installed.

cutting the tweeter baffle to build it out to double thickness (*Photo 11*). Trial-fit the driver through the porthole, trimming the flange as necessary for clearance. Make sure that the driver will fit now, because you won't be able to fix it later (*Photo 12*).

Using ¼″ plywood, cut the mouth horn sheaths. Attach them starting at the tweeter baffle, with screws every 3″ or so (*Photo 13*). Hitting the braces and baffle will involve drilling a few "exploratory" holes; don't forget to fill them when done.

After the adhesive has set, sand the sheaths flush to the tweeter baffle and the braces. The ducts are either 1¼″ or 1½″ FVC pipe or electrical conduit. Cut a sliver of it at a 60° angle to trace the duct holes on the sheaths, cutting them out with a saber saw, and chamfering them with a drum sander for proper fit (*Photo 14*). Cut the holes for the tweeters, spread just far enough apart that they do not hit the woofer magnet (*Photo 15*).

Cut the back braces from the same material as the horn divider, chamfering all edges. Using a clamp and two pieces of scrap, clamp them in place and use hot melt to fix them in place (*Photo 16*). Using a table saw, slice a 20″ piece of 4″ schedule 40 FVC pipe in half lengthwise, first clamping it to a simple jig (*Photo 17*). An abrasive blade is preferred, and do not cut it all the way through, because it will close on the blade.

Finish the job with a utility knife or handsaw. Cut the halves into four pieces to just fit on the back of the horn supports, between the back braces and the top and bottom. Chamfer the edges of the FVC, fill them with poly-fill, and hot-melt them in place, making sure of airtight joints (*Photo 18*).

Cut the side braces from the same stock as the horn divider and back braces, cutting them at least ½″ too wide. Temporarily clamp them in place and use a pencil, with a spacer if necessary, to trace the true shape of the mouth horn onto them (*Photo 19*). Cut the curved edges to final shape, clamp

PHOTO 19: Marking final cut line on a side brace.

PHOTO 20: All bracing installed.

PHOTO 21: Using clamp as a "third hand" while installing back half.

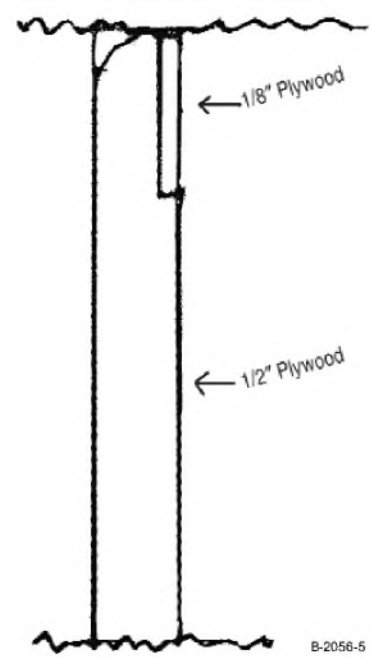

FIGURE 5: Joint detail, back and sides.

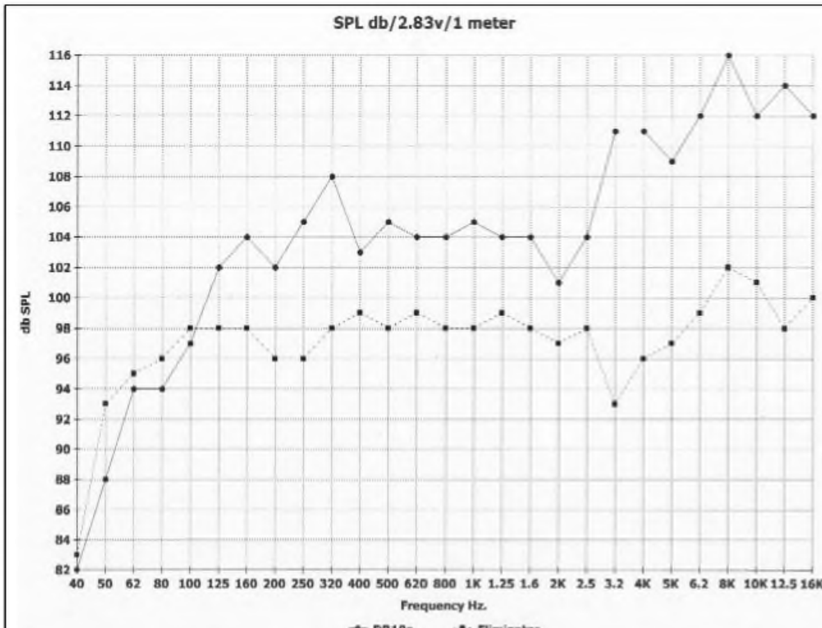

FIGURE 6: DR10a/EV Eliminator SPL comparison.

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the braces in place again, and use a straightedge spanning from the top to the bottom to mark the true edges of the braces.

Trim the braces to final shape and attach them to the assembly (*Photo 20*). If you used ½″ plywood for the braces, you may use adhesive and screw them to the assembly, working from inside the horn. For thinner materials use hot melt.

The back halves are ⅛″ plywood, cut to overhang the top and bottom about ¼″, and to extend beyond the curvature of the back at least 1″. Insert both pieces into the back groove, using a long clamp to pull one half out of the way while you attach the other (*Photo 21*). Use adhesive and screws every 3″ to attach the back to the top and bottom, and to the back and side braces if you made them from ½″ plywood. For thinner braces use hot melt, gluing as you go after each pair of screws.

The ⅛″ plywood flexes easily enough to be bent into position by hand, but you may also use a webbing clamp to pull them into place. The final few inches of the halves will bow away from the back braces, so use long clamps, guideboards, and plywood scraps to pull them into place for gluing (*Photo 22*). When the adhesive has set, flip the cabinet on its face and use the hot-melt glue gun to liberally fill the joint between the two back halves.

DRIVER AND DUCT INSTALLATION

The sides are made of ½″ plywood, with the trailing edge routed or sawn with a rabbet to overlay the back halves (*Fig. 5*). The sides are attached in the same fashion as the back halves, again using clamps and cauls to pull the joint tight with the back halves until the adhesive sets. When the adhesive has set, sand all the cabinet edges flush, rounding them off if you wish.

Vacuum the box and install the woofer, working through the tweeter holes to drive the attaching bolts. If you wish to tune the box, test the woofer now. Start charting impedance in the vicinity of 50Hz; the impedance will peak at the box $f_{S(h)}$.

Now use plywood scraps, weatherstripped for an airtight seal, to close off the tweeter holes. Install your jack, either through the porthole cover or on the side (which will require a hole through the sheath for the wire, hotmelted airtight). If you want to use a speaker stand top hat, install it through the porthole, caulking it airtight. Rim the porthole flange with heavy neoprene weather stripping, drill pilot holes, and install the porthole.

You can cut the ducts square, or cut the protruding ends at a 60° angle and sand them to the contour of the sheathing, resembling submarine torpedo tubes (*Photo 23*). If you go for the smoother look, remember that the angled parts of the ducts do not count towards their effective lengths. Using angled cuts, the two lower ducts are limited to effective (inside the cabinet) lengths of 8″, while the upper ducts are limited to about 3″.

Using 1½″ PVC, two 8″ and two 3″ ducts will give an f_B of about 44Hz. To get the required f_B of 36Hz on the prototype, I used two 8″ ducts and one 3″ duct, with the fourth duct closed at one end, relegated to cosmetic duty only. Somewhat shorter lengths will be required using 1¼″ FVC or conduit. Conduit is harder than schedule 40 pipe, so it sands better, and its gray color elimi-

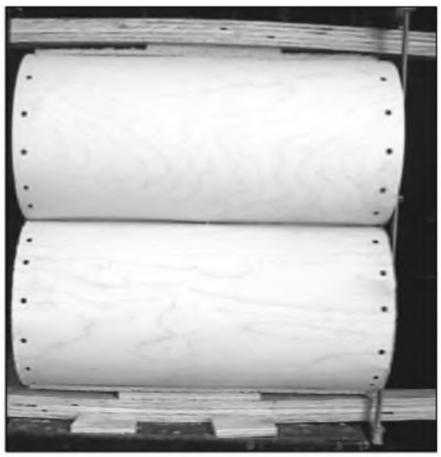

PHOTO 22: Using clamps and cauls to hold back/side joints while adhesive sets.

PHOTO 23: "Torpedo tube" duct on finished cabinet.

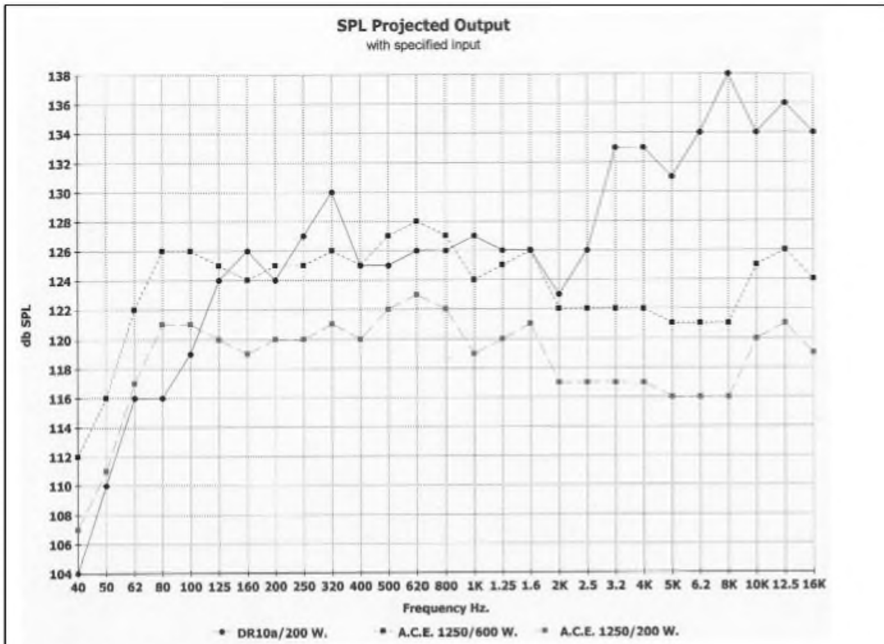

FIGURE 7: DR10a/A.C.E. 1250 SPL comparison.

B-2056-7

nates the need for paint. Lightly glue the ducts into place while running an impedance test on the driver until you find the length combination which gives an impedance low (the f_B) where the $f_{S(h)}$ peak previously occurred.

After determining duct lengths, remove the porthole cover, the tweeter hole covers, and the woofer. Permanently install the ducts with hot melt, gluing from both the exterior and the interior of the cabinet. Sand the ducts flush (if desired) and apply your finish of choice. Lightly but thoroughly stuff all cabinet voids with poly-fill as you install the drivers, keeping the entryways to the ducts and the frame of the woofer clear. Wire the drivers to the jack; the tweeters are wired parallel, in-phase.

With a carpet finish the tweeters need not be sealed, but otherwise weather-strip or caulk their frames to their baffle. Reinstall the porthole cover. Install handles and casters as desired. Step back. Admire your work. It's done.

COMPARISONS

I know that a lot of you have been persuaded by the propaganda from speaker manufacturers and music retailers into thinking that you can't have a bass-capable cabinet without using either a 15″ driver or a quartet of tens. You've already seen how the DR10a compares to the DR10, but what is significant is how it compares to a commercial cab.

Figure 6 compares the DR10a to an industry standard, the EV Eliminator™. This cabinet uses a premium 15″ woofer, along with a horn-loaded HF driver. It measures 7ft^3 in volume, 30% larger than the DR10a. At 66 lbs it is also 30% heavier than the DR10a, but it is not, as you might expect, 30% more powerful.

From 32 to 100Hz it averages only 2dB more sensitive than the DR10a (an advantage that putting a 98dB rated driver in the DR10a would nullify). From 125Hz to 16kHz the Eliminator averages 98dB SPL, while the DR10a averages better than 105. That 7dB differential means that to match a DR10a you'd need two Eliminators, along with more than twice the amp power to drive them with.

There is one spec that the DR10a can't match, however. At discounted "street retail" Eliminators go for about $450 each. You can build a DR10a for as little as $125.

The DR10a, like all the DR series, was designed to be reproducible in a molded construction commercially. I decided to compare its performance to a high line cabinet, the Audio Composite Engineering 1250™ (*Fig. 7*). This state-of-the-art unit (at least as far as commercial cabs go) uses molded carbon-fiber construction, allowing it to house a 600W premium woofer and HF horn and still come in at only 40 lbs. Since the cab design is pure T/S, its sensitivity is no better than the Eliminator, but its 600W rating does enable it to play about 4dB louder. The DR10a cannot handle that kind of power, but with efficiency that no T/S box can match, it doesn't have to.

Figure 7 compares maximum SPL levels—the 1250 driven by 600W (top trace), and the DR10a by only 200. Both average 125dB output, with the 1250 having the advantage in the bass, the DR10a in the highs. However, you won't be able to capitalize on the 1250's bass advantage if you don't have a 600W amp. With 200W input the lower trace is all you'll be able to get out of this box.

As telling a comparison as this is, a far more important consideration is the 1250's price, $2,895. I can only think of one (printable) word to describe that figure: Ouch!

When I first showed up at a gig with my 10a, it turned quite a few heads with its unique look. It turned even more ears with its unbelievable sound. Despite its small size and lightweight construction, it really cranks. As for its resistance to vibration, I gave it "the drink test."

At the start of the first set I placed a drink—non-alcoholic, of course—atop the cab, and started to play my bass through it. The drink never moved, and there were no audible vibrations between it and the cabinet. Try doing that with your commercial bass cabinet. Chances are it will do its own version of St. Vitus Dance before taking a header onto the stage.

Since saving money is my passion (OK, I'm just plain cheap), I always try to design my newest cabinets around drivers I pulled from older boxes. My next project, the DR12a, will continue that tradition, expanding the DR10a to house a 12. Watch for it. And for you guys who have to have DRs big enough for a 15—either club or touring size—those will be coming next season.

I start my building season in May. If sometime in the summer there are reports of seismic activity in New Hampshire, it's no earthquake. It just means that I've started my testing. ❖

PARTS LIST

All parts listed with approximate size, not all parts listed (see text), listed in order of assembly.

1. Throat horn sides 6″ × 20″
2. Throat horn divider 6½″ × 5″
3. Throat horn sheaths 6″ × 9″
4. Top, bottom 20″ × 20″
5. Horn supports 20″ × 3¾″
6. Back braces 5″ × 10″
7. Horn braces 1½″ × 16″
8. Tweeter baffle 3¾″ × 20″
9. Baffle 11¾″ × 16″
10. Mouth horn sheaths 15½″ × 20″
11. Side braces 12″ × 7″
12. Back halves 21½″ × 16½″
13. Sides 21″ × 15½″

Universal Test Board for SMT Operational Amplifiers

Keep breadboarding and the spirit of experimentation alive with this test board. **By Fernando Garcia**

Am I becoming eyesight-impaired, or are electronic components becoming smaller all the time? Unfortunately, both statements are true. Surface mount technology (SMT) devices, once the exclusive realm of cell phones, laptops, and other portable electronics, has now found its way into most mainstream products.

Recognizing the trend, component manufacturers now offer everything from arc-suppressors to zeners in the SMT format. Many of the newer, high-performance devices are sometimes found exclusively in SMT versions with no through-hole versions available. Therefore, what once was a curiosity, now has become a necessity for the engineer and hobbyist.

OPTIONS

While poor eyesight can be corrected, what are the options for an electronic experimenter who wishes to breadboard with SMT components? There are generic solutions such as Surfboards®, which allow a few components to be soldered into a small carrier board (*Photo 1*). On the other hand, there are some specialized, application-specific boards usually provided by semiconductor manufacturers as evaluation boards for their proprietary products.

Then, there are quasi-generic solutions such as the universal operational amplifier evaluation board offered by Texas Instruments[1], which the manufacturer calls EVMs. Although only op amps may be evaluated here, most devices follow industry-standard pinouts, and therefore many different types and vendors may be interchanged. In addition, there is enough flexibility in the feedback components so that the most popular op-amp circuits may be breadboarded and tested.

I applaud TI's decision to offer the EVM series. Operational amplifiers are at the heart of every analog circuit. While selecting the appropriate op amp for the task was a simple task a decade ago, now it has become a substantial chore.

There are a number of trade-offs between op-amp parameters; for instance, a micropower consumption usually means sacrificing slew rate and/or output drive capability. For that reason, there are literally dozens of op-amp varieties to fulfill each individual need or application. Careful assessment of data sheets may reduce choices to a few devices. However, some parameter compromises are not clear-cut, and actual testing may be necessary to determine the suitability for the application.

Of course, as audiophiles, we also have the tweak-listen-tweak again itch. Therefore, TI's evaluation board achieves precisely that, providing a versatile test bed for experiments.

The board is divided into four distinct areas—SOIC (small-outline IC), MSOP, and two types of SOT23 op-amp packages (*Photo 2a*). A close-up of one of the sections is shown in *Photo 2b*. Since an op-amp's functions depend solely on the feedback components, the board provides several built-in feedback combinations, so that by populating the appropriate components different functions may be realized.

Texas Instruments is giving away the boards for free, but limits quantities to one per individual, or for larger quantities, 20 pieces for $50. Clearly, TI is subsidizing the price of the boards on the grounds that it will help bolster its op-amp sales. But what if you want to use these boards for personal gain? Or to employ some other vendor's op amps? Although there are no strings attached to the end use of the EVMs, I did not feel comfortable with the idea.

Therefore, I decided to build my own version of the boards. But rather than just outright plagiarize the board, I decided to improve upon the idea.

EVALUATION BOARD DESCRIPTION

My first decision on the new board layout was to limit the package type exclusively to the SOIC-8. Dual op amps are, for obvious reasons, favored for audio work, and most, if not all, of the dual op amps are available in this package. The SOIC-8 is also relatively simple to handle and solder. The tiny footprint of the MSOP and other miniature packages makes them extremely difficult to handle. Because the goal of this test bed is ease of assembly, then the decision to limit it to SOIC-8 appears to be a sensible compromise.

Referring to the schematic in *Fig. 1*, you can see that on the op-amp area the board is laid out with a number of blank component spaces arranged around the op-amp's inputs and output. By stuffing the appropriate passive

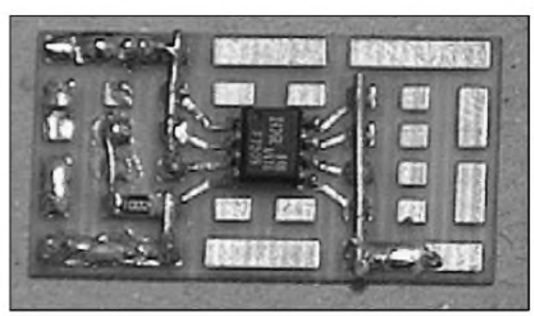

PHOTO 1: Sample carrier board.

components or jumpers in these spaces, you can build and evaluate most of the traditional op-amp circuits.

I attempted to maintain a straightforward identification for the feedback component's labels. It consists of four alphanumeric characters, whose meanings are as follows:

1st Character	I = Inverting input component, N = Non-inverting input component
2nd Character	F = Feedback component, S = Shunt component
3rd Character	A = Op-amp section A, B = Op-amp section B
4th Character	Sequential number, as more than one component is used on each node

Thus, a component labeled NSB2 would be the second available component in section B of the op amp, and it is located in the shunt arm of the non-inverting input. For added versatility, those blank spaces consist of both a SMT pad (0805 size) and a pair of holes (at 0.300″ centers) for leaded components.

Eventually all of the branches for each input are brought to the edge of the board. They are surrounded by a ground (common) plane. These are labeled with the same convention as used with the component labeling and match the reference designators of the schematic. From there you can connect them to ground via short jumper wires to the ground points next to them, or you can connect them to the external world or finally to the general prototyping area where additional circuitry may be installed.

The output also has some flexibility, with two components labeled OUT A and OUT B. Normally, a jumper will be installed there, but you may insert either a coupling capacitor or a termination resistor as necessary. The output also provides for two pull-up resistors, which are employed if an open-collector comparator is being used.

The power input and biasing area provide the usual decoupling capacitors and some flexibility for either single or dual supplies. As shown in the schematic of *Fig. 2* (bottom), for a dual supply a pair of ceramic (C1, C2) and electrolytic capacitors (C3, C4) are connected from the power pins to ground. The "dual-supply" jumper should be in place, thus the pin "common" is attached as the ground for both supplies.

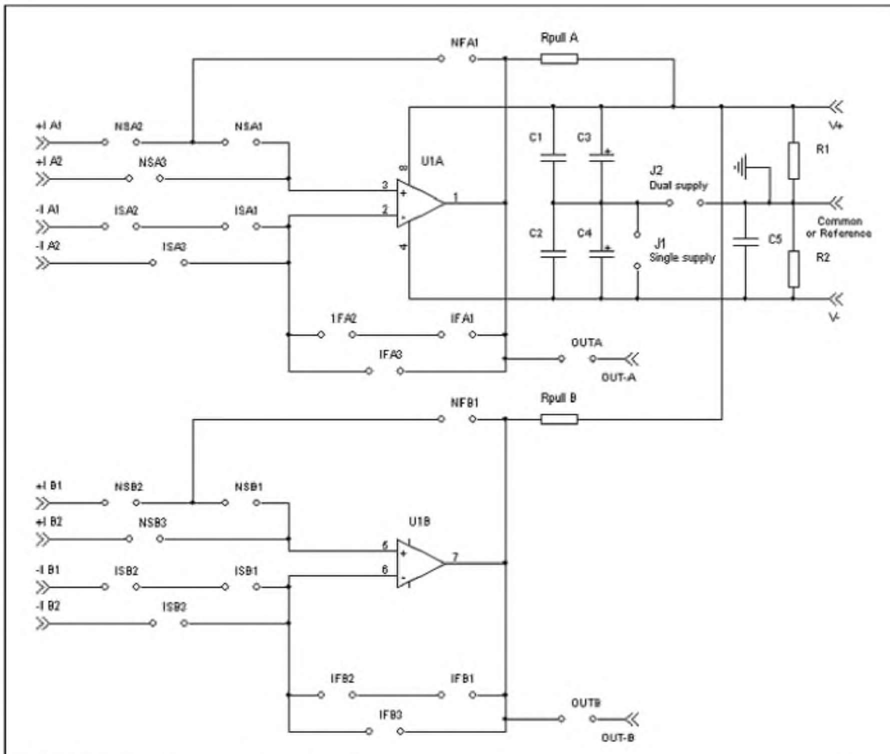

FIGURE 1: Test board circuit layout. A-2060-1

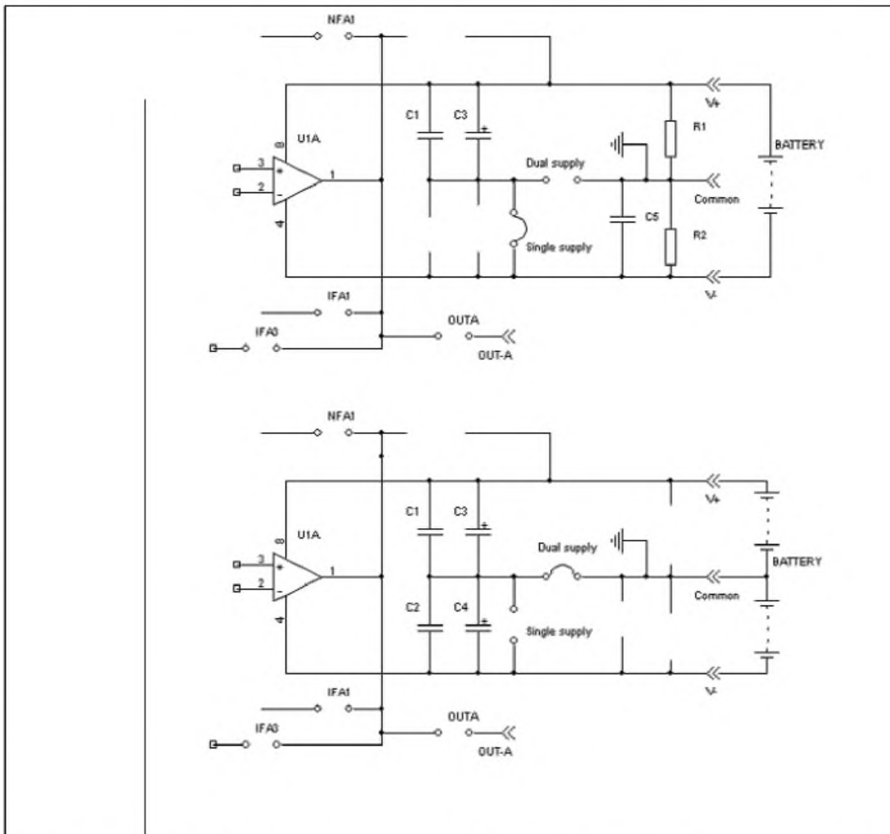

FIGURE 2: Dual-supply operation (top) and single-supply operation (bottom). A-2060-2

If you desire single-supply operation, then as shown in *Fig. 2* (top), the "single-supply" jumper must be in place, and a single ceramic (C1) and electrolytic (C3) capacitor is used. To bias the inputs to the halfway level, a resistor divider area, R1 and R2, is provided, along with its decoupling capacitor C5.

Of course, there will be individuals who like a better "virtual ground" than what a simple resistor divider can provide. A reference voltage or a virtual ground generator (such as Texas Instruments' very interesting TLE2426 device) may be assembled in the general prototyping area or supplied externally.

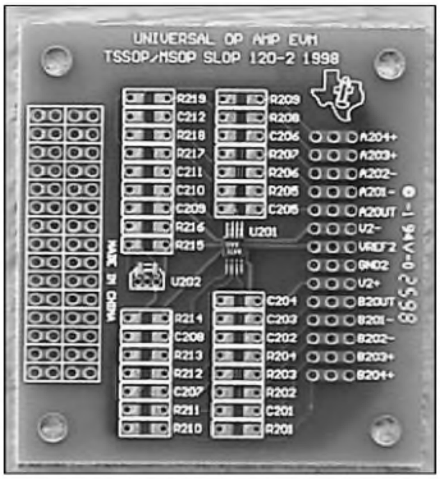

PHOTO 2A: TI's evaluation board.

A side note about the ground plane: The bottom of the board is almost fully covered by unetched copper foil to produce a good ground plane. However, for flexibility this plane is uncommitted; you must remember to always select the appropriate components and jumpers for either single- or dual-supply operation. This flexibility is especially necessary in single-supply operation, where you must decide whether you would like it connected to either the negative supply rail (ground) or to the virtual ground.

Figure 3 shows the layout of the board. In addition to the op-amp section and its associated components, there is a small uncommitted prototyping area in the lower right-hand corner of the circuit. You may use this section, as its name implies, to assemble small auxiliary subcircuits for additional flexibility. This area is arranged in the familiar protoboard fashion, with holes arranged in an array of rows and columns, separated by 0.100″ centers.

A few examples will illustrate how to use this board, which is fairly straightforward. With this knowledge, you may tackle any project you desire.

DUAL-SUPPLY DIFFERENTIAL AMPLIFIER

Figure 4 shows this common and very useful op-amp function. As it is shown in the literature, the transfer function is:

$$A_{vd} = \frac{IFB3}{ISB3}$$

as long as IFB3 = NSB2 and ISB3 = NSB3

Since this application calls for a dual supply, the grounding point is then the common of the two supplies.

As you well know, the common-mode rejection is completely dependent on the tolerance ratio of the resistor pairs. For commonly available 1% leaded resistors, the worst-case CMRR can be as low as 35dB.[2] It is usually better than that because it is very unlikely that all resistors are simultaneously at their worst tolerance. The results I've taken from random unselected pairs of 1% resistors is around 45dB, but to ensure the highest CMRR, you should employ high-precision resistors.

Unfortunately, 0.1% resistors are not widely available in leaded versions at affordable prices. The good news is that SMT resistors are readily available with 0.1% tolerance (Panasonic's ERA series, available through Digi-Key). Thus, having the capability to employ SMT resistors is a major asset. By the way, using the same calculations as before, the worst-case calculated CMRR for this resistor tolerance increases to 54dB, and with a little luck in real life you can expect about 5 to 8dB better.

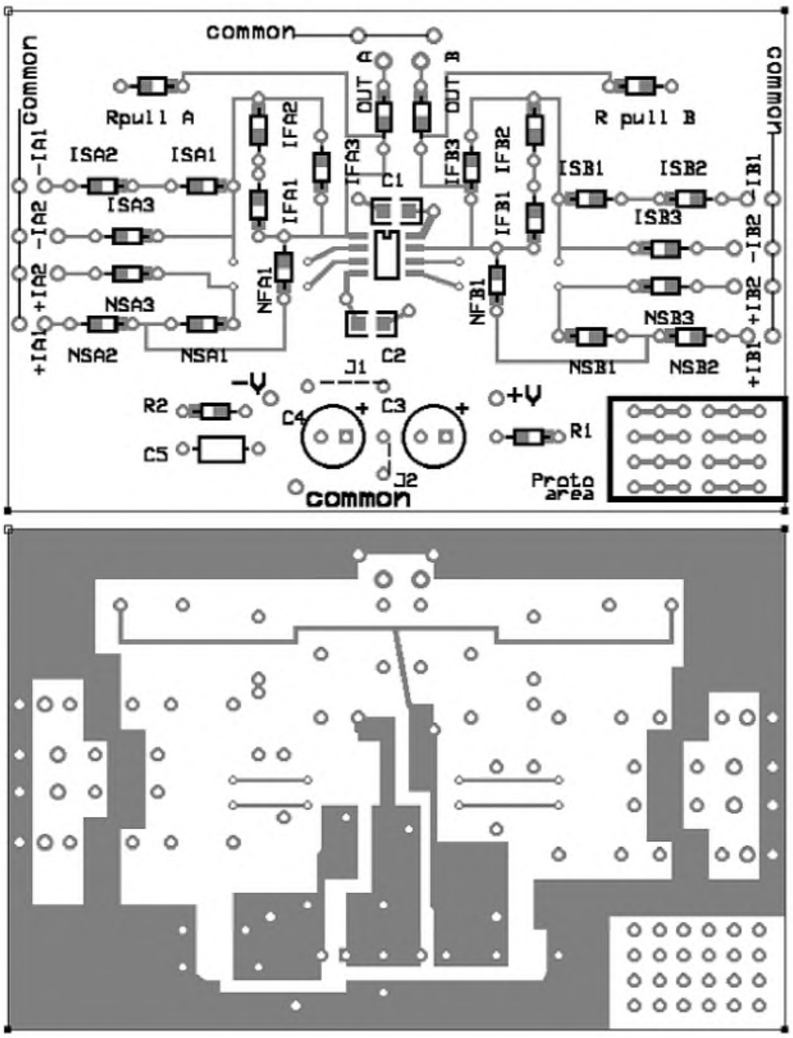

FIGURE 3: Test board layout. A-2060-3

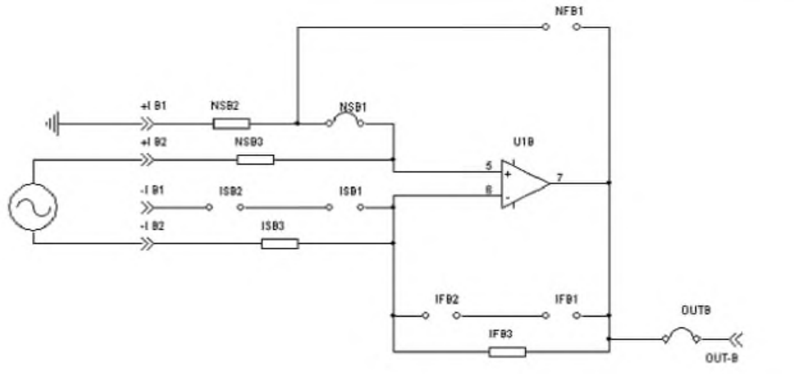

FIGURE 4: Sample dual-supply application. A-2060-4

COMPARATOR

Comparator circuits are not used directly on the audio path itself, but are extremely useful as supplementary control devices. Most comparator applications utilize a single supply, unless you employ specialized comparators such as the LM311, which offers a dedicated pin for the output emitter, allowing it to be referenced to a potential other than the V-supply. Unfortunately, these devices do not follow an industry standard pinout. Thus for the most common applications, a single supply is employed.

Therefore, I utilize what would have been the "ground" node as a reference by installing a resistor from supply V+ that biases a reference diode. This may be either a zener or a bandgap reference. In either case a small ceramic capacitor helps to filter out noise. This circuit may be assembled in the prototyping section of the board.

Comparators almost always employ an open collector output, and therefore a pull-up resistor to the positive supply is required. Provisions are made for this; simply install Rpull A and/or Rpull B, as shown in *Fig. 5*.

Most comparator applications require some hysteresis, or positive feedback, to be applied to enhance noise immunity. This is accomplished via NFA1 and NSA3. Hysteresis is given approximately by the equation:

$$\Delta V \approx \frac{NFA1 \times V^{+}}{NFA1 + NSA3}$$

which assumes that the source resistance is negligible compared to the input resistance. This is not always the case, and you must employ a different technique if the source impedance is variable or if it is a substantial percentage of the input resistance.[3]

Sometimes it is necessary to have frequency-dependent hysteresis; that is, the circuit may need to be fairly immune to slowly changing voltages but have high sensitivity to high slew-rate signals. It is not obvious from the schematic, but all locations have both a SMT pad and holes for leaded components. Thus, with a little resourcefulness, it is possible to install a SMT resistor and a leaded capacitor in parallel in the same NFA1 location to achieve precisely this.

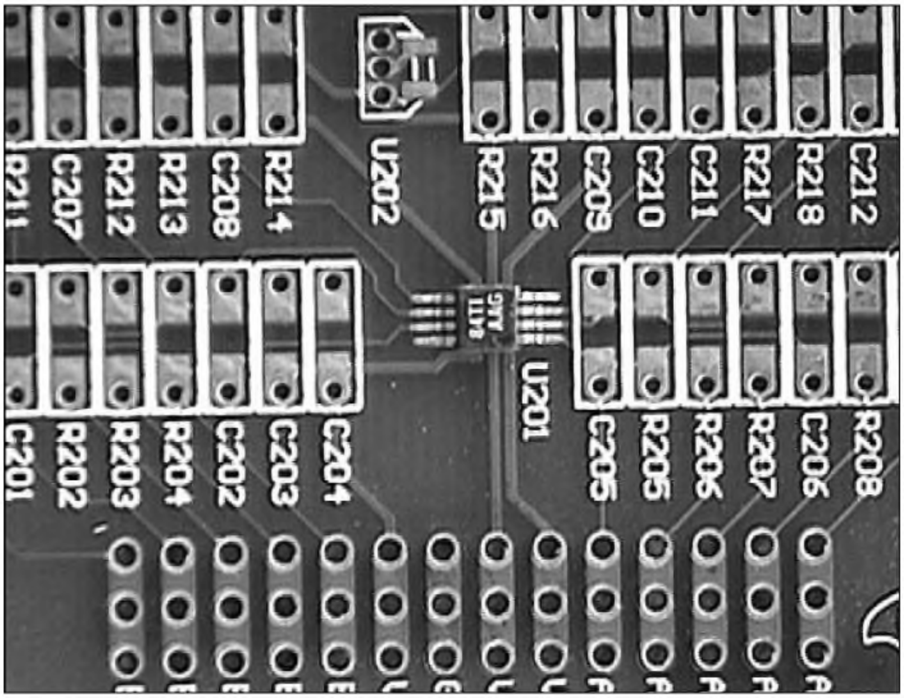

PHOTO 2B: Close-up of TI board.

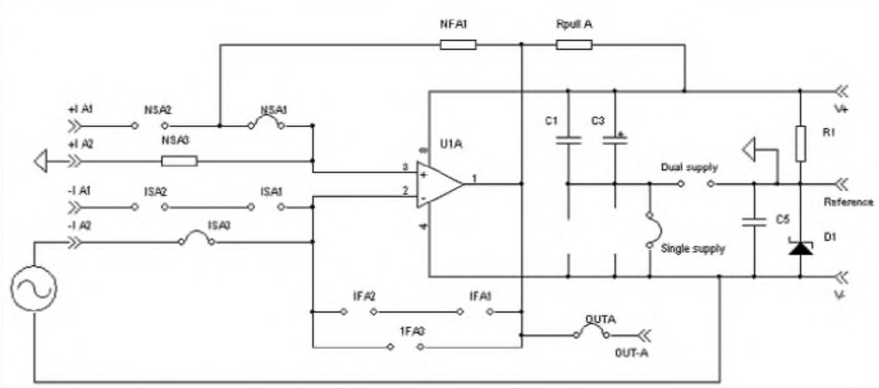

FIGURE 5: Sample comparator application. A-2060-5

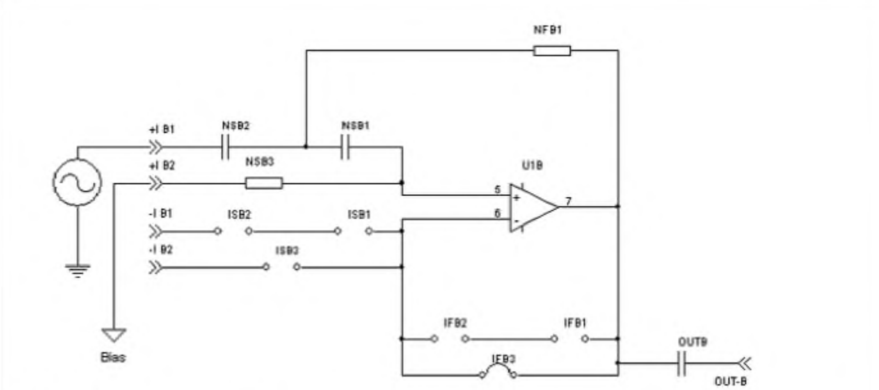

FIGURE 6: Sample single-supply application. A-2060-6

SALLEN-KEY SINGLE-SUPPLY FILTER

Filters are the bread and butter of audio applications. Although there are dozens of filter topologies, the Sallen-Key or VCVS filters are by far the most popular. These simple, easy-to-understand, almost foolproof topologies are easy to cascade, and there are tons of published information for them.[4,5] Most of the time, audio-application filters will employ dual supplies, but there will be instances where a single supply is all that is available (*Fig. 6*).

The high-pass filter lends itself very nicely to a single-supply operation, because by nature the DC signals will be blocked. A second-order filter is easily accomplished by inserting the appropriate resistors at NFA1 and NSA3 and capacitors at NSA1 and NSA2.

The design equations for the desired crossover frequency and filter response are beyond the scope of this short article. You should consult the references or any of the scores of excellent available op-amp books, as all cover filter design in substantial detail. The decoupling capacitor OUTA, which in most cases will be an electrolytic cap, should be stuffed with a high-quality device.

I hope that this and similar evaluation boards will assist audiophiles to enjoy the delight of experimentation for a few additional years. Unfortunately, vendors are rapidly moving towards smaller and smaller device packages, such as National's micro-SMD package, which is barely a 1.6mm (63 mil) square. Vacuum tubes, anyone? ❖

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4. Jung, Walter G., *Audio IC Op-Amp Applications*, 3rd ed., Howard W. Sams & Co., 1987.

5. Horn, Delton T., *Designing and Building Electronic Filters*, TAB books, 1992.

Also see: Unruh, Erland, "Going Surface Mount?," AE *6/99, pp. 32–35.*

The Pearl Phono Stage

Operating under the KISS principle, this author presents a Zen phono stage preamp design. **By Wayne Colburn**

Nelson Pass' Zen amplifiers have generated a tremendous amount of interest over the past seven years. At Pass Labs we enjoy a small daily flood of e-mails and telephone calls about them, and a substantial number of these are requests for a magnetic phono stage preamp design.

INTRODUCTION

The philosophy of the Zen amplifiers is based on the premise that simpler audio circuits tend to sound better, and the design examples explore just how simple they can be. Ideally they comprise a single gain stage—perhaps a single gain device—and achieve good performance without the use of negative feedback. This makes for a clear, if not always achievable, goal.

Pass says that it's much easier to design a line level or even a power amplifier with a simple single gain stage than it is a phono stage. There are three reasons for this: high gain, low noise, and dramatic frequency equalization.

At the lowest frequencies, the phono circuit must amplify the input voltage by roughly a thousand times. By contrast, a typical power amplifier amplifies by 20 to 30 times, and a typical line-level preamplifier by maybe ten times. A thousand times is a lot more, particularly if high quality is also required.

The input noise of the phono stage must be small in comparison to the output of the magnetic cartridge. Typical signal values are on the order of a thousandth of a volt (1mV), and for acceptable performance you need the noise contributed by the circuit to be about a thousandth of that. This value is about a millionth of a volt (1µV).

The equalization curve, known as the reverse RIAA equalization, boosts the bass frequencies by a factor approaching ten times the reference gain at 1,000Hz, and reduces the high frequencies by a factor of 1/10 of the reference. It must do this with about 1% accuracy over the entire audio band, and ideally with the same sonic signature. All this and sound musical, too.

Previous Zen circuits, such as the Bride of Zen (*Audio Amateur* 4/94), do not meet these requirements. First, there was no way to get 60dB gain from a single gain device. Second, MOSFETs are simply too noisy to be practical with microvolt input levels. Third, the equalization circuitry is only practical either as part of an active feedback loop or as a passive circuit in between gain stages. This leads to a design which sandwiches a passive RIAA equalization network in between two simple Zen stages.

SIMPLIFIED CIRCUIT

Figure 1 shows a simplified version of such a circuit. At the input, Q8 is an N channel JFET operated in common-source mode without feedback. The drain of the JFET delivers current to the RIAA equalization network, developing amplified voltage which follows the reverse of the recording curve. The output of the equalization network drives the gate of Q6, which like the input stage operates common source without feedback and drives a drain resistor which develops the output voltage.

The gain devices chosen are very low noise JFETs which have a noise floor on the order of 1µV—much quieter than

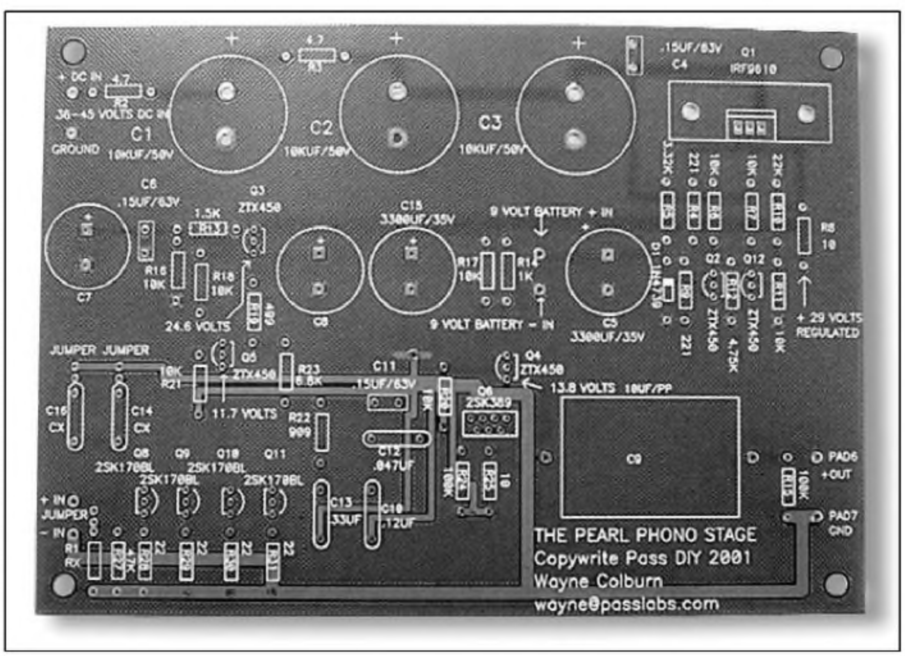

PHOTO 1: Resistors on PC board. (Photos by Karen Douglass)

ABOUT THE AUTHOR
Fifteen years or so ago, Pass placed an advertisement in the back of *Audio Amateur* which said: "Assistant Wanted. High Pay, No Work, All Glory." Wayne is the guy who got the job.

Look!!

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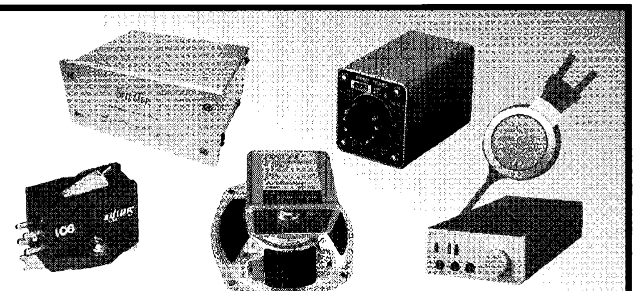

■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	650
Shelter Model 901 (CROWN JEWEL SE)	1,300

Postage (Air Economy)

Area I $18	China, Korea, Hong Kong, Taiwan
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Area III $27	North America, Oceania, Europe
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These Area I ~ IV are for all products except book.

■Japanease Audio Book

Postage $15

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

■MC STEP UP TRANS

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

** Air Economy

■STAX

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

■Speaker

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

*Price is for a pair **Air Economy

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

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

Price is for a Pair

** Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US $)	I	II	III	IV
F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	740	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	760	60	70	110	145
F-2013	40	10	20Hz~50kHz	211,242	730	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160

Price is for a Pair

** Air Economy

E·I·F·L EIFL Corporation

1-8, Fujimi 2-chome, Sayama City, Saitama Pref. 350-1306 JAPAN.
Phone:+81-(0)42-956-1178 FAX:+81-(0)42-950-1667
E-mail:eiflcorp@netspace.or.jp
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the MOSFETs used in previous Zen projects, and quiet enough to work with higher output moving coil cartridges.

Figure 2 shows a less simplified version of the circuit in which some additional features are included to extract greater performance from the two gain stages. First, Q8 has been expanded into Q8 through Q11 operating in parallel. The input noise energy of these devices is inversely proportional to the number of devices in parallel, and so having four of them gives a 6dB improvement in signal to noise. It also allows more gain from the stage, and more linearity by virtue of the higher current going through the set of devices.

You also see something new in Q5—a cascode transistor which shields Q8-Q11 from the voltage output while interposing itself very little on the character of the signal. Cascoding this stage further increases the available gain, improves the bandwidth, and lowers the distortion. You can find a discussion of cascode operation in the article "Cascode Amp Design," which is available on the passlabs.com website.

Similar treatment is given to the second gain stage, where Q6 is actually a pair of JFETs in a single package and is cascoded by Q4.

Figure 3 shows the simplified circuitry for the power-supply regulation for only one channel. The input to this circuit is about 40V DC unregulated, and the output is about 30V DC with as much noise removed as possible. R2 and C1 form the first passive filter removing noise, and R3 and C2 form the second passive filter. D1 is used to form a 9.1V DC reference, which serves as the input to the discrete op amp formed by Q2, Q12, and Q1, whose output is seen at the drain of Q1. This regulated voltage is passively filtered by R8 and C15, and drives the V+ of the second gain stage.

The first gain stage needs a little more filtering, which is provided by capacitance multiplier Q3, which follows the filtered DC voltage across C7 and is filtered again by C8.

Figure 4 shows a diagram of the unregulated power supply, whose components you will not find on the PC artwork in this article or in the parts list. It

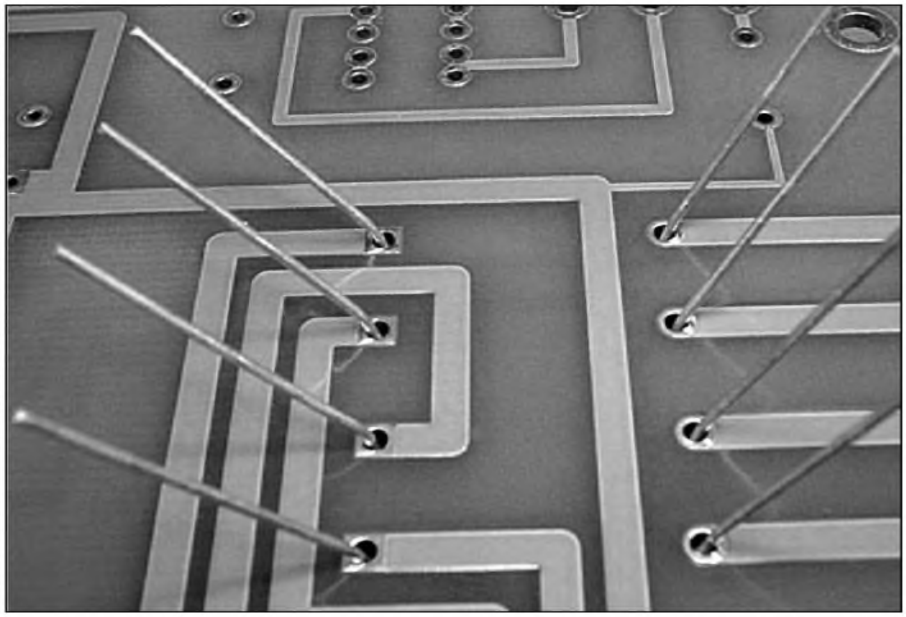


PHOTO 2: Component leads on bottom before solder.

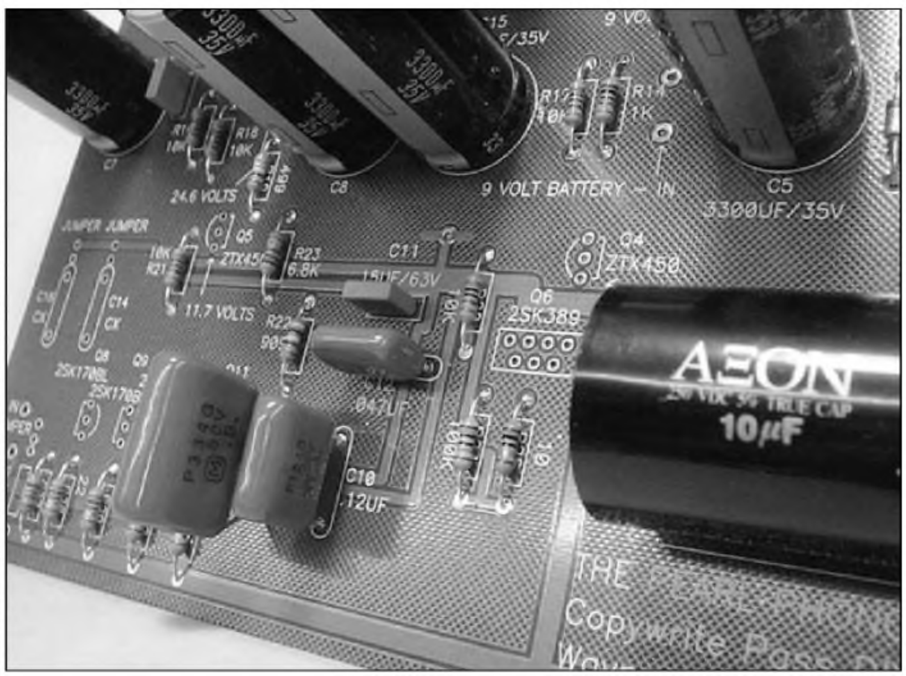


PHOTO 3: Height spacing of components.

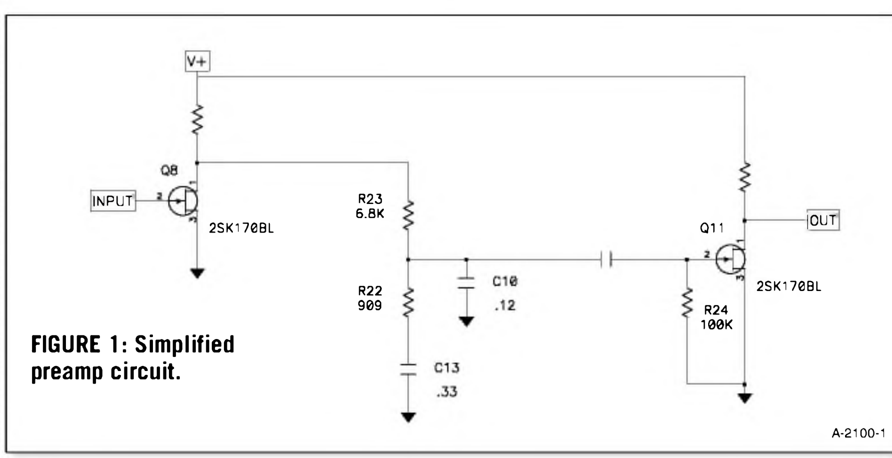


FIGURE 1: Simplified preamp circuit.

provides two isolated DC voltages at about 40V—one for each channel and each powering a regulator from *Fig. 3*.

Figure 4 is simply a suggestion using an Avel Lindberg 4007 transformer, Plitron 017017201, or its equivalent. It is not essential to use this unregulated supply. If you have another source of 40V DC, feel free to try it. This circuit should be fused with about a 1A fast-blow fuse.

In this example, the AC line is filtered by two 4.7Ω 2W resistors and by a 0.1μF film capacitor rated at AC line voltages. A good example is Digikey part P4603-ND. This filter is not essential.

The twin 30V AC secondary windings of the transformer are rectified by bridges rated at 2A and 100V or more such as Digikey 2KBP01M-ND. The DC is stored in 10,000μF electrolytic capacitors. An example of this part is Digikey P6939-ND.

Keep the unregulated supply at a distance, say a couple of feet, from the active gain circuitry to avoid noise pickup from the magnetic fields of the supply.

THE UNSIMPLIFIED CIRCUIT

Figure 5 shows the detailed schematic of all the components appearing on the PC board. In the first gain stage the addition of R28-31 provides ballast to the JFETs, improving the matching, raising the input overload voltage, and lowering the distortion. R27 is the input load resistor and is the maximum value recommended. You can load the phono cartridge with lower impedances simply by placing resistors and capacitors across R27.

Unloaded, the output gain is 4 × 499/(45+22), 29 times and 29dB. The four in the equation is from the four transistors, the 499 is the output resistance, the 22 comes from the 22Ω resistors, and the 45 is the equivalent ohms from the transconductance of each FET.

The resistors and capacitors of the equalization circuit shape the response by both loading the first gain stage and dividing it. R19, R22, R23, and R24 along with C10, C13, and C11 + C12 form the correct response to about ±.1dB when used with 1% tolerance resistors and 2% capacitors.

The second gain stage uses a

2SK389-BL dual FET ballasted by R25 and cascoded by Q4. The gain of this stage is a flat 30dB, and the output from Q4 goes out to the world through coupling capacitor C9.

CONSTRUCTION

All parts are available from Digikey and MCM Electronics. Resistors are 1% ¼W. Capacitors are Panasonic for electrolytics and Panasonic polypropylenes for the RIAA section.

The two isolated supplies will ultimately have their grounds meet each other, and the best place to accomplish this is at the output. If you construct this preamp with two channels in the chassis and an external supply, then the grounds should meet at the output connectors. These output connectors

FIGURE 2: Less-simplified circuit.

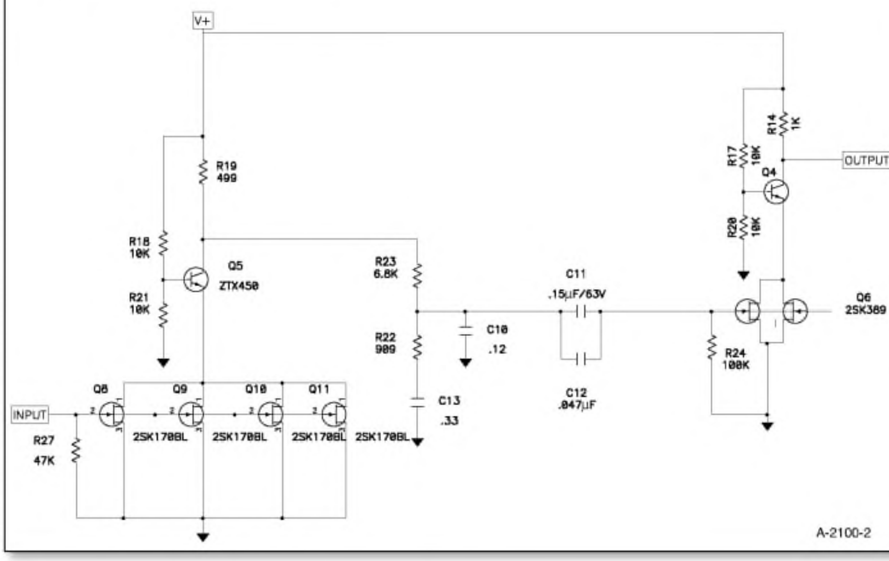
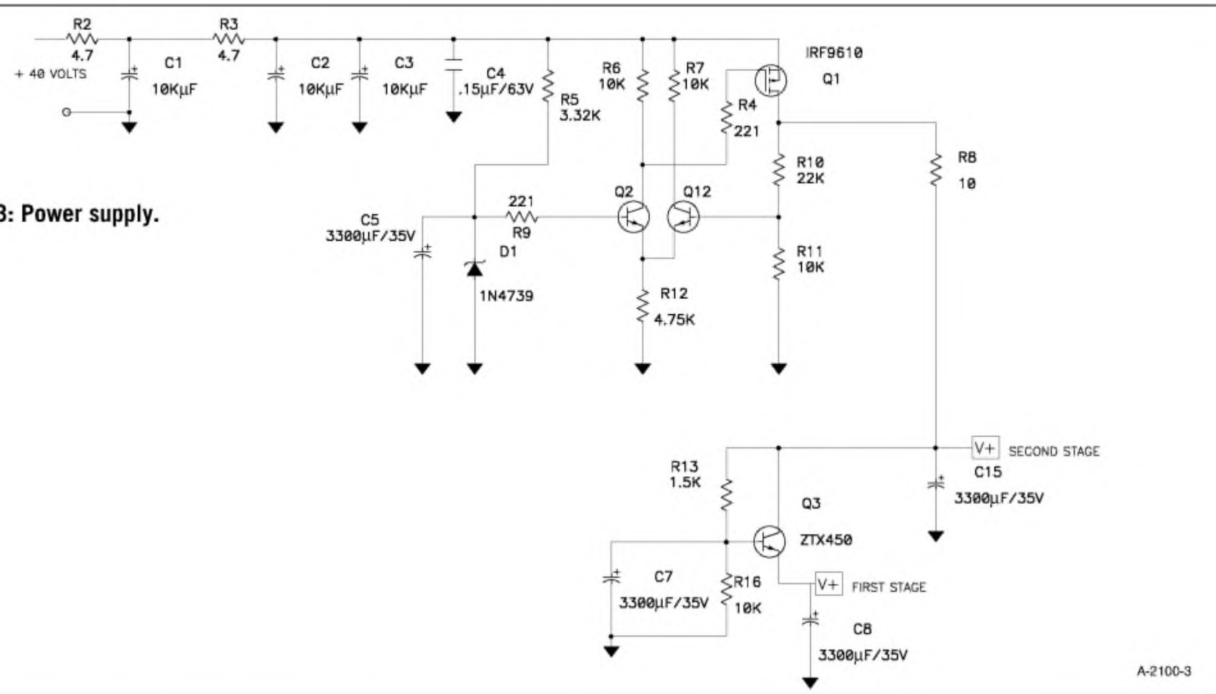


FIGURE 3: Power supply.



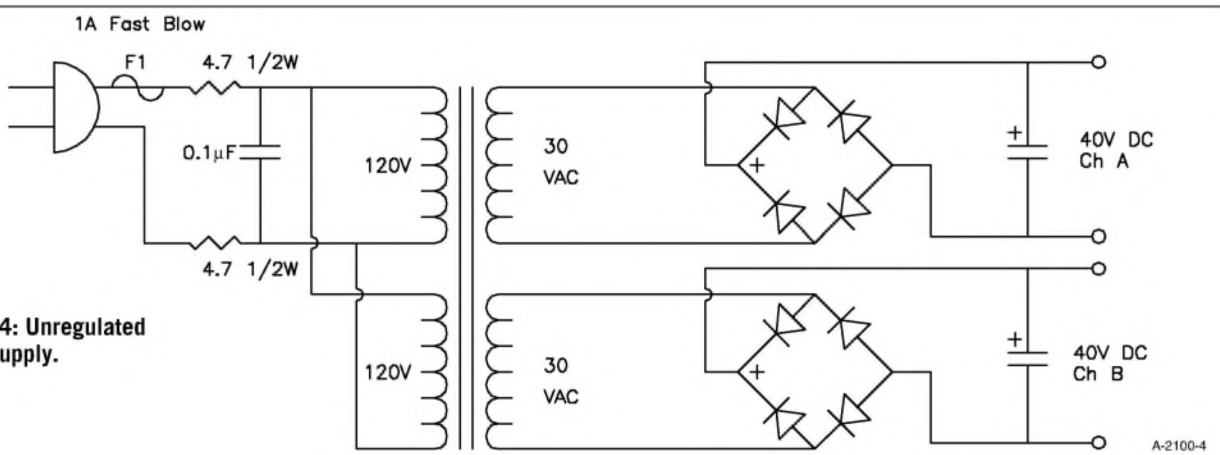


FIGURE 4: Unregulated power supply.

should be close together, and attached to chassis ground at this point.

Check your part numbers against the parts list. Be sure to do this while stuffing the board, also. The component placement on the board is shown in *Fig. 6* and is also silk screened on the PC board itself.

PHOTO 4: PCB with resistors and capacitors.

PHOTO 5: Completed PCB.

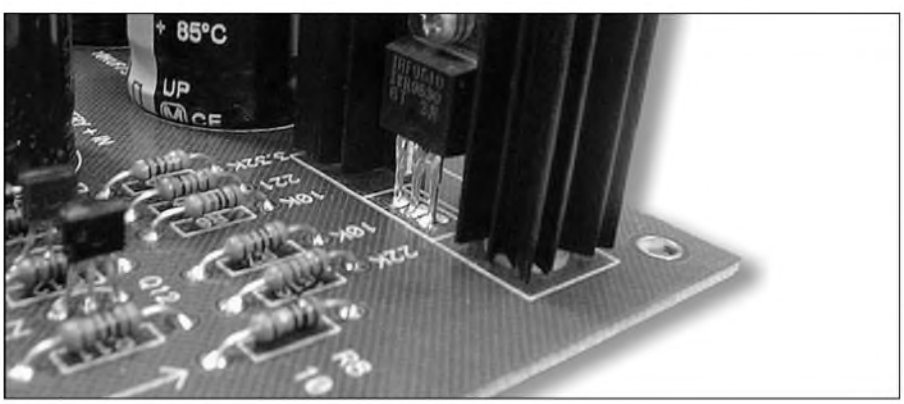


PHOTO 6: Heatsink installation detail.

Stuff and solder the ¼W resistors first (*Photo 1*). Measure component spacing and bend resistors with a resistor bender, so the leads have sharp fitting angles. You can include stuffing the diode (D1) at this point, as it is almost the same size. Be sure to double-check the diode direction. The cathode side is indicated on the PC board as well as the part.

Be sure that the resistors and diode are nice and flat on top, then bend the

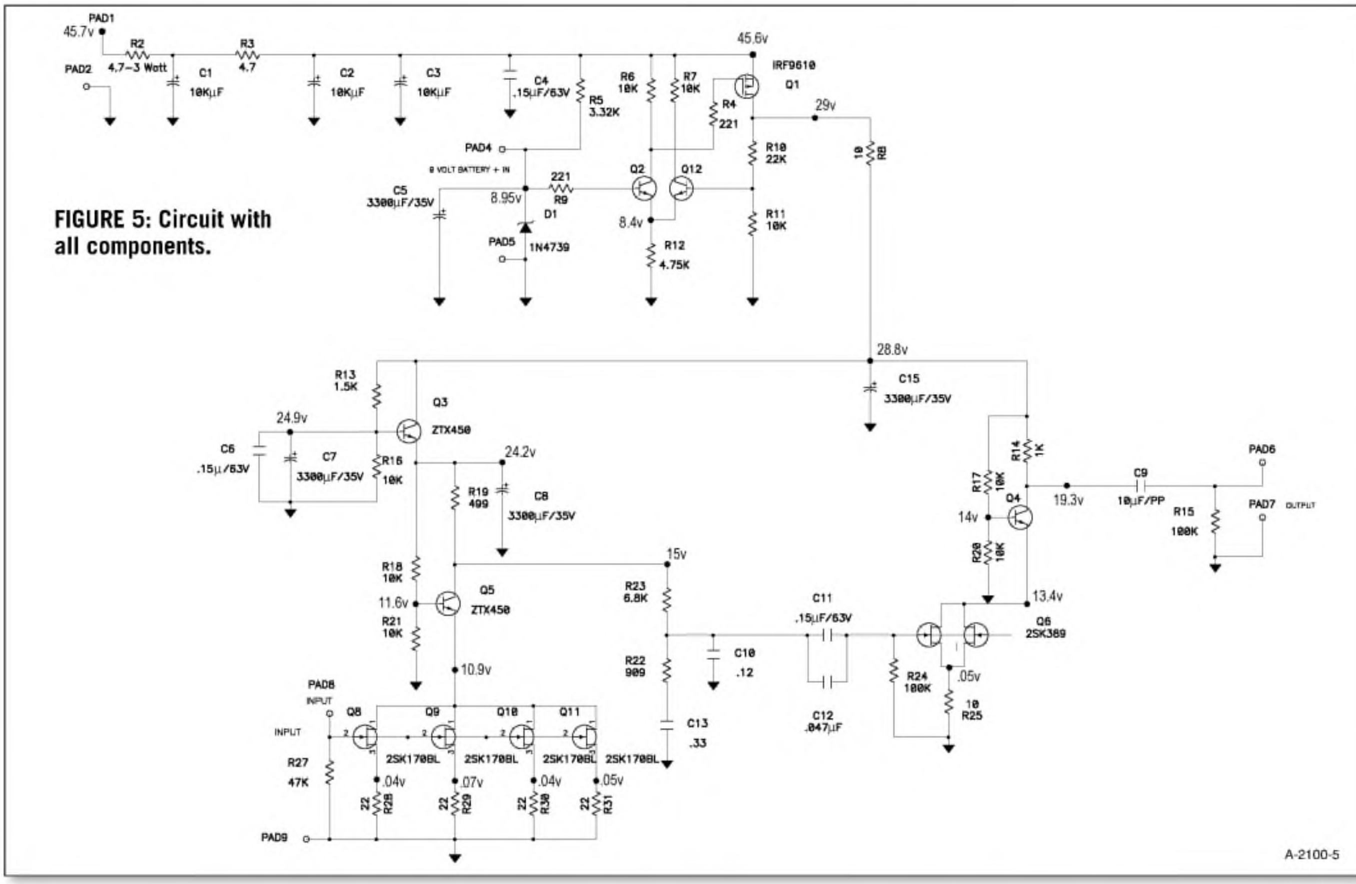
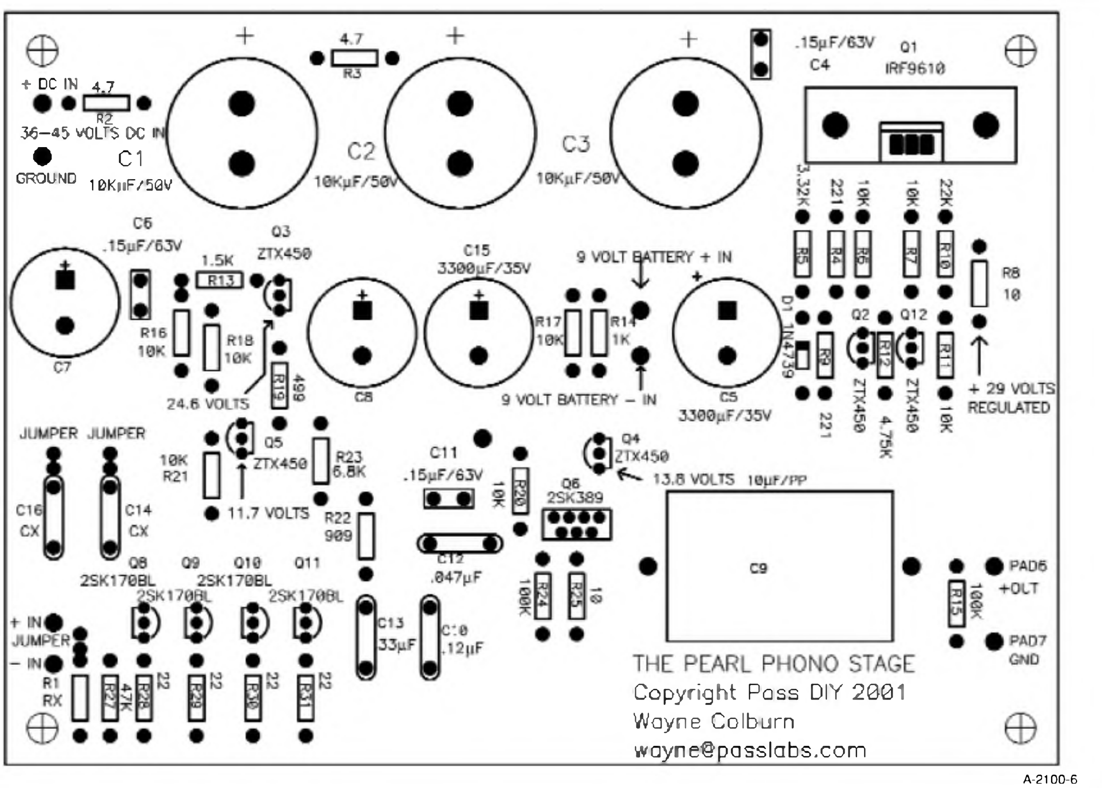


FIGURE 5: Circuit with all components.

FIGURE 6: PC board layout. (Shown here at 50%)



leads on the underside of the board outward at about a 45° angle (*Photo 2*). If you bend them too far apart on the underside, it will inhibit the removal of the parts later. Solder and clip, making sure all solder joints are of good electrical connection.

Place the capacitors in the board next. Capacitors can help protect the static sensitive transistors that you will add later. Some of the ceramic caps may need to be raised slightly, due to the size of the part versus the spacing of the lead holes (*Photo 3*). Be careful to note the direction of the electrolytic caps (*Photo 4*). The polarities are indicated on the PC board as well as the parts. Carefully solder in place and clip the leads.

Next, insert the transistors Q2–Q12, standing them up neatly (*Photo 5*). Don't forget these parts are static sensitive. They also need to be placed on the board in the proper direction. Just match the shape of the transistor package with the shape indicated on the PC board. Solder and clip.

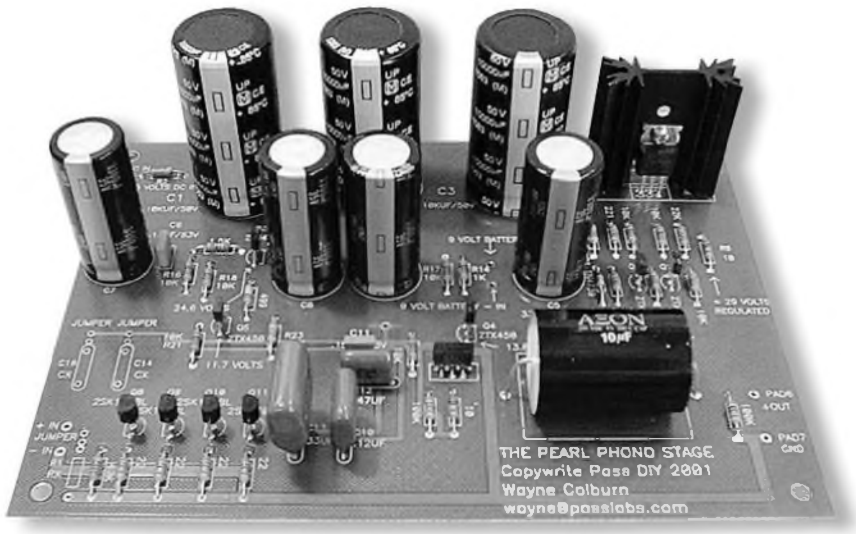


PHOTO 7: Completed channel.

BILL OF MATERIALS

QTY	REF DES	VALUE	DIGIKEY PART #	MOUSER PART #
3	C1–C3	10kµF/50V	P6939-ND	
1	C10	.12µF	P3124-ND	
1	C12	.047µF	P3473-ND	
1	C13	.33µF	P3334-ND	
2	C14, C16	User value		
3	C4, C6, C11	15µF/63V	P3154-ND	
4	C5, C7, C8, C15	3300µF/35V	P5557-ND	
1	C9	10µF/PP		
1	D1	1N4739		
1	Q1	IRF9610	IRF9610-ND	
5	Q2–Q5, Q12	ZTX450	ZTX450-ND	
1	Q6	2SK389		
4	Q8–Q11	2SK170BL		
1	R1	User value		
1	R10	22k		71-RN60D-F-22K
1	R12	4.75k		71-RN60D-F-4.74K
1	R13	1.5k		71-RN60D-F-1.5K
1	R14	1k		71-RN60D-F-1.0K
2	R15, R24	100k		71-RN60D-F-100K
1	R19	499		71-RN60D-F-499
2	R2, R3	4.7 2W	P4.7W-3BK-ND	
1	R22	909		71-RN60D-F-909
1	R23	6.8k		71-RN60D-F-6.81K
1	R27	47k		71-RN60D-F-47.5K
4	R28–R31	22		71-RN60D-F-22.1
2	R4, R9	221		71-RN60D-F-221
1	R5	3.32k		
8	R6, R7, R11, R16–R18, R20, R21	10k		71-RN60D-F-10K
2	R8, R25	10		71-RN60D-F-10
1		Heatsink	345-1029-ND	

Place the 9610 transistor on the board mount heatsink, using a 4/40 - ¼″ screw and kep nut. Tightening the transistor correctly will ensure proper heat distribution of this part. Use the plastic washers to isolate the heatsinks from the PC board (*Photo 6*). Solder and clip the transistor and the heatsink to the board. It should now look like *Photo 7*. You are ready to test it!

TESTING

If you have access to a Variac™ and can bring up the voltage slowly, then do so, watching the current draw through R2 in the power supply of each channel. The circuit draws about 25mA per channel, so you are looking for a voltage drop across R2 of about 0.25V.

If you don't have a method of bringing up the supplies slowly, you will need to plug it in and hope for the best. Observe the voltage across R2 as in the previous paragraph and keep an eye out for smoke.

Figure 5 details the DC voltages to be found at all points of interest in the circuit, and the PC board artwork labels many of these. You will need an inexpensive DC voltmeter to confirm these voltages to within a few percent accuracy, say 5% or so. If the voltages are correct, then the chances of the preamp working correctly are very good.

OBJECTIVE PERFORMANCE

Figure 7 shows the gain of the phono stage referenced to the input voltage and reflecting the inverse RIAA equalization curve. *Figure 8* shows a more refined version of this curve showing deviation from a perfect RIAA curve. Note that using the parts specified ensures accuracy to within 0.15dB over the audio band. There is a slight bump up in the 30–40Hz range of about .15dB, which is not objectionable at all, particularly since the mid to high frequency character is exceptionally smooth.

Figure 9 shows total harmonic distortion (THD) plus noise. For phono cartridges in the 300µV to 1mV range, the characteristic is about optimal and the distortion is an easy to listen to second harmonic at a quite low level.

SUBJECTIVE PERFORMANCE

I tested the phono stage on two differ-

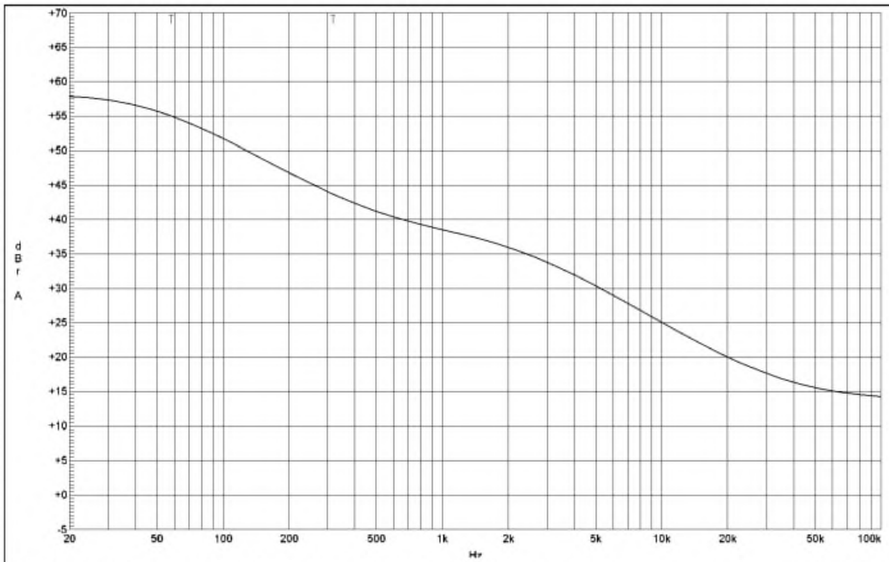


FIGURE 7: Gain versus frequency referenced to input voltage. A-2100-7

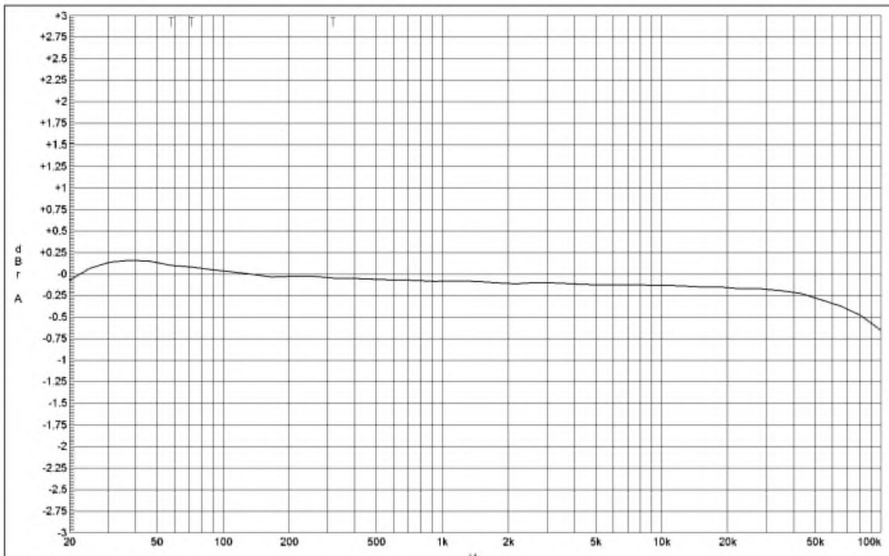


FIGURE 8: RIAA frequency response accuracy. A-2100-8

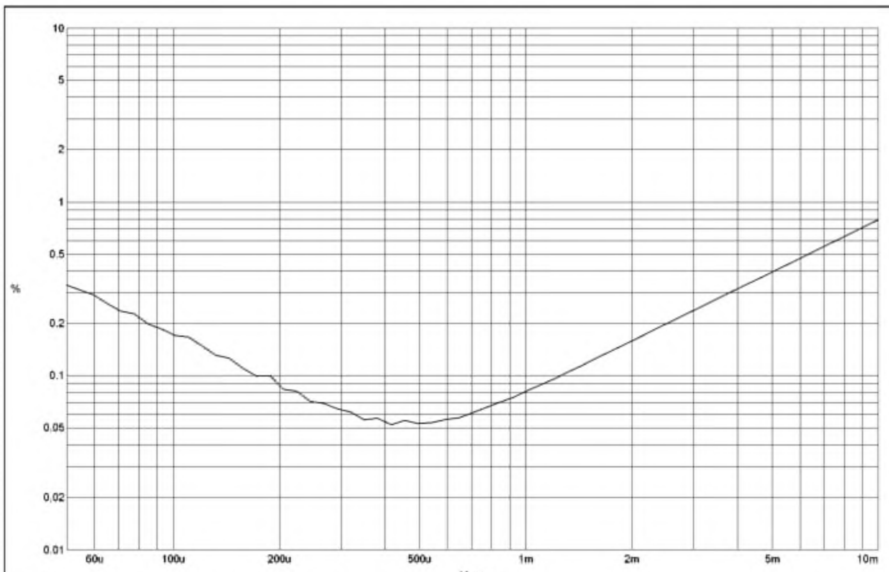


FIGURE 9: THD+noise versus level with 1kHz signal. A-2100-9

SOLDERING "TIPS"

PC board soldering can be a kind of art form when done properly. The result should be a perfect physical and electrical connection between the part lead and the solder pad on the board.

I recommend that you find some solder with low lead content, and rosin core. This can help reduce the amount of lead being dumped into our landfills. You should also have a good soldering iron. Be sure the tip size is appropriate for the job. I suggest a ⅛″ 3mm screwdriver tip.

Hold the iron tip on one side of the joint (where the part lead and solder pad should connect) and feed solder through the other side, on the solder pad surface. If the pad is hot enough, the solder will melt and flow like water around the whole surface of the joint. The cooled solder will have a "volcano" shape to it, appearing shiny, and the joint will have no voids or cracks. Success! (See *Photo 8*.)

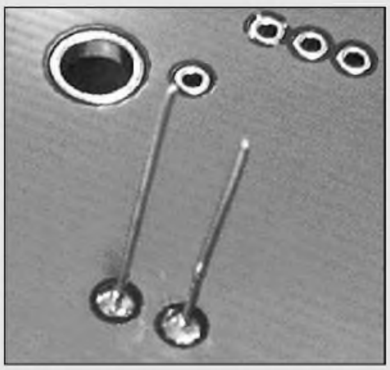


PHOTO 8: Soldering done right.

ent systems with a Grado Sonata (47kΩ load), a Kiseki Lapus Lazuli (100Ω load), and a Sumiko Celebration (1000Ω load) cartridges, all of which have outputs in the range of 400–1500μV.

The circuit was quiet and had sufficient gain. A subjective consensus agreed that the phono stage offered precise imaging and spatial placement and good depth. Compared with the Xono, the Pass Labs reference, it was slightly noisier, and had less gain. From a sonic standpoint, it acquitted itself well in comparison, lacking only slightly in dynamics.

CONCLUSION

Not bad for a simple circuit costing a couple hundred bucks. It acquits itself well against products costing several thousand dollars, and, in the tradition of the Zen amps, does so with single gain stages and an absolute minimum of components. ❖

A Five-Channel Tube Home Theater Amp

Experience home theater with this versatile tube amp...and at a fraction of the cost of commercial units. **By Rick Spencer**

While many home theater five-channel amps are on the market today, most of these are solid-state, and tube enthusiasts are left with only a couple of choices. The available tube amps are expensive (one costs over $2,000), and the total output power is fairly low per channel. The amplifier described and detailed in this project should be very useful to almost everyone who is looking for a versatile, multi-channel tube amp that is a lot less costly than commercial units.

Even though it may sound very involved to construct an amplifier with five separate channels on one single chassis, I think that with the information contained in this article you will find it is not only fairly easy to build this amp but also very enjoyable to try a new project that will give you an amplifier which is certainly different from all the others you may have tried! You can even modify the circuits to allow the use of different tubes in order to obtain other power ratings. The power supply, which is totally contained on the bottom cover plate, offers multiple heater circuit voltages and has enough B+ capacity for a broad range of power output tubes. In the event you are not ready for home theater yet, you can use this amplifier for bi-amping a system or other multi-channel purposes.

GETTING STARTED

The power supply and amplifier circuits are very simple and straightforward, which helps keep the parts count down and the build time to a minimum for such a large project. Speaking of size, the chassis is a Hammond aluminum box #H1444-32 and measures 17 × 10 × 3″. Every square inch of space is put to good use, and yet the amp retains an open and well-arranged appearance.

With 15 glowing valves atop the chassis, I think most will agree that it is quite striking to look at (*Photo 1*). The heat output is minimal, and the total power usage is only about 175W, which is actually less than some two-channel amps.

Using this information, you can build the amp on one single chassis or you may choose to separate the power supply and use two chassis. The choice is yours. I like the appearance of the single chassis myself, and it is much easier to place in the average equipment rack.

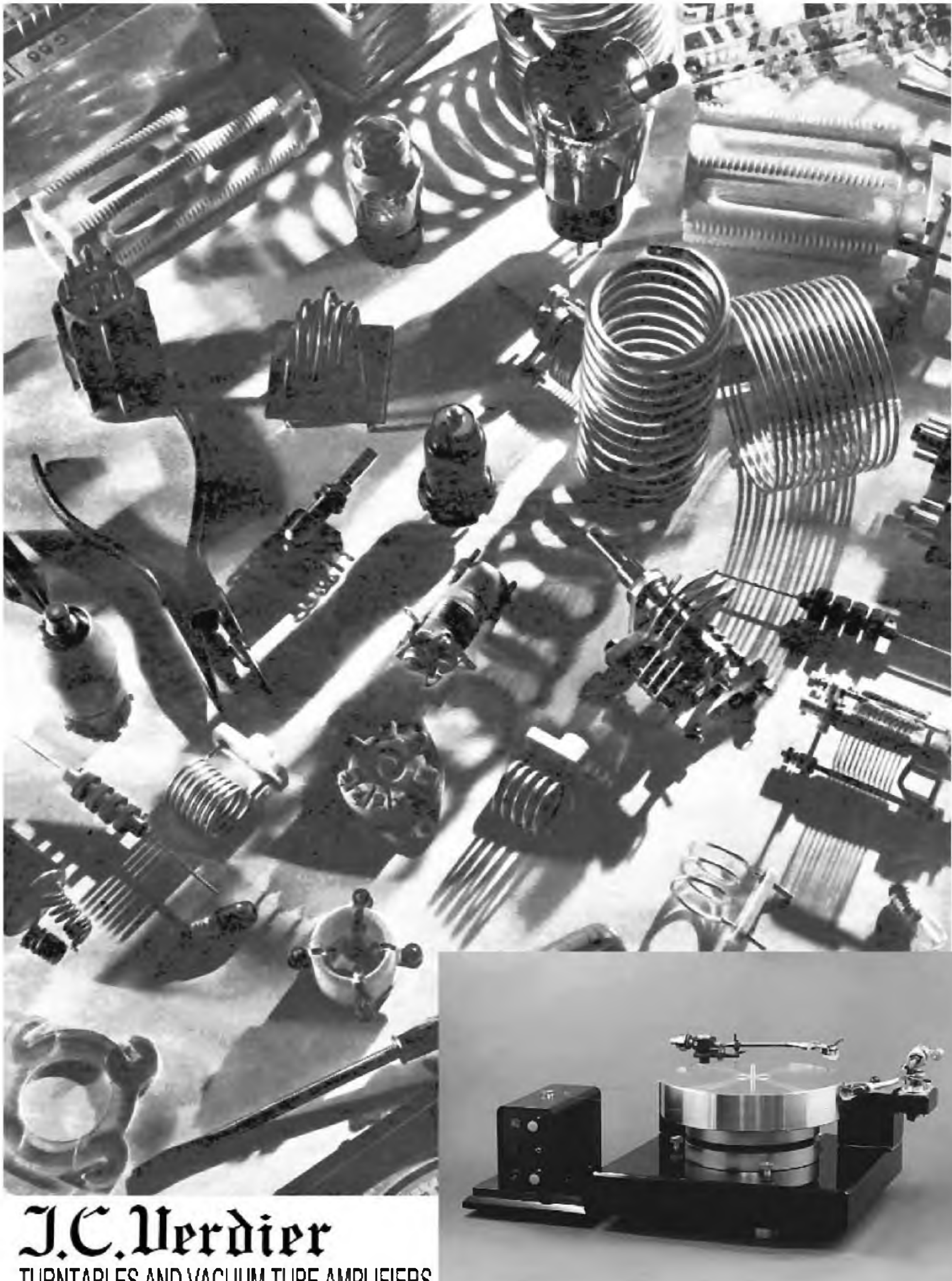
In order to build on a single chassis, I needed to find transformers that would fit in the space of the interior of the Hammond chassis and yet not induce any extra hum in the circuits. I settled on the Plitron toroidal transformers, which are shown in the *Table 1* parts list. They were the perfect size and had the ratings needed for powering all five channels (*Photo 2*). If you use any other type of transformer, then you will certainly need two chassis!

I personally like the Plitrons. They are extremely well made and have a lot of high power for their size and weight. The B+ transformer has a rating of 1,360mA, and the heater tranny has a rating of around 9.5A at either 12.6 or 6.3V!

For more info on the virtues of toroidals over other types of transformers, be sure to read the article by Pete Millett in *audioXpress* June '01 ("Power Transformers for Audio Equipment," p. 14). It contains many great reasons for choosing toroidals over other types of transformers. By the way, the people at Plitron were very nice to deal with, and I found Carol in the order department to be very helpful indeed.

The Plitrons fulfilled all of my expectations about toroidals when I found that I could detect no extra induced hum even when they were mounted in close proximity to the low-level signals in the circuits. Almost unbelievable—but true! This low hum level results partly from the fact that toroidals have only about 10% as much magnetic field

PHOTO 1: Amp in operation with 15 glowing valves (three for each of the five channels).



output as do standard EI types.

According to Plitron, the unique way the windings are arranged acts like a shield. The magnetic field is thereby contained, and that is the main reason I believed I could place the transformer so close to the 12SL7s. There was also no mechanical vibration present.

TABLE 1
PARTS LIST

REFERENCE	PART	SOURCE
RESISTORS		
R1, R2, R6, R11, R12	470k ½W	AES
R3, R10, R14	1k ½W	AES
R8, R9	100k, ½W	AES
R4	200R ½W	AES
R5	1k5 ½W	AES
R7	510k ½W	AES
R13, R15	500R 5W or 2 ea. 1k 2W par.	AES
R16	470k 2W	AES
R17	7k5 25W	Mouser
R18	2R 25W	Mouser
(All of the resistors listed except for R16, R17, and R18 require sufficient quantities to complete five channels.)		
CAPACITORS		
C1	100µF × 16V	Mouser
C2, C4, C5	47nF × 400V Wima (MKP10)	AES
C3	6N8 × 630V	AES
C6	10nF × 1kV	AES
C7, C8	47µF × 50V	AES
C9, C10	220nF × 600V MIT or Solen	AES or TPC
C11	1mF × 350V Sprague or 2 ea. 470µF × 450V Cor. Dub.	Mouser
C12	100µF × 350V	AES
C13	1.2mF × 35V	AES
(All of the capacitors listed except for C6, C9, C10, C11, C12, and C13 require sufficient quantities to complete five channels.)		
TRANSFORMERS		
T1	#067012201	Plitron
T2	#077029201	Plitron
T3–T7	Hammond #1608	AES
RECTIFIERS		
FWB1	6A 800V	Mouser
FWB2	25A 50V	AES
TUBES		
V1 (5 ea.)	12SL7	AES
V2, V3 (10 ea.)	12V6	AES
FUSES		
F1	1A Slo-Blo	
F2	2A Slo-Blo	
SWITCHES		
S1	SPST 3A	AES
S2	SPST 12V 16A	Local auto supply store
S3	SPST 3A	AES

Tube sockets, speaker binding posts, RCA jacks, hardware, hook-up wire, and so on are available from AES. Other copper wire from local hardware stores.

Chassis box and cover plate: Hammond #H1444-32 and #H1434-30 available from Antique Electronic Supply.

AMPLIFIER CIRCUITS

The amp circuits used here are fairly easy to assemble (*Fig. 1*). By following the diagrams you should have no problems getting the amplifier to operate properly on initial start-up. A 12SL7 functions as both a voltage amplifier and a phase inverter. A pair of 12V6s in push-pull configuration serves as the output. A Hammond #1608 output transformer supplies the proper load for the output tubes.

The circuit uses a cathode bias to help keep the amp simple and the parts count down. I used two 1k resistors paralleled for the bias resistor (*Photo 3*). I did this because I had them in stock. You may use a single 500R 5W resistor instead (see parts list).

Also note that each channel is almost a "mirror image" of the other. So, after wiring one channel, just move over to the next one and repeat the process. This method of building will help to avoid the mistakes that can happen when you don't have the benefit of a pictorial diagram to aid you. One important note here about the wiring diagram: Be sure to follow the color codes for the output transformer connections and you should have no wild oscillations on start-up and shouldn't need to reverse any of the leads.

The parts listed will give you a very good sound, but if you wish, you can always upgrade any of the components to suit your own ears. You will find some upgrade suggestions at the end of this article.

You will find that the last amp circuit

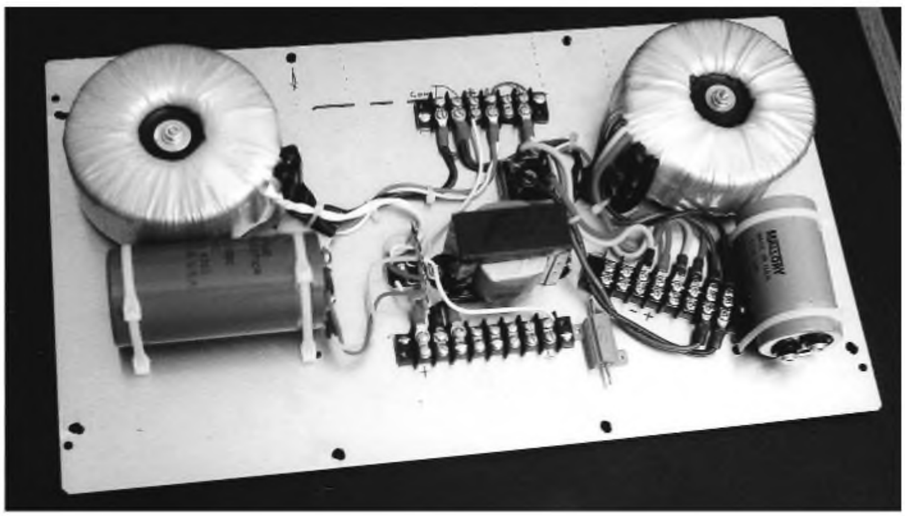


PHOTO 2: Plitron transformers, which are compact, powerful, and were perfect for the power supply.

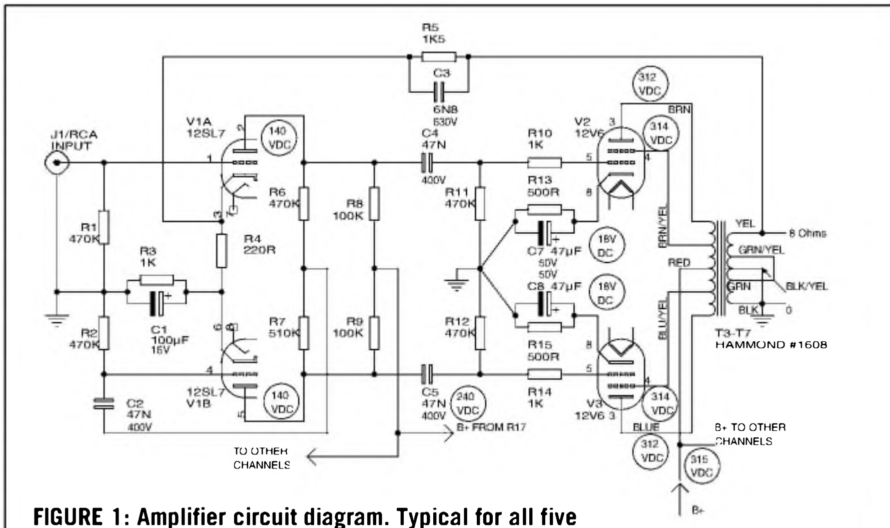


FIGURE 1: Amplifier circuit diagram. Typical for all five channels.

at the left of the photo, which is channel "left front," looks slightly different from the others because the terminal strip mounted by the 12SL7 is attached to the left side of the tube socket to allow more clearance for the transformer. Of course, all of the wiring is exactly the same as the other channels. You can arrange your parts and wiring in any manner you wish, but I found that the arrangement you see in the photos will keep everything neat and close to the chassis.

Star grounding is used throughout the amp, and in the photos you can see the bare copper wires arranged next to the rows of tube sockets. This makes it very easy to mount all of the grounded parts and helps keep lead lengths to a minimum. Remember to insulate the bare leads, where practical, with some heat shrink or other insulating tubing. This will prevent shorted leads and will make testing with a probe much safer.

The types of tubes I chose for the project, the 12V6 and the 12SL7, were used because they kept the cost of the amp down somewhat and because they are in plentiful supply at Antique Electronic Supply as well as some other dealers. You can also use types 6V6 and 6SL7 here because the Plitron heater transformer will supply the necessary 6.3V, and the wiring change is very simple. You only need to wire the primaries in series to obtain the voltage drop needed for the use of the 6.3V tube heaters (*Fig. 2*).

With the use of the 12SL7s and the 12V6s, the total amperage of all the heaters is only around 3A and the Plitron doesn't even become warm. I used only one of the dual secondaries to power the heaters, but if you use the 6.3V tubes be sure to wire them in parallel. This will keep your transformer cool and just loafing along! The B+ tranny was only slightly warm to the touch after two hours of operation.

For some great information about using transformers in power supplies, be sure to read the wonderful article in *Glass Audio* 3/99 by Joseph Marshall ("Power Transformers and Rectifiers," p. 48). I have yet to see a greater wealth of facts about transformer use than those found here.

You could probably use toroidal transformers with a lower power rating than the Plitrons listed here, but just remember that whatever you use they must be up to the task of heating and powering the 15 tubes operating those five channels on your amplifier. If you do use other types, try to keep the total power draw within 65–70% of the transformer's capacity. This will reduce heat buildup and will ensure a long life for the insulation used in the transformer.

If you use the 6.3V tubes, and depending on your power utility voltage, you may need to experiment with the filter capacitor size to maintain the proper heater voltage. If it is too low, add capacitance, and if it is too high, then reduce the capacitance.

When building your amp, wire in the heater circuits first so that you can install all of the tubes and check the voltage readings to make any adjustments

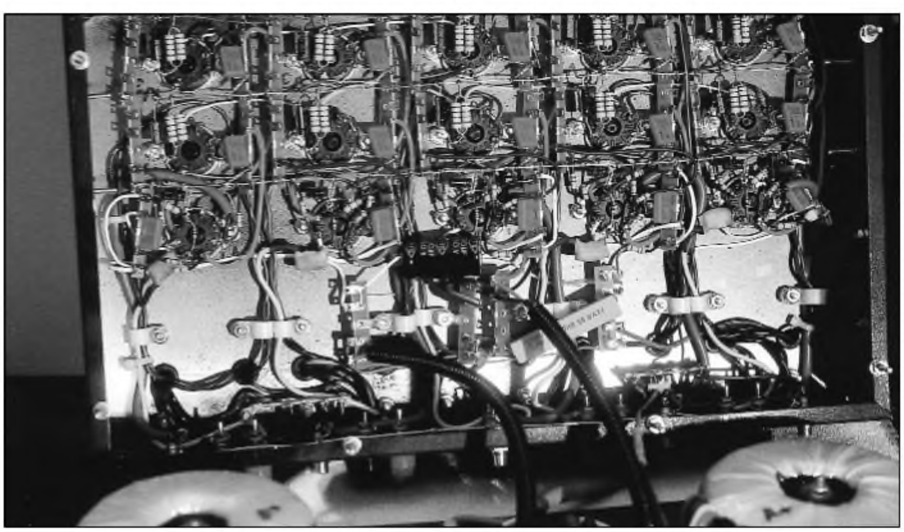


PHOTO 3: Arrangement for bias resistors and "mirror image" layout of all five channels.

at that time. This should make the other circuit wiring easier because the heater wiring is usually larger in size and therefore would be harder to install after all of the other parts were in.

When connecting the wiring for the 15 tubes, try to use the hook-up style shown in the diagram (*Fig. 2*). This should allow all of the heaters to receive an equal voltage. Be sure to remove all of the tubes before you do any other soldering on the sockets!

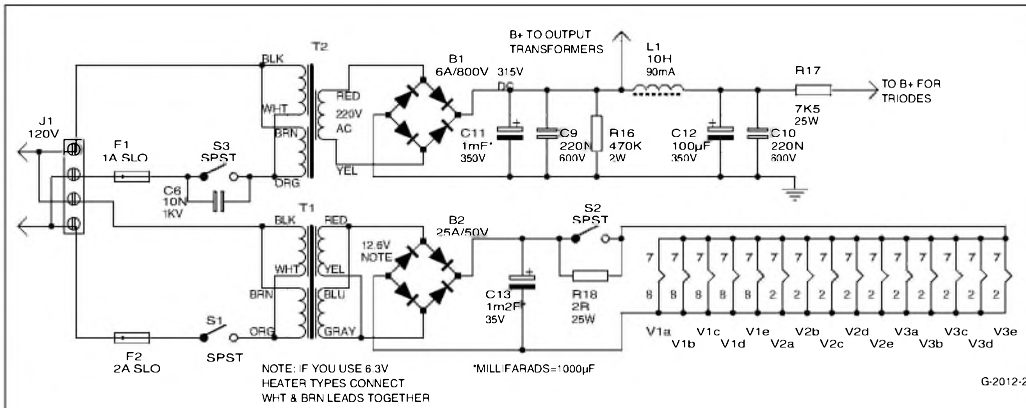


FIGURE 2: Power-supply diagram for all five channels.

POWER SUPPLY

The power supply for this unique amplifier is compact, strong, and efficient. The full-wave bridges, capacitors, choke, soft-start resistor, and all of the terminal strips are grouped in such a way as to form a very neat package. All these parts are mounted on the bottom cover plate, and with the wiring harnesses that connect the B+ and the heater circuits, you can service and test the amp with the bottom plate off the main chassis.

I used type MTW and TFFN wire for the harnesses. This high-grade stranded copper wire is rated for 600V and is available from most hardware stores.

For extra protection of the wiring, I used split spiral wrap covering. I left enough slack to allow for open chassis operation of the amp. The harnesses tuck neatly inside when the plate is attached to the main chassis. Be sure to follow the measurement guide (*Fig. 3*) and *Photo 4*, and your power supply should mount inside OK.

The filter capacitor for the B+ is 1mF and rated at 350V. This is bypassed with a 220nF 600V MIT capacitor. This gives a very "stiff" B+ to the output tubes, and with choke L1 and another capacitor it also makes for a very smooth B+ to the triodes. The large value cap for the B+ was chosen as much for its physical size as for its ratings. If you have trouble finding a suitable single capacitor, you can always parallel two smaller ones (see parts list).

As stated previously, the only choke in the power supply is L1, a 10H 90mA rated coil for the B+ circuit that feeds the triodes only (*Fig. 2*). Because of the stiff B+ rail, I did not think that it was necessary to have any other chokes, and I found that the hum in the speakers was very minimal. Vacuum Tube Logic has used a high capacitance B+ with great success for many years, and they've done it without any chokes in their circuits!

I used in-line-type fuses for the simple reason that I didn't want any high voltage at all on the rear panel. The capacitors are secured on the plate with nylon zip ties (*Photo 4*), which are routed through holes drilled for that purpose. Make sure all of the power-supply parts are secured tightly on the plate because you don't want things to move around. For details about the routing of the power cord, see the section entitled "Construction Notes."

The barrier strips are all 600V rated and make for a very neat and tidy wiring arrangement. Try to keep all of your wiring tightly bundled when building the power supply, and try to make sure that no parts are much higher than 2″ above the plate. In case you do exceed the height limitation, though, don't worry, there are some helpful hints in the "Construction Notes" section.

SWITCHES

I used a couple of extra switches here because they help to protect the tubes in two ways. Switch S2 protects the heaters from the inrush surge that is caused by the elements being cold and thereby having a different resistance than when heated. This switch is installed in the DC circuit for the heaters and bypasses a 2Ω 25W resistor, which is there to allow about 7 or 8V to start the heaters slowly, and when turned on lets the full 12.6V through.

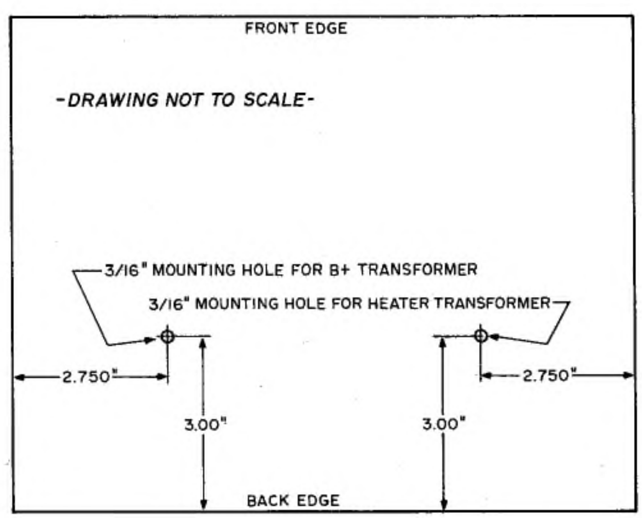


FIGURE 3: Power-supply plate (view from bottom of plate). G-2012-3

Switch S3 prevents the stripping of the cathodes by allowing a heater warm-up period before letting the B+ hit the tubes. This switch is installed in the primary circuit for the B+ transformer. You can build the amp without these switches, but just remember that you have a lot of tubes on top of that chassis, and it's always a good idea to protect any investment of your time and money! (See *Fig. 4* for switch locations.)

The switches were the best choice here over automatic time-delay circuits, because of simplicity and of the obvious space constraints. By the way, switch S2 is an automotive type with a rating of 16A at 12V. It is available at all auto-parts stores. It even has a pilot light inside that will light up the handle when turned on; that is, just in case you think you will really need an indicator telling you all of those tubes are on! (I just left the pilot light disconnected on mine.)

The correct start-up procedure for the amplifier is to turn on switch S1 first, wait about 10 seconds, turn on switch S2, and after another 20 seconds or so turn on switch S3. After your listening session is over, just turn off S3 and S1 and reset S2 to off. Now you're ready for the next time.

CONSTRUCTION NOTES

If you follow the measurement guide (*Fig. 5*) and *Photo 5*, you will have no problem with the layout and the tube spacing arrangement. It took some time getting the spacing just right, especially when trying to make all of the odd measurements come out even. If you don't have a chassis punch yet—size 1 1/8″—this project offers the perfect excuse to buy one.

I drilled and punched all of the holes first, then cleaned the metal with regular rubbing alcohol and sprayed it with the Krylon wrinkle paint from Antique Electronic Supply. When mounting the output transformers and tube sockets, I used standard brass Phillips pan head screws, which gave a nice overall appearance and good contrast with the black wrinkle paint. You'll find these neat little #6-32 screws at all hardware stores.

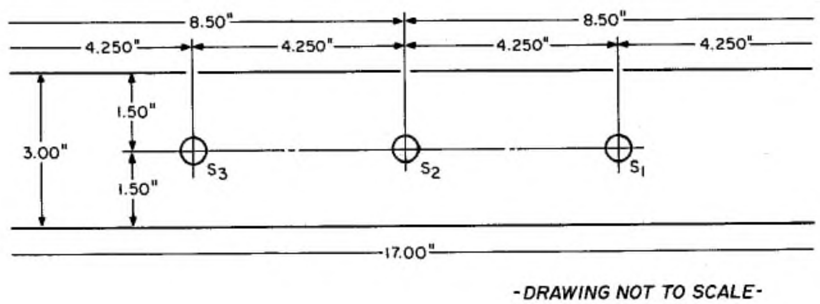


FIGURE 4: Front panel of chassis. Front view showing placement of switches. Note: Drill switch mounting holes as per your switch size, as all types may vary. G-2012-4

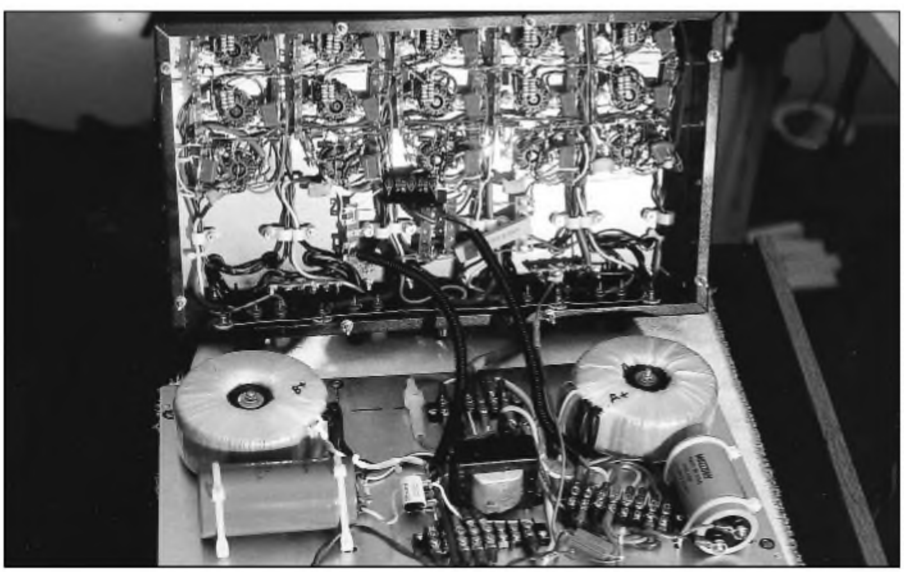


PHOTO 4: Power supply completed and connected into main chassis. Also note switch leads coming from barrier strips.

If you use a different power supply with two chassis, you will still be better off using the Hammond #1444-32 chassis for the amp itself, because this one will hold all of the tubes and transformers without looking cramped. Try to keep the circuit components and terminal strips no higher than 1″ inside the chassis in order for the Plitrons to clear everything. When building with the single chassis box, be sure to also order the bottom cover plate #H1434-30, because the entire power supply is mounted on it. This power supply actually turned out fairly neat and compact.

In the event you are a little uncomfortable with the closeness of the power supply inside the chassis, you can use spacers to hold the bottom plate away slightly. I drilled a total of eight holes in the chassis and plate. That gave me two on each side and two on the front and back edges.

I then installed a #6-32 1″ long screw in each of the holes of the chassis with a washer and nut (*Photo 4*). This made for very steady mounting of the power-supply plate, and by adding an extra nut to the screw, it gave me an extra ¼″ between the chassis and plate. This provided good ventilation for the internal components, more spacing for the power supply, and it also keeps any prying fingers out of the chassis.

The plate comes with four holes already in it, but I found that these did not correctly line up with the edges of the box, so I just drilled my own. When mounted, the bottom plate, which was flexing somewhat under the weight of the transformers, now becomes a very solid part of the chassis.

Always use rubber grommets on the chassis holes that have wires running through them. Use lots of terminal strips (*Photo 3*) on the sockets, because you have a lot of wire connections to hook up. If you bend these over slightly, it will give you even more clearance for the power supply. By the way, the power cord is routed through a hole and notch in the rear panel, which allows it to remain connected to the power supply when the plate is removed. See *Fig. 6* for rear panel details.

TESTING

One of the most important things to remember here is that as you are testing, any of the channels that you don't have a speaker connected to must be terminated with a dummy resistor load. I always use the Radio Shack part #271-120, which has a rating of 8Ω at 20W. However, any brand of resistor with the same rating will do.

I tested part of the time with some old speakers that were left over from some speaker building projects (*Photo 6*). They provided the proper load and allowed me to do certain signal tests with an audible reference. For the final sound tests I used five full-range drivers.

All of the operating parameters are shown on the schematic diagram, and your readings should be within about 10% of these. Because it is slightly cramped inside the chassis, be sure to use extra care and don't get zapped! A quick tip here on how to make a safer test probe is to slip a length of heat-shrink tubing over the metal tip of the probe and leave only a small part of the end exposed. Now shrink the tubing to make a nice tight sleeve to give you an extra margin of safety just in case the probe slips off the contact point while you are testing. While testing I laid an old towel over the power supply to keep my forearms away from the wiring terminals, which gave me an even greater margin of safety.

I was really amazed to find that all five channels tested almost identically, and that they all sounded so close in sonic characteristics. That made all of the effort really worthwhile.

Was there any crosstalk? Not that I could hear. I even injected different audio tones into the various channels at the same time, and each individual tone was only heard on that channel's speaker.

After testing, properly discharge the B+ capacitors with a resistor. Of course, the heater circuit capacitor doesn't need to be discharged because of the load of the heaters being constantly connected.

FINAL ASSEMBLY NOTES

After you have completed your final testing of the amplifier, and you are sure everything is up to par, it is time to attach the bottom plate to the main chassis. With the transformers and choke mounted on it, the sheet metal plate will flex somewhat, and you should grasp it in a manner that will not allow it to bend far enough to loosen any of your parts. I found it easier to attach the plate while the main chassis is tilted back and resting on the output transformers (*Photo 4*).

I lowered the back edge of the plate down until the studs (installed earlier) could enter the two screw holes I had punched in the plate. These two studs will hold the weight of the power supply and keep the plate steady for you. You should even have enough room to get your hand inside the chassis with the plate tilted out.

I then pushed the three switches into their respective holes and attached them with their hardware. These switches were already wired into the power-supply circuits with enough wire length to reach the front panel. Next I tucked the harnesses inside toward the center of the amp. Now all that remained was to center the other screw

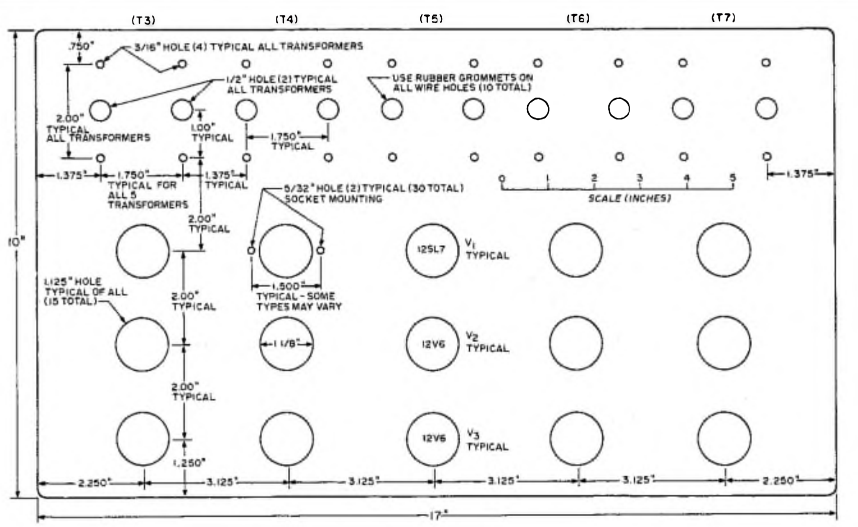


FIGURE 5: Layout and tube placement. G-2012-5

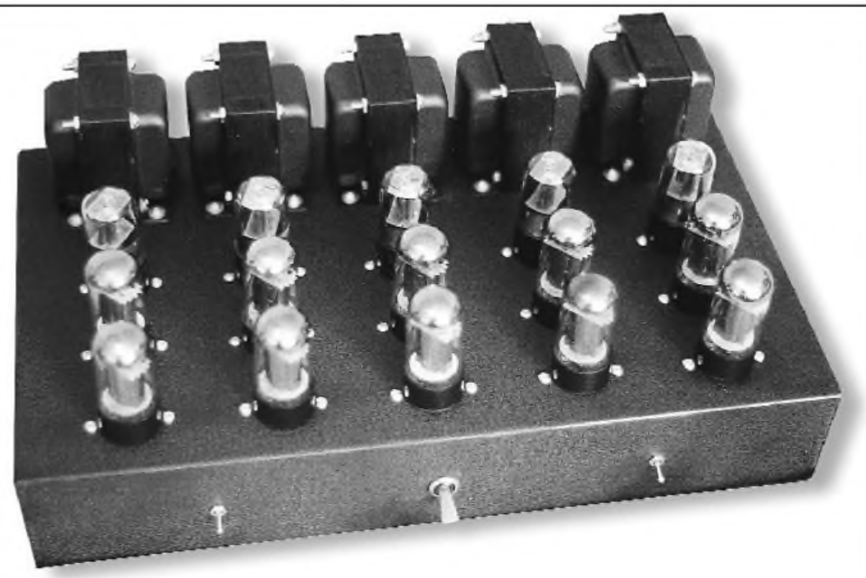


PHOTO 5: Front view of amp showing tube and output transformer placement for each channel.

holes over the studs and install the washers and nuts on each of them.

With everything nice and tight, you can now place the amplifier on its bottom. So far the amp has felt fairly light, but now with the power supply attached it will begin to have a more hefty feel as you move it around. If you were careful with your component placement and wiring, you can feel pretty confident that no shorts will exist. As a matter of fact, while attaching the power supply, I was able to look inside and see that all was OK as I was closing the cover.

If you do have a shorted connection, the fuse in that circuit should blow and you can go back inside, find the culprit, and correct the situation. Remember, if spacing is your problem, simply adjust the distance between the main chassis and the bottom plate with the stud hardware and everything should clear just fine.

SOUND

Allow about 25 hours of operation for the amplifier to break in and settle down. The amp has a very good overall frequency response and a very clean and crisp sound. It was quite different to be surrounded by so many channels of sound coming from a single chassis amp. DVD movies in surround sound will just blow you away, especially when those sounds are coming from a tube amplifier that you built yourself!

So, good luck, take your time constructing this project, and enjoy yourself because when completed you will truly have an amplifier that is really unlike most all others available today!

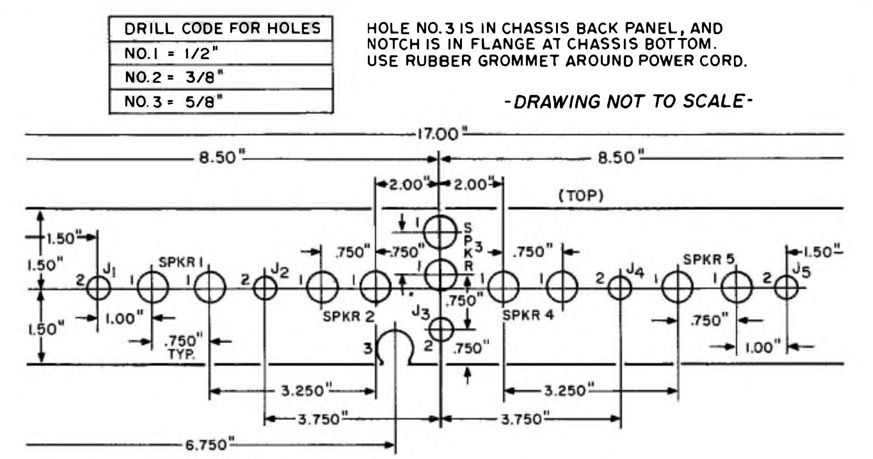


FIGURE 6: View of chassis box back panel.

PHOTO 6: Amplifier under test with speakers connected to all five channels. (Three are visible and the other two are mounted under workbench.)

UPGRADES

Probably the best upgrade you can perform on the circuits is to replace the inexpensive Wima coupling capacitors with Hovland Musicaps. At about $10 each, this will add around $150 to the cost of your amplifier, but if you possess a critical ear these caps are really worth it. Any upgrades of your choice should be done in such a manner as to leave adequate clearance for the power supply. You can also go for the "high end" tube sockets, wiring, and resistors if you wish.

To me, that's the best part of this hobby, having it our way! Of course, all of these upgrades will drive up the cost of your project. After building this amp, I found my total investment to be only about one-fourth of what a commercial five-channel unit cost, and this amp has more power and can be easily modified. The total audio power of this amplifier is around 60W, while the $2,000 model has about 40W.

In case you are into better power supplies, and you are building with two chassis, you can use any method of decoupling you wish for the various channels; i.e., chokes or resistors.

If you desire more power, you can build this amp with 6BQ5s for about 18W and with 6L6s for 20W per channel. The Hammond output transformers for those tubes should fit the chassis if you are careful with your layout arrangement. If enough interest is shown for the tube and power upgrade, I will pass the information and diagrams along to *audioXpress*.

In case you already have a stereo amplifier that you are fond of and want to use, you can always build this amp with only three channels using the Plitron transformers and information in this article. In the event you want to construct a three-channel version, remember to keep the amplifier circuits in the middle of the chassis area and you should have lots of room for the power supply to fit in. Also, with a three-channel amp you should delete resistor R17 and install a 15k 2W resistor in the B+ that feeds the triodes in each of the channels. ❖

Measuring Passive Radiators

The parameters of a passive radiator can be calculated from impedance curves. A method is described. **By David Harris**

PHOTO 1: Two views of an impressive 15″ Lambda PR.

A passive radiator is a suspended, moveable diaphragm in the wall of a speaker. It vibrates in response to the movement of the woofer's diaphragm and serves much the same purpose as a vent. A long time ago passive radiators were called "drone cones." They can be easily constructed from a woofer by removing its magnet assembly. I refer to passive radiators by their commonly used acronym, "PR," for the remainder of this article.

The popularity of PRs (*Photo 1*) has grown in recent years. Unfortunately, a common impediment to their use is a lack of accurate parameters. Even when parameters are available, they are often incomplete.

PR PARAMETERS

A complete set of PR parameters might include familiar terms such as F_S, Q_{MS}, V_{AS}, C_{MS}, M_{MS}, R_{MS}, S_D, X_{MAX}, and X_{MECH}. Since these terms are usually associated with drivers, you can adjust their names as follows to show that they apply to PRs:

F_{PR} The free-air resonance frequency of the PR in hertz (Hz).

Q_{MPR} The mechanical resonance magnification of the PR at F_{PR}.

V_{APR} A volume of air equal to the mechanical compliance of the PR's suspension in cubic meters (m^3).

C_{MPR} The mechanical compliance of the PR's suspension in meters per newton (m/N).

M_{MPR} The moving mass of the PR's diaphragm including the air load in kilograms (kg).

R_{MPR} The mechanical resistance resulting from suspension losses in kilograms per second (kg/s).

S_{DPR} The moving diaphragm area in square meters (m^2).

X_{MAXPR} The maximum linear excursion of the diaphragm in meters (m).

X_{MECHPR} The mechanical excursion limit of the diaphragm in meters (m).

Note: Both X_{MAXPR} and X_{MECHPR} are measured in one direction from rest. P-P or peak-to-peak values must be halved.

You can divide these parameters into two sets, the "small-signal" and "large-signal" parameters. The small-signal parameters are measured with a relatively small or low-power audio signal, and they describe the general characteristics of the PR. They include F_{PR}, Q_{MPR}, V_{APR}, C_{MPR}, M_{MPR}, R_{MPR}, and S_{DPR}.

The large-signal parameters are measured with a large or high-power audio signal to test the extreme limits of the PR. They include X_{MAXPR} and X_{MECHPR}. The scope of this article is limited to the small-signal parameters, because large-signal parameters require a much more sophisticated test setup.

Once you know the small-signal parameters of a PR, you can use them with a computer box design program to model the response of a speaker with a PR. This makes PR box design almost as simple as vented box design.

PR SYSTEM IMPEDANCE

The system electrical impedance of a woofer with a PR is shown in *Fig. 1*. This is the voice-coil impedance of the woofer. Readers who are familiar with vented boxes will quickly recognize the similarity of the impedance curve to that of a vented box speaker.

The first impedance peak (F_L) is the resonance of the PR in the box. The second peak (F_H) is the resonance of the woofer in the box. The height of F_L is usually a little lower than F_H due to PR losses. The minima (F_M) between the two peaks is very close to the system resonance of the box. The impedance of F_M is normally a little higher than the DC resistance (R_E) of the woofer's voice coil.

There are a variety of ways to measure the parameters of a PR. The

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David E. Harris has had a keen interest in audio since his teens. His eclectic background includes graphic art, writing, programming, audio recording and mixing, pro-sound reinforcement, radiological health/nuclear chemistry, and theology. As a design engineer for Ambassador University, he developed software for speech intelligibility and sound coverage prediction. He holds two degrees and now serves as the president of his own company, Harris Technologies, Inc., which produces the speaker design software BassBox Pro and X•over Pro.

method presented here is based on woofer electrical impedance measurements. As such, a woofer with known small-signal parameters is required. For convenience I include the measurement of relevant woofer parameters in this procedure.

In the interest of making this procedure accessible to a wide range of speaker designers, I use an inexpensive "Woofer Tester" from Peak Instruments for all measurements (available from Parts Express for $200). The Woofer Tester is a small external device that connects to a PC via a serial port. Woofer Tester software runs on the PC to control the device and make measurements. (Note: The author's Woofer Tester shown in *Photo 2* is an early model. Newer models function the same but are packaged in a more attractive case.)

You can use other test systems with this procedure as long as they can measure driver small-signal parameters and make impedance response measurements. I recommend complex impedance measurements containing both the magnitude of the impedance and the difference in phase angle of the input and output test signals. The phase angle can be extremely helpful in locating the frequency of F_L, F_M, and F_H (*Fig. 1*) because the phase angle should normally pass through zero degrees at these three frequencies.

PR MEASUREMENT PROCEDURE

This procedure is based on Douglas Hurlburt's paper, "Complete Response

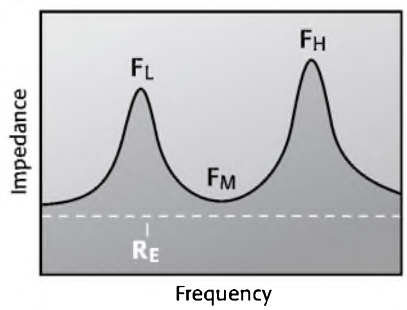


FIGURE 1: Electrical impedance of a woofer with a PR.

FIGURE 2: The "Plot impedance fs, Qts" function.

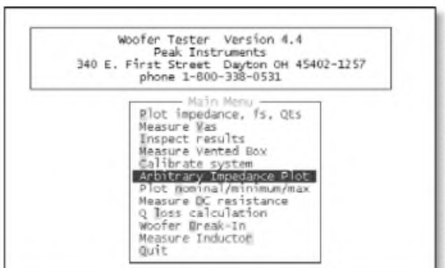


FIGURE 3: The "Arbitrary Impedance Plot" function.

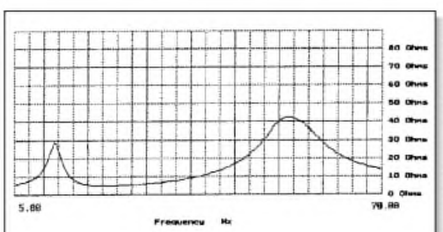


FIGURE 4: The "Arbitrary Impedance Plot" graph.

Function and System Parameters for a Loudspeaker with Passive Radiator," published in Volume 48, Number 3 issue (March 2000) of the *Journal* of the Audio Engineering Society.

Required Test Equipment

- Woofer Tester
- Low-frequency driver (woofer)
- Test box large enough to accommodate the driver and PR with airtight mountings for each. You will need a plug to block the PR mounting hole during the test.

Step 1: Calibrate the Woofer Tester with its calibration resistor.

Step 2: Position the woofer out of the box (in free air) and connect it to the Woofer Tester. Measure and record the woofer's F_S, Q_{MS}, Q_{ES}, and R_E with the "Plot impedance, fs, Qts" function (*Fig. 2*).

Step 3: Measure the internal volume (V_B) of the test box. V_B should be measured in, or converted to, cubic meters (m^3).

Step 4: Mount the woofer on the test box while being careful to produce an airtight seal. You might find it easier to mount the woofer backwards (*Photo 3*) so that its voice-coil terminals are outside the box, making for easy connections. Seal shut the PR mounting hole with a solid plug or panel so the test box is completely sealed.

Step 5: Using the Woofer Tester, calculate the woofer's F_S and Q_{ES} parameters again. Since the woofer is now mounted to a closed box, record these parameters as F_C and Q_{EC}, respectively.

Step 6: Calculate the compliance ratio, α (alpha), and the resonance frequency of the driver with the box air load, F_{SB}:

$$\alpha = (F_C \times Q_{EC})/(F_S \times Q_{ES}) - 1$$
$$F_{SB} = F_C/(\alpha + 1)^{1/2}$$

F_{SB} is in hertz (Hz).

Step 7: Measure the moving diaphragm or piston diameter (D_{IAPR}) of the passive radiator. This is the diameter of the diaphragm or cone plus about half of the surround (*Photo 4*). D_{IAPR} should be measured in, or converted to, meters (m).

Step 8: Calculate the moving diaphragm or piston area, S_{DPR}, from D_{IAPR}:

$$S_{DPR} = \pi \times (D_{IAPR}/2)^2$$

Pi (π) is approximately equal to 3.1416. S_{DPR} should be in square meters (m^2).

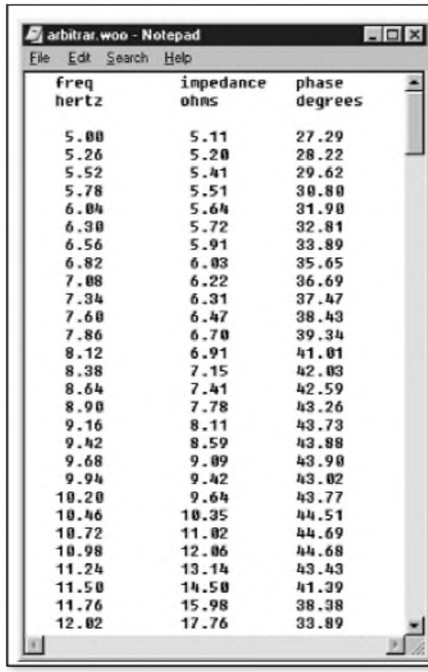
Step 9: Remove the plug or panel from the PR mounting hole and mount the PR to the test box as shown in *Photo 5*.

Step 10: Using the "Arbitrary Impedance Plot" function of the Woofer Tester (*Fig. 3*), measure the impedance of the woofer's voice coil while it is mounted to the test box with the PR.

Begin with a wide frequency band (low endpoint = 5Hz, high endpoint = 100Hz) with only a few steps (number of steps = 25) and identify the approximate frequency of F_L and F_H. Then redo the impedance measurement with a low endpoint about 10Hz below F_L and a high endpoint about 10Hz above F_H. Use about 200 sample steps so that the impedance data will be detailed (*Fig. 4*).

Step 11: Open the "arbitrar.woo" data file produced by the "Arbitrary Impedance Plot" function of the Woofer Tester. It should be located on your computer's disk drive in the same folder as the Woofer Tester software. You can open the file with any word processor or the Windows Notepad program as shown in *Fig. 5*. Notice that "arbitrar.woo" contains three columns of data. The first column lists the frequency in hertz of each data point. The second column lists the impedance in ohms, and the third column lists the phase angle in degrees.

Record the frequency, F_L, and impedance value, Z_L, of the first impedance peak. (Note: Ideally, the phase angle should pass through zero degrees at F_L.)



arbitrar.woo - Notepad

File Edit Search Help

freq hertz	impedance ohms	phase degrees
5.00	5.11	27.29
5.26	5.20	28.22
5.52	5.41	29.62
5.78	5.51	30.80
6.04	5.64	31.90
6.30	5.72	32.81
6.56	5.91	33.89
6.82	6.03	35.65
7.08	6.22	36.69
7.34	6.31	37.47
7.60	6.47	38.43
7.86	6.70	39.34
8.12	6.91	41.01
8.38	7.15	42.03
8.64	7.41	42.59
8.90	7.78	43.26
9.16	8.11	43.73
9.42	8.59	43.88
9.68	9.09	43.90
9.94	9.42	43.02
10.20	9.64	43.77
10.46	10.35	44.51
10.72	11.02	44.69
10.98	12.06	44.68
11.24	13.14	43.43
11.50	14.50	41.39
11.76	15.98	38.38
12.02	17.76	33.89

FIGURE 5: File "arbitrar.woo."

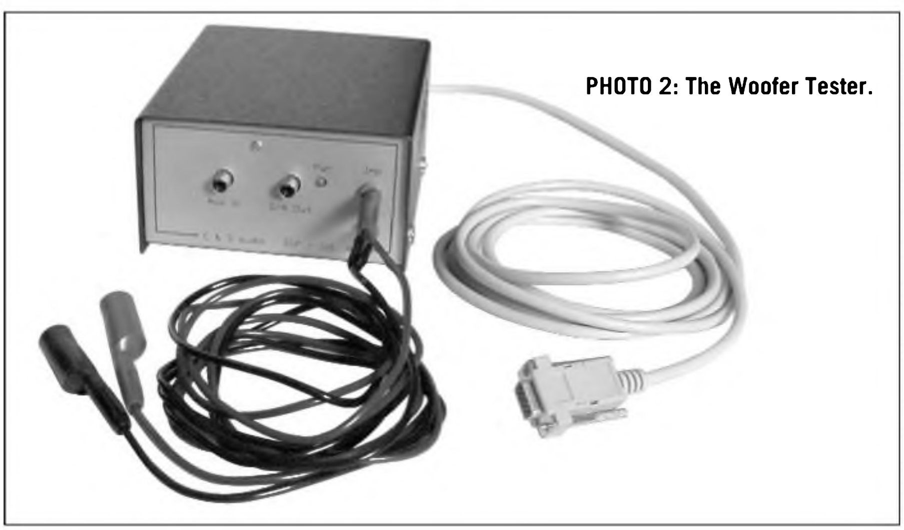


PHOTO 2: The Woofer Tester.

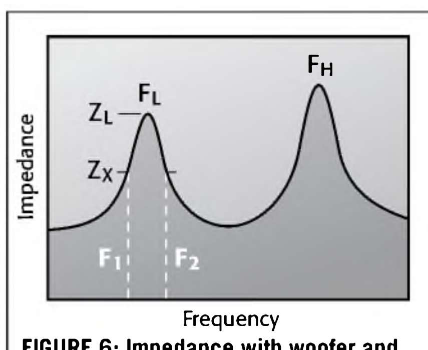


FIGURE 6: Impedance with woofer and PR on the test box.

Step 12: Calculate Z_X as follows:

$$Z_X = (Z_L/R_E)^{1/2} \times R_E$$

Z_X is in ohms.

Step 13: Find the frequency on each side of F_L where the impedance equals Z_X. These frequencies are labeled F_1 and F_2 (*Fig. 6*).

Step 14: Test the validity of F_L, F_1, and F_2 with the following comparison:

$$F_L = (F_1 \times F_2)^{1/2}$$

If you measured F_L, F_1, and F_2 accurately, then F_L should be within 1 or 2Hz of the square root of F_1 times F_2.

Step 15: Calculate the Q of the PRs' mechanical losses (Q_{MPRB}) in the box:

$$Q_{MPRB} = F_L \times (Z_L/R_E)^{1/2}/(F_2 - F_1)$$

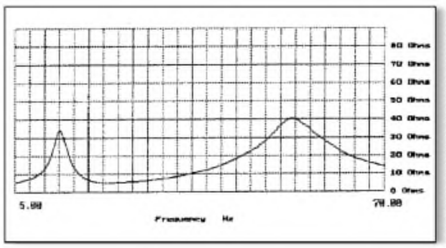


FIGURE 7: Measured impedance curve.

Step 16: Locate and record the frequency, F_H, of the second impedance peak. (Note: Ideally, the phase angle should pass through zero degrees at F_H.)

Step 17: Calculate the following:

$$F_B = (F_H{}^2 + F_L{}^2 - F_C{}^2)^{1/2}$$
$$H = F_H \times F_L/F_B$$
$$\delta = (F_C{}^2 - H^2)/(H^2 - F_{SB}{}^2)$$
$$F_{PR} = F_B/(\delta + 1)^{1/2}$$
$$V_{APR} = \delta \times V_B$$
$$C_{MPR} = V_{APR}/(c^2 \times \rho_0 \times S_{DPR}{}^2)$$
$$M_{MPR} = 1/((2 \times \pi \times F_{PR})^2 \times C_{MPR})$$

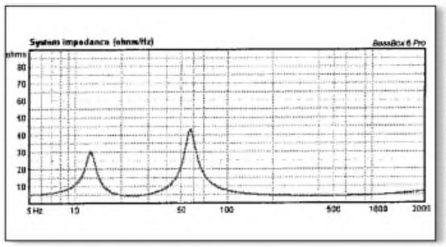


FIGURE 8: Predicted impedance curve.

$$Q_{MPR} \cong Q_{MPRB} \times F_L/F_{PR}$$
$$R_{MPR} = 1/(2 \times \pi \times F_{PR} \times C_{MPR} \times Q_{MPR})$$

This concludes the PR Measurement Procedure, with all of the small-signal parameters calculated. Definitions of many of the terms used in Step 17 are as follows:

α (alpha) The compliance ratio of the woofer and box.

c The velocity of sound in air (344.67m/s at ρ_0 listed later).

δ (delta) The compliance ratio of the PR and box.

F_B The system resonance frequency of the box in hertz (Hz).

F_C The resonance frequency of the woofer with the closed test box in hertz (Hz).

F_S The free-air resonance frequency of the woofer in hertz (Hz).

F_{SB} The resonance frequency of the woofer with the air load of the test box in hertz (Hz).

Q_{EC} The electrical resonance magnification of the woofer at F_C with a closed box.

Q_{ES} The free-air electrical resonance magnification of the woofer at F_S.

Q_{MPRB} The mechanical resonance magnification of the PR on the test box at F_L.

Q_{MS} The free-air mechanical resonance magnification of the woofer at F_S.

R_E The DC resistance of the woofer's voice coil in ohms.

ρ_0 (rho) The density of air (1.1955 kg/m^3 at 72°F and a barometric pressure of 29.92″ of Hg).

V_B The net internal volume of the test box in cubic meters (m^3).

SAMPLE PR MEASUREMENT

As an example, consider the measurement results of a 15″ (381mm) PR which I made recently. Begin by calibrating the Woofer Tester (Step 1). This does not need to be done before every PR measurement, but I recommend doing it once a day at the start of a measurement session. This step nulls the resistance of the test leads and their clip connectors.

I used a 12″ (305mm) woofer for this measurement, connected it to a Woofer Tester, and measured the following free air F_S, Q_{MS}, Q_{ES}, and R_E (Step 2):

$$F_S = 27.78\text{Hz}$$
$$Q_{MS} = 2.836$$
$$Q_{ES} = 0.25$$
$$R_E = 3.89\Omega$$

Next, measure the net internal volume of the test box (Step 3) with the following result:

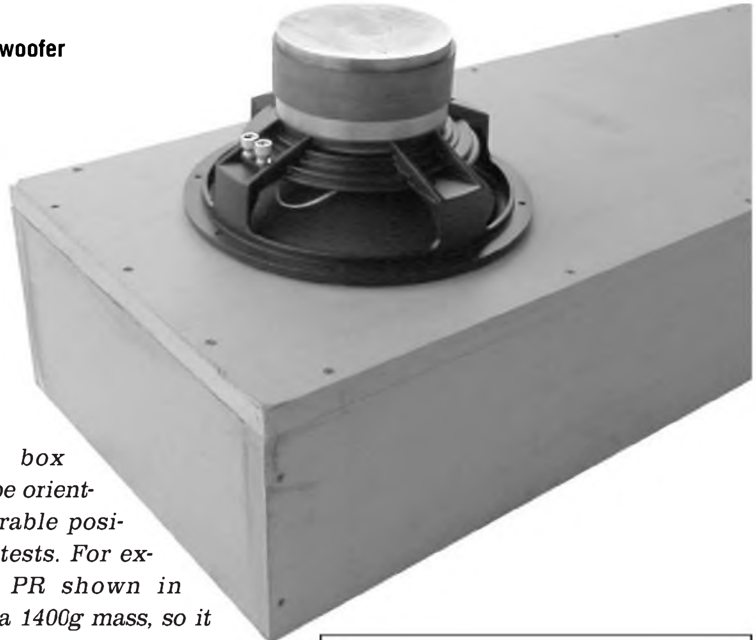
$$V_B = 0.0721\text{m}^3\ (2.545\text{ft}^3)$$

Then I mounted the woofer on the test box (Step 4), installed a solid plug in the PR mounting hole, and measured the closed-box F_C and Q_{EC} (Step 5) with the following results:

$$F_C = 50.60\text{Hz}$$
$$Q_{EC} = 0.478$$

Notes: It is important that the test box be as lossless as possible. It should be constructed of thick, dense walls which are carefully braced to minimize all leakage losses and unwanted resonance.

PHOTO 3: The woofer and test box.



The test box should also be oriented in a favorable position for the tests. For example, the PR shown in Photo 5 *has a 1400g mass, so it should not be tested in a horizontal position as shown in the photo because its heavy mass will deform the suspension during the test. Instead, you should set the test box on its side so that the PR is oriented vertically. I also recommend that you test the woofer in the same position for both the closed box and PR tests.*

Next, calculate α and F_{SB} (Step 6):

$$\alpha = (50.60 \times 0.478)/(27.78 \times 0.25) - 1 = 2.483$$
$$F_{SB} = 50.60/(2.483 + 1)^{1/2} = 27.11\text{Hz}$$

Note: You need to perform Steps 1 through 6 only once. They do not need to be repeated if more than one PR will be tested with the same woofer and test box.

Next, measure the piston diameter, D_{IAPR}, of the PR (Step 7):

$$D_{IAPR} = 0.327\text{m}\ (12.875″)$$

Then use it to calculate the piston area, S_{DPR} (Step 8):

$$S_{DPR} = 3.1416 \times (0.327/2)^2 = 0.084\text{m}^2$$

I removed the plug from the PR mounting hole and mounted the PR on the test box (Step 9), then made an impedance measurement with the "Arbitrary Impedance Plot" function of the Woofer Tester (Step 10). The resultant impedance graph is shown in *Fig. 7*.

Using the "arbitrar.woo" file produced by the Woofer Tester made during the

PHOTO 4: The piston diameter includes half of the surround.

impedance measurement, you can locate the frequency and maximum impedance of the first peak (Step 11).

$$F_L = 13.06\text{Hz}$$
$$Z_L = 34.7\Omega$$

Next, calculate Z_X (Step 12):

$$Z_X = (34.7/3.89)^{1/2} \times 3.89 = 11.62\Omega$$

Then I located the frequencies flanking F_L that had an impedance equal to Z_X (Step 13). They were:

$$F_1 = 9.94\text{Hz}$$
$$F_2 = 16.44\text{Hz}$$

I tested the results (Step 14) as follows:

$$(9.94 \times 16.44)^{1/2} = 12.78\text{Hz}$$

This was within 0.28Hz of F_L, which is less than 1Hz, confirming that you have a good measurement. If $(F_1 \times F_2)^{1/2}$ had varied by more than 1 or 2Hz from F_L, then you would have carefully repeated the measurement.

Next, calculate Q_{MPRB} as follows (Step 15):

$$Q_{MPRB} = 13.06 \times (34.7/3.89)^{1/2}/(16.44 - 9.94) = 6.00$$

Then locate the frequency of the second resonance peak (Step 16):

$$F_H = 54.10Hz$$

Finally, perform the remaining calculations (Step 17):

$$F_B = (54.10^2 + 13.06^2 - 50.60^2)^{1/2} = 23.17Hz$$

$$H = 54.10 \times 13.06/23.17 = 30.49$$

$$\delta = (50.60^2 - 30.49^2)/(30.49^2 - 27.11^2) = 8.376$$

$$F_{PR} = 23.17/(8.376 + 1)^{1/2} = 7.567Hz$$

$$V_{APR} = 8.376 \times 0.0721 = 0.6039m^3 \ (21.33ft^3)$$

$$C_{MPR} = 0.6039/(344.67^2 \times 1.1955 \times 0.084^2) = 0.0006026 \ m/N$$

$$M_{MPR} = 1/((2 \times 3.1416 \times 7.567)^2 \times 0.0006026) = 0.7341 \ kg$$

$$Q_{MPR} \cong 6.00 \times 13.06/7.567 \cong 10.36$$

$$R_{MPR} = 1/(2 \times 3.1416 \times 7.567 \times 0.0006026 \times 10.36) = 3.369 \ kg/s$$

As a final check, enter the woofer, PR, and test box information into a box design program and model the impedance response (*Fig. 8*). The result closely matches the original impedance measurement, confirming the validity of the calculations.

Note: Don't be fooled by the different appearance of Figs. 7 *and* 8. *The Woofer Tester uses a linear frequency scale, while the box modeling program*

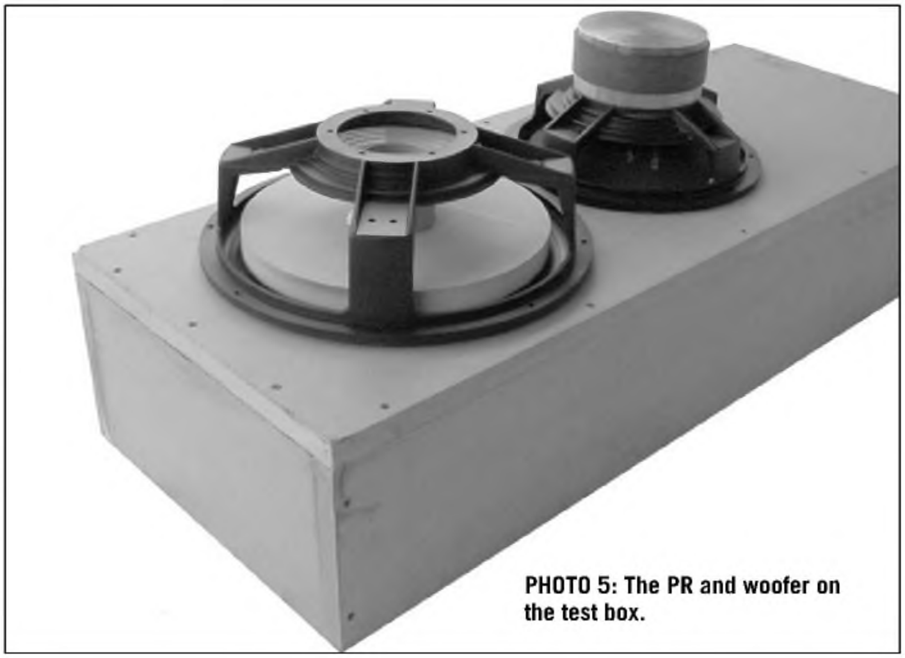


PHOTO 5: The PR and woofer on the test box.

that produced Fig. 8 *uses a logarithmic scale. This accounts for the varying widths of the peaks. The important thing is the location and relative height of the peaks, which correlates very well. Slight variations in their height is usually due to losses in the test box, driver, and PR.*

The PR Measurement Procedure in this article requires many manual calculations. Although several measurement systems (such as the Woofer Tester) are available for drivers which can alleviate the computational work load of driver parameter calculations, I am not aware of an existing measurement system for PRs.

For this reason, my box design program, BassBox Pro, includes a measurement procedure to help with PR parameter calculation (using a different test setup and procedure). An undocumented feature was recently added which incorporates the same PR Measurement Procedure explained in this article. It will step you through this procedure with a Woofer Tester and prompt you for the required inputs. All calculations are performed automatically.

To utilize this undocumented feature, you will need the latest copy of BassBox Pro (version 6.0.12, dated Jan. 25, 2001, or later). Licensed users can download a free update from the Harris Tech website at www.ht-audio.com.

To access the undocumented feature and run the PR Measurement Procedure of this article, open the "Passive Radiator Test Procedure" window by selecting "Passive Radiator" from the "Test" menu. Hold down the Shift+Ctrl+Alt keys and click with the left mouse button on the test setup picture that appears in the upper half of the window.

To switch back to BassBox Pro's standard PR test procedure (which requires voltage measurements), simply repeat the undocumented trigger: Shift+Ctrl+Alt+ left mouse click on the test picture.

MULTIPLE PRS

A PR must displace more air than the woofer it augments. This means that the PR will need to have a larger piston diameter and/or a larger excursion limit than the woofer. One guideline is to select a PR whose diaphragm sweeps a volume of air approximately 2.5 times larger than that of the woofer at the woofer's excursion limit.

One way speaker designers have accomplished this is to use two or more identical PRs in a speaker. When more than one PR is used, you can calculate the net PR parameters as follows:

F_{PR} doesn't change.
Q_{MPR} doesn't change.
$\text{Net } V_{APR} = V_{APR} \times N$
$\text{Net } C_{MPR} = C_{MPR}/N$
$\text{Net } M_{MPR} = M_{MPR} \times N$
$\text{Net } R_{MPR} = R_{MPR} \times N$
$\text{Net } S_{DPR} = S_{DPR} \times N$
X_{MAXPR} doesn't change.
X_{MECHPR} doesn't change.

where N equals the number of PRs.

If your box design program does not make internal adjustments for multiple PRs, then you will need to calculate these net values and use them when you model the speaker.

An important consideration when using multiple PRs is the need to add quite a bit of mass to their diaphragms. For example, a speaker with two identical PRs will require four times the total PR moving mass (M_{MPR}) as the same speaker with just one PR. Care should be taken when adding a lot of mass to avoid exceeding the capability of each PR's suspension.

CHANGING M_{MPR}

Tuning a speaker with a PR is accomplished by adding or subtracting mass from the PR's diaphragm. This mass directly adds or subtracts from the M_{MPR} parameter. Changing the mass also causes F_{PR} and Q_{MPR} to change. You can use the following equations to calculate new values when M_{MPR} is changed:

$$F_{PR} = (1/(M_{MPR} \times C_{MPR}))^{1/2}/(2 \times \pi)$$
$$Q_{MPR} = M_{MPR} \times 2 \times \pi \times F_{PR}/R_{MPR}$$

These equations assume that the same MKS units used earlier are also used here. F_{PR} is in Hz. M_{MPR} is in kg. C_{MPR} is in m/N. R_{MPR} is in kg/s.

UNITS CONVERSION

Length

Inches (in) × 0.0254 = Meters (m).
Feet (ft) × 0.3048 = Meters (m).
Millimeters (mm)/1000 = Meters (m).
Centimeters (cm)/100 = Meters (m).

Area

Square Inches (in^2) × 0.0006452 = Square Meters (m^2).
Square Feet (ft^2) × 0.09291 = Square Meters (m^2).
Square Millimeters (mm^2)/1000000 = Square Meters (m^2).
Square Centimeters (cm^2)/10000 = Square Meters (m^2).

Volume

Cubic Inches (in^3) × 0.00001639 = Cubic Meters (m^3).
Cubic Feet (ft^3) × 0.02832 = Cubic Meters (m^3).
Liters (l)/1000 = Cubic Meters (m^3).

Frequency

Kilohertz (kH) × 1000 = Hertz (Hz).
Megahertz (MH) × 1000000 = Hertz (Hz).

Mass

Ounces (oz) × 0.02835 = Kilograms (kg).
Grams (g)/1000 = Kilograms (kg).

Compliance

Inches per Pound (in/lb)/175.1 = Meters per Newton (m/N).
Micrometers per Newton (µm/N)/1000000 = Meters per Newton (m/N).
Millimeters per Newton (mm/N)/1000 = Meters per Newton (m/N).
Centimeters per Newton (cm/N)/100 = Meters per Newton (m/N).

Mechanical Resistance

Pounds per Second (lb/s) × 0.4536 = Kilograms per Second (kg/s).
Mechanical Ohms (Mohms) = Kilograms per Second (kg/s).

FINAL COMMENTS

PRs can offer several advantages over vents. For example, PRs don't suffer from "pipe" resonance like vents, and internal reflections can't escape through a PR as easily as a vent. Since PRs are tuned by adding and subtracting mass, they can tune a small box to a lower frequency than a vent, which, at low frequencies, can become too long to fit within the box. Some simply prefer the "sound" of a PR rather than of a vent.

On the other hand, PRs can often have a higher group delay, and some claim this can audibly reduce the low-frequency transient response of a speaker (although there is disagreement on this point). Vents also do not suffer suspension nonlinearities like some PRs. Vents are much less expensive than PRs, and when a vented box is designed well, it should not suffer audible vent "coloration."

Whichever view you prefer, one thing is certain—PR speaker design and evaluation is all guesswork without accurate PR measurements. The PR Measurement Procedure presented in this article should help to satisfy that requirement. ❖

RESOURCES

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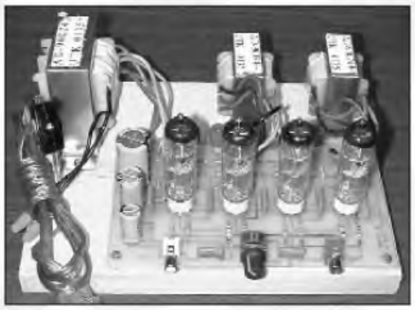

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FM Tuners: NAD and Parasound

Reviewed by Charles Hansen and Nancy and Duncan MacArthur

PHOTO 1: The NAD C 420 stereo tuner.

NAD C 420 stereo AM/FM tuner. NAD Electronics International, 633 Granite Court, Pickering, Ontario, Canada L1W 3K1, (800) 263-4641, www.NADelectronics.com. $249 U.S. Dimensions: 435mm W × 70mm H × 285mm D. Net weight: 8.8 lbs (4kg). Warranty: two years parts and labor.

Parasound TDQ-150 stereo tuner. Parasound Products, Inc., 950 Battery St., San Francisco, CA 94111, (415) 397-7100, www.parasound.com. $269 U.S. Dimensions: 9.5" W × 1.75" H × 7" D. Net weight: 4 lbs. Ten-year limited warranty.

The NAD C 420 and Parasound TDQ-150 AM/FM stereo tuners represent the budget end for high-end tuners. These two manufacturers' higher-priced tuners (NAD C 440, $349; and the full-size Parasound TDQ-1600, $400) offer better FM sensitivity and lower audio distortion specifications. However, if you don't require fringe area reception, these two tuners represent excellent values.

NAD C 420

Photo 1 shows the C 420 front panel. Just above the power switch on the left is a small green LED indicator, which turns amber in Standby mode. A row of six push buttons selects Blend, Memory, FM Mute/Mono, AM/FM, Display modes, and Preset/Tune functions. The white fluorescent display screen shows the band and frequency of the station being tuned, FM stereo, whether the station is a memory preset, and whether Blend and FM Mute are engaged. It also has a vertical seven-bar-graph signal strength indicator.

Pressing the display button cycles the display to RDS program service. Another push cycles it to RDS text. The display then returns to the default mode after a few seconds. The infrared sensor is located on the left side of the display window. A rocker switch on the right side of the unit is for the Preset/Tune functions.

The tuner chassis is black painted steel, and the cover is held on with five screws. The front bezel is plastic and the unit sits on four plastic feet with elastomer rings on the bottom. There is adequate finger space under the unit to easily lift it.

The rear panel (not shown) has the attached two-prong polarized AC line cord, a +12V trigger jack, the "NAD-Link" input and output jacks, stereo audio jacks with gold-plated shells and tin center contacts, a two-conductor AM antenna connector, and a threaded 75Ω F-type FM antenna jack.

The C 420 provides up to 30 station presets, which you can use in any combination of AM and FM stations. FM Mute and FM Blend status information is stored with each FM preset. Empty presets are skipped over during tuning to provide quicker access to the stored stations.

The preset information is stored in non-volatile EEPROM memory, which provides permanent power-off storage. Many tuners use "super capacitors" to hold the preset memory in CMOS chips for a limited time—maybe a month or so. My own NAD 4155 uses a now-tired super cap, and it must be powered up at least once a week to maintain all the presets.

FM Mute/Mode switches the C 420 to mono and disengages the muting circuitry so you can manually tune weak stations. FM Blend provides a means to automatically reduce noise and hiss on weak stations while still retaining some stereo separation. Once the signal level drops below a certain threshold, it will revert to mono. The tuning increments for FM are in

PHOTO 2: Front view of Parasound TDQ-150.

steps of 50kHz. The AM tuning steps are 10kHz (9kHz for the 230V version).

The RDS PS (Program Service) automatically displays the name of the radio station you are listening to. The RDS RT (Radio Text) button displays any additional information broadcast by the radio station, such as program format, song titles, and so on.

You can operate the C 420 with one of NAD's system remote controls via the front panel IR sensor, or through the rear panel NAD-Link jacks. You can also switch tuner AC power from amps, preamps, and AV processors that use the 12V-trigger system.

Eight pages of the 43-page manual are devoted to instructions in English. Other languages are French, German, Spanish, Italian, Portuguese, and Swedish. Programming isn't all that intuitive, so there is a two-page section on storing, recalling, and labeling presets. Ancillary items include an AM loop antenna, the usual FM dipole antenna, and a set of generic RCA audio interconnects.

PARASOUND TDQ-150

The TDQ-150 (*Photo 2*) is half the width, height, and depth of most conventional audio components. It is designed for main or remote-zone use in a custom sound system installation where space is at a premium. The unit sits on four plastic feet with foam-rubber inserts. Holes are provided on the front plate for rack mounting. The unit is very light and easy to move.

The front panel has six push buttons: On-Off, FM-AM, Preset Up/Down, and Tune Up/Down. The yellow backlit LCD display shows band and frequency, the selected station preset, and a stereo/mono indicator.

The rear panel (not shown) has an IEC power receptacle with integral fuse holder, external IR control port, +12V trigger jack, audio output jacks with gold-plated shells and tin center contacts, a two-conductor AM antenna connector, and a threaded 75Ω F-type FM antenna jack. The third pin of the AC receptacle is not connected to the chassis.

The TDQ-150 comes with a full-function remote control with separate on and off codes, "Zpre" Zone Preamplifier control, and AC line and DC triggering capabilities. To store memory presets you must use the remote. While the operation is a bit more intuitive than the NAD C 420, I still prefer the car radio method: tune the station, press the preset button for 5 seconds, voilà, you're done!

You can assign up to 30 presets to any AM or FM stations. The CMOS memory retains preset stations by means of a super-capacitor for up to 30 days without AC power. RDS program service data or text is not displayed.

FM mono is automatically engaged below 15μV RF signal level. The tuning increments for FM are the odd 200kHz U.S. spacings. The AM tuning steps are 10kHz.

The ten-page manual is entirely in English. Ancillary equipment includes the remote control with batteries, power cord, FM dipole antenna, 300Ω-to-75Ω balun, and AM loop antenna with self-adhesive bracket.

INSIDE THE NAD C 420

Photo 3 shows the NAD C 420 with the cover removed. The power transformer occupies the left rear of the chassis, and connects to the switch/LED board in the left front. The display/control PC board sits behind the front panel. Occupying most of the right side of the chassis is the large single-sided phenolic tuner board. A schematic was not furnished with the unit.

The display board connects to the tuner board through three ribbon cables, and the power transformer secondary connects via

PHOTO 3: Interior view of NAD C 420.

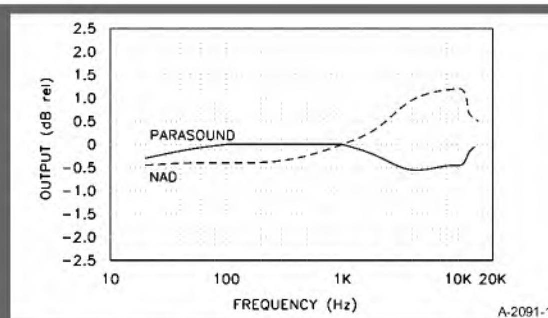


FIGURE 1: Frequency response—FM tuners.

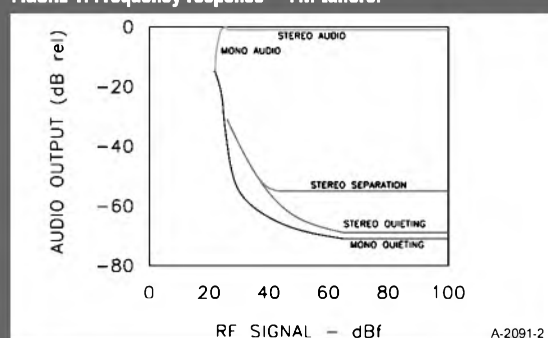


FIGURE 2: FM quieting—NAD C 420.

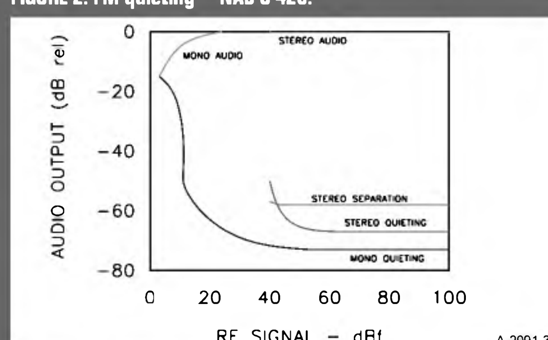


FIGURE 3: FM quieting—Parasound TDQ-150.

TABLE 1

NAD C 420 SPECIFICATIONS AND MEASURED RESULTS

FM SECTION SPECIFICATIONS	NAD C 420	MEASURED RESULTS
Usable sensitivity, mono	2.2μV, IHF	22dBf, 3.4μV (75Ω)
Usable sensitivity, stereo	16μV, IHF	36dBf, 17μV (75Ω)
Mono quieting @ 50dB	23.3dBf, 4.0μV (75Ω)	25dBf, 5μV (75Ω)
Stereo muting/blend		37dBf
Stereo quieting @ 50dB	40.2dBf, 28μV (75Ω)	38dBf, 22μV (75Ω)
S/N, A-wtd, 65dBf, mono		71dB
S/N, A-wtd, 65dBf, stereo		69dB
Frequency response	30–15kHz, ±1.5dB	20–15kHz, +1.2, −0.5dB
THD, 1kHz mono	0.20%	0.087%
THD, 1kHz stereo	0.30%	0.29%
THD, 100–10kHz, mono		0.22%
Separation, 1kHz	>45dB	54dB
Capture ratio, 45dBf	2.4dB	
Image rejection, 400kHz	>60dB	
Auto search threshold		27dBf
Output impedance		420Ω, 1kHz
Output level		550mV, 1kHz

CRITIQUE—NAD C 420, PARASOUND TDQ-150, AND ADCOM GFT-555II

By Nancy and Duncan MacArthur

Some years ago we moved toward a minimalist audio system: one source, one volume control, one amplifier, and one set of speakers. In the process we eliminated much equipment, including our old Dynaco FM-5 tuner.

In the intervening years we forgot how much fun playing with a tuner could be. After burning in the NAD C 420, the Parasound TDQ-150, and the ADCOM GFT-555II, we spent hours flipping through the FM band, finding a huge variety of music—some pieces familiar, some not. We ran across Navajo chants, opera, rock, symphonies, and Spanish music, not to mention the local deejay who termed himself the "Commissar of Your Radio Commune."

Eventually the situation got out of hand. When the public radio station ran a long program of Colombian music, I found myself dancing around the living room to the Swamp Cumbia. Halfway through the song, I turned and ran smack into a scandalized eleven-year-old.

"Cut it out, Mom," he growled.

"Whatever for?" I asked.

He considered the matter briefly. "'Cause if you don't, I'm gonna get out the video camera and put you on the Internet."

Oh. Perhaps it was time to stop dancing and start writing this review.

REVIEWING TUNERS

An FM tuner review requires a different listening approach from reviews of other components. Local FM stations rarely provide a clean, uncompressed signal. Even if their signal quality is good, they typically don't play the same piece repeatedly for the convenience of reviewers. Obviously, under these circumstances a listening critique of tuners can't be as rigorous as critiques of other components. But at least three aspects of tuner performance beyond lab test results have some importance.

First, listening tests may reveal some sonic differences. If the tuners all sound the same, we'll say so; when we hear obvious differences, we'll point them out. Second, if the sonic signatures are similar, the choice may devolve upon price, features, and operational convenience. Third, a brief comparison between these tuners and other sources may be useful. Our goal is to give you an idea of what to expect before making a purchase.

The NAD C 420 and the Parasound TDQ-150 arrived from the manufacturers by way of Chuck Hansen's lab. The ADCOM GFT-555II was an older model borrowed from Ed Dell for use as a baseline.

We burned in each tuner with an antenna signal sufficient to exceed its muting threshold for at least 100 hours. Tuners are significantly easier to burn in than many other components because they do not have moving parts or large sources of heat. The sound of the NAD and Parasound changed significantly over the burn-in period. (Presumably the ADCOM had burned in long ago.)

Following burn-in, we listened to each tuner in turns using the same set of popular and classical stations. As might be anticipated, the popular stations provided a horribly compressed signal. (On the other hand, they frequently played songs we knew well.) The rest of the reproduction system was identical in each case, and all three tuners were plugged into the same circuit of a Monster Cable HTS 2000 power conditioner.

Each tuner also receives the AM band, although we concentrated on FM performance in this review.

ANTENNAS

The reception quality of any tuner critically depends on the antenna connected to it. For all our serious listening we used a multi-element roof-mounted Radio Shack special.

Out of curiosity we briefly connected the twin lead dipoles supplied in each box to their respective tuners. Because most of the stations we listen to are about a hundred miles away, we didn't anticipate satisfactory performance from the dipoles. *Table 1* lists the number of stations in our area that each tuner received cleanly. When we say *cleanly*, we mean with lack of noise and sibilant distortion. (Many more stations were intelligible from a DXer's standpoint.)

Practically speaking, all these sensitivities were comparable except for the NAD's reception with the indoor dipole. All three tuners exhibited good sensitivity when attached to a "real" antenna.

Even if your system hasn't suffered from them before, the introduction of an external antenna that is grounded for safety reasons often causes a ground loop. The antenna safety ground is likely to be located far from the audio system ground and may easily be at a different potential. These problems may be exacerbated by a grounded tuner (such as the Parasound) but can be present even if the tuner itself is ungrounded. Numerous companies, including Jensen, Mondial, Tributaries, and MIT, manufacture products intended to break these ground loops.

APPEARANCE

The NAD C 420 is a rarity among components: a good-looking black box, well proportioned with an elegant oval display window. Its preset/tune rocker echoes the shape of its display window. The most salient feature of the Parasound's appearance is its tiny size—roughly half as wide, half as high, half as deep as the other tuners. (If you have a 17″ rack, you could mount the Parasound side-by-side with its matching preamplifier.) The ADCOM GFT-555II is a standard chunky black box; its most noticeable feature is a long row of buttons on the front panel.

Popping open the Parasound reveals a single, densely packed, PC board. The board dominates the interior of the Parasound and fills the available space. Although the Parasound is much smaller than the other tuners, it weighs about the same.

The NAD's main PC board is well laid out and less densely packed. It fills about half the enclosure. As you might expect in an older component, the ADCOM's main board nearly fills its box. Although all three tuners are well laid out and cleanly constructed, the NAD and ADCOM probably would be easier to service or modify due to the extra "elbow room" within the enclosure.

EASE OF USE

Both the Parasound TDQ-150 and the ADCOM GFT-555II were easy to use. The Parasound has five front-panel buttons and includes a remote. To set the presets you must use the remote. (You can tune the presets sequentially using buttons on the front panel). The remote is also handy for changing stations and accessing the presets in random order; in addition, it has preamplifier controls intended for use with a matching preamplifier (also half rack width). The Parasound uses the U.S. frequency interval of 0.2MHz and can be tuned rapidly across the FM band.

The 30 presets on this tuner are accessible in sequence by the up-and-down buttons on the front panel. We preferred the random access ability provided by the remote. The tuning buttons on the Parasound operate in two modes. A short press on one of the tuning buttons will change the frequency by 0.2MHz. According to the manual a continuous press will tune to the next strong station; however, our sample would not stop at any station regardless of strength. Like the Energizer Bunny®, it kept going and going and going.

The ADCOM GFT-555II has only 16 presets (8 AM and 8 FM) but has an individual front-panel button permanently assigned to each. It thus provides totally random access for setting and tuning the preset frequencies. The ADCOM didn't come with a remote, and its manual makes no mention of remote capability. It tunes in 0.1MHz intervals.

The ADCOM has three modes of tuning. A short press on one of the tuning buttons will change the frequency by 0.1MHz, and a continuous press will tune continuously. Activating the "FM scan" switch will stop the tuning at the next strong station. These features are straightforward and easily understood; describing them here takes more time than learning to use them.

The NAD C 420 also features 30 presets that are accessible sequentially from the front panel. A remote control is optional with the NAD: it's the same remote supplied with the matching NAD preamp, and the manufacturer doesn't want to charge you twice. If you buy the tuner but not the preamp, we would strongly recommend purchasing the remote separately.

Many of the NAD's features were not self-explanatory: we frequently had to refer to the manual. The technique for erasing presets, which required multiple timed pushes of two buttons, seemed particularly opaque.

Different buttons operated in different ways. Some toggled front-panel lights, and some didn't. We had to push some for a certain number of seconds to enable one feature and a different number of seconds to enable another.

The NAD uses a seek mode of tuning: it stops at every strong station whether you want it to or not. This characteristic becomes less important once the presets are set but can lengthen the process of moving from one end of the band to the other.

This tuner also incorporates RDS, a useful feature *if* nearby stations transmit RDS information and *if* the

TABLE 1
NUMBER OF STATIONS RECEIVED CLEANLY FOR EACH TUNER/ANTENNA COMBINATION.

	OUTDOOR ANTENNA	INDOOR DIPOLE
NAD C 420	25	19
Parasound TDQ-150	29	26
ADCOM GFT-555II	27	25

TABLE 2
PARASOUND TDQ-150 SPECIFICATIONS AND MEASURED RESULTS

FM SECTION SPECIFICATIONS	PARASOUND TDQ-150	MEASURED RESULTS
Usable sensitivity, mono		10dBf
Mono quieting @ 50dB	11.0dBf, 1μV (75Ω)	11dBf, 1μV (75Ω)
Stereo muting (Fixed)		40dBf
Stereo quieting @ 50dB	37.2dBf, 20μV (75Ω)	40dBf, 27μV (75Ω)
S/N. A-wtd, 65dBf, mono		74dB
S/N. A-wtd, 65dBf, stereo	>74dB	68dB
Frequency response	30–15kHz, ±1dB	20–15kHz, +0, –0.5dB
THD, 1kHz mono	0.08%	0.09%
THD, 1kHz stereo	0.20%	0.15%
THD, 100–10kHz, mono		0.28%
Separation, 1kHz	50dB	58dB
Separation, 100–10kHz	40dB	
Alt. ch. selectivity, 400kHz	80dB	
Capture ratio, 45dBf	<1.5dB	
AM suppression	60dB	
Auto search threshold		20dBf
Output impedance		600Ω, 1kHz
Output level		580mV, 1kHz

eight individual wires. The shielded MOSFET RF front end sits just behind the two antenna connectors. A Sanyo LA7218 and LA1837 chip set handles PLL frequency synthesis and AM/FM tuning and RDS decoding operations. The EEPROM preset storage memory chip is under the wide ribbon cable.

The FM tuner appears to have a three-stage IF (intermediate frequency) section. A pair of emitter-follower audio transistors feed the audio jacks.

INSIDE THE TDQ-150

Photo 4 shows the interior of the TDQ-150 tuner. The power transformer sits on the left side of the chassis, with the display/control PC board behind the front panel. The compact double-sided epoxy tuner board occupies most of the chassis. A schematic was not furnished with the unit.

The display board connects to the tuner board through three Molex-style connectors, and the right side wiring loops through a toroidal ferrite core. The transformer secondary is hard-wired to the PC board, where a pair of fuses deliver low-voltage AC to the power supply. The power transformer primary remains energized when the tuner is plugged in. The front panel On-Off switch operates a power-supply relay that switches the low-voltage secondary. Linear regulators provide +5V DC and ±12V DC to the circuitry.

The shielded MOSFET RF front end sits just behind the AM antenna connector. A Sanyo LA3401 and LA1266 chip set handles PLL frequency synthesis and AM/FM tuning operations.

The FM tuner appears to have a two-stage IF section, with ground braid straps connecting the IF transformer cases to the RF front-end shield. Parasound describes it as an ultra-wideband IF section for low distortion, flat response, and a wide dynamic range. Several audio transistors are located near the output jacks, so the TDQ-150 also appears to have a discrete audio output stage.

MEASUREMENTS—NAD FM SECTION

I did not run any tests on the AM sections of either tuner, except to make sure they were functional.

The C 420 does not invert polarity. The output impedance at 1kHz was 420Ω, delivering 550mV into a load of 100k.

The frequency response (*Fig. 1*) was within +1.2, −0.5dB from 20Hz to 15kHz. The response curve above 10kHz may not be entirely accurate. In this area there are three filter responses: the 75μs pre-emphasis and steep 16kHz LP filter on the audio that is fed to the FM signal generator, and the 75μs de-emphasis in the tuner under test. Audio crosstalk perfor-

signals are changing—in a car, for example. Neither possibility holds true in a fixed installation in the U.S. (Only one station in our area broadcasts an RDS signal.) Clearly the NAD is intended for European markets as well as the U.S. The C 420 has RDS capability and a multilingual manual, and it tunes in 0.05MHz increments, requiring four steps between U.S. stations.

SOUND JUDGMENTS

Given an adequate antenna, all three tuners produced a completely acceptable sound. One small exception: all three grated on us occasionally during operatic soprano solos, but we ascribed this effect to the listeners' taste rather than to the tuners. None of the tuners bested our reference SACD player (Sony SCD C333ES) sonically, a not-unexpected result.

Although all three tuners sounded pleasant, each had a distinct sonic signature. The sound of the NAD could best be characterized as inoffensive: most defects in reproduction were subtractive rather than additive. The midbass response of the NAD was slightly loose or boomy; this effect was especially apparent on rock recordings. The NAD's response seemed a bit recessed at both frequency extremes, but this effect was small and may have been due to the source material.

The NAD presented a good soundstage, extending from speaker to speaker, but the images within this stage were not particularly well defined. The NAD's sound seemed slightly compressed, even more compressed than the source material. This effect was especially noticeable when we listened to classical music stations, which tend to transmit less compressed signals.

The Parasound produced a precise, detailed sound. I characterized it as having detail and clarity, while Duncan saw it as having a slight high-frequency emphasis. In any event, the high-frequency response was clean, extended, and never fatiguing.

The Parasound's imaging was sharp and well-defined. Its soundstage was similar to the NAD's; however, the detail and clarity of the Parasound extended to the spatial characteristics as well. With well-recorded material the images of individual instruments were well separated and sized appropriately. The Parasound seemed capable of reproducing as much dynamic information as was transmitted.

In comparison, the ADCOM presented a smooth, natural sound. No frequency region was missing or particularly emphasized. The soundstage was very wide, occasionally extending beyond the speakers. The ADCOM's imaging was somewhat smeared: each instrument appeared to originate from a space several feet across rather than a single location.

The dynamics produced by the ADCOM were good without being obtrusive. Again, with this tuner we had the impression that the dynamic range was limited more by the transmitted signal than by the tuner.

FINAL THOUGHTS

NM: All three tuners had a pleasant sound; none produced fatigue even after hours of listening. To choose among them, focus on which specific characteristics mean most to you. If looks are your top priority, buy the NAD. If you plan to make extensive modifications, pick the NAD or the ADCOM—you'll have more working room. If you want a spare, precise sound with good imaging and good dynamics, go for the Parasound. If you prefer a fuller sound, again with good dynamics, look for the ADCOM on the used-equipment market.

DM: As with most decisions, the choice of "best" tuner in this group depends on which features are most important to the buyer. Both the NAD and the Parasound are currently available; you would need to purchase a used ADCOM. Both newer tuners have automation features (remote control, DC switching, and so on) that are not available on the ADCOM. The ADCOM was designed as a stand-alone stereo tuner, while both the NAD and the Parasound seemed intended to be part of a home theater system.

Both the NAD and the Parasound are visually interesting, although in different ways. I like the small size of the Parasound, but this same size would make it less appropriate in a stack of 17″ components (unless you pair it with Parasound's matching preamp, amplifier, or phono preamp.) The ADCOM is supremely easy to use but lacks some features in comparison with its remote-controlled brethren. (If you purchase the NAD, I strongly recommend buying the optional remote control—use of the presets, in particular, is arcane when using the front-panel controls.)

Sonically each tuner offers a different picture. Depending on the program material, I alternately preferred the sound of either the Parasound or the ADCOM. The Parasound possesses good imaging and a very detailed sound. It works well in a tube-based system, such as ours, which has a smooth high-frequency response and no need of additional bass emphasis.

The ADCOM is fuller and arguably more natural but lacks some detail when compared to the Parasound. Although the NAD didn't match well with our system, it might be better matched to a solid-state system that could use a bit more bloom in the lower midrange.

mance was 54dB at 1kHz.

THD+N at 1kHz was 0.087% mono, and 0.29% stereo. During distortion testing, I engaged the test set 80kHz low-pass filter to limit the out-of-band noise. THD from 100–10kHz, mono, did not exceed 0.22%.

The mono audio distortion residual waveform shows mainly the second harmonic, overlaid with noise.

The C 420 tuner quieting characteristics are shown in *Fig. 2*. I had to switch to FM/Mute mode to measure mono sensitivity. The station auto search threshold, where it would stop at the signal generator's frequency during a station scan, was 27dBf. There was no overload at the maximum RF input of 100dBf. The audio output disappeared when I moved the FM test signal ±50kHz to either side of the tuned center frequency. Stereo audio output was –0.9dB below the mono audio at 65dBf signal strength.

The seven bars of the display's tuning strength meter change at the RF signal levels are:

Bar 1 (always on)
Bar 2 at 25dBf
Bar 3 at 30dBf
Bar 4 at 33dBf
Bar 5 at 36dBf
Bar 6 at 39dBf
Bar 7 at 41dBf

PHOTO 4: Interior view of Parasound TDQ-150.

MEASUREMENTS—PARASOUND FM SECTION

Again, I did not run any tests on the AM section, except to verify its operation. The TDQ-150 inverted polarity; the tuner's audio output being out of phase with the composite audio signal fed to the FM signal generator's varactor modulator. The output impedance at 1kHz was 600Ω, delivering 580mV into a load of 100k.

The TDQ-150 tuner frequency response (*Fig. 1*) was within +0, –0.5dB from 20Hz to 15kHz. Crosstalk performance at 1kHz measured 58dB.

THD+N at 1kHz was 0.09% mono, and 0.15% stereo. THD from 100–10kHz, mono, did not exceed 0.28%.

The mono audio distortion residual waveform again showed mainly the second harmonic, overlaid with noise.

The TDQ-150 quieting characteristics are shown in *Fig. 3*. The station auto search threshold was 20dBf. There is no method for manually switching the FM/Mute mode, and the tuner switches back to mono at a high 40dBf, right where the –50dB stereo quieting occurs.

The audio output didn't disappear until I moved the FM test signal ±150kHz to either side of the tuned center frequency, probably reflecting the reduced sensitivity of the two-stage IF design. This is not an issue with the U.S. 200kHz FM radio station spacings. There was no overload at the maximum RF input of 100dBf. Stereo audio output was only –0.3dB below the mono audio at 65dBf signal strength. ❖

CORRECTION

It's great to see my article ("A New Class-B Amplifier," April 2002) in print! There are three small mistakes, most likely due to my lousy proofreading:

1. Page 7, middle column, line 13 should read "for the EL34 (Fig. 1)." Note that Figure 1 is EL34, Figure 2 is 6550 (evident from the picture of tube on the figure).
2. Figure 7 is low-power distortion at 20kHz, not 2kHz.
3. In Figure 10 I refer to a "feedback switch," which is, of course, not shown on the schematic (Figure 4). I added the switch later, and it is merely a variable tapped resistor in place of R12. I did this so I could fool around with speaker damping. Three values of feedback are provided: 15, 9, and 6dB.

Now I have to get working on an article for the preamp and tuner.

Richard Modafferi
Vestal, N.Y.

6L6 POWER

One night in the mid '70s, my group used a cabinet with two 15″ woofers to augment the low fundamental of the Fender bass. I plugged it into a jack next to the one in the amp feeding the standard speaker, one with eight 10″ drivers (with plenty of upper bass punch, but no depth). The amp had four 6L6 tubes (I'm pretty sure), was rated at 100W (true average, wrongly called "RMS"), but sounded like 300 (real watts). Since we played "to the wall" of volume, the amp was almost always clipping and probably outputting 300 actual watts of saturated tube power.

Anyway, combining the two 15s did add nice, deep bass down to that low-E fundamental, as those four 6L6s happily drove eight 10″ plus two 15″ drivers. (Impedance? What's that?) For a while, that is. Midway into the second set, I smelled smoke, followed quickly by a loss of deep bass. The 6L6s were still happy, but those 15″ woofers had gone to the big nightclub in the sky!

I don't remember the brand of bass amp, but it lasted probably 10 plus years, with constant saturated output for about 12 hours a week. That's about one solid year of saturated 300W output*. Then during one particularly alcohol-fueled song, the output transformer blew,

"possibly" because we had previously "fixed" a blown fuse with aluminum foil!

But not once in those 10 plus years had a tube failed. This suggests a good way to test your home tube amp for ruggedness: just get an electric bass with preamp and speaker, and play through your prized hi-fi tube amp, at saturated output, along with the music you listen to. Do this for 10 years.

If no tubes blow (or anything else), you know you have a rugged amp! And as a fringe benefit, you'll have learned to play the bass!

* 2600kWh or 3500 horsepower-hours; enough energy to propel a car using 50hp @ 60 MPH for 4200 miles! (Is that why speaker units are called drivers?)

Dennis Colin
Gilmanton, I.W., N.H.

VENEER SOURCE

In October 2001 *aX*, there was an article about the Audax home theater system. I was wondering about the veneer that was used and where it could be obtained. I am from Sweden, and it seems that veneer (self-laminating veneer) isn't that easy to find in this small country.

Could you help me out with this?

Christopher Swenson
Sweden

John Calcote responds:

Chris has a great question. My first suggestion is to simply search the Internet for references to keywords such as "veneer" and "3M," but after doing some extensive searching on the internet myself, I realized that it's not as easy to come by as I'd originally thought. I called my supplier and requested manufacturer information for the product. The company is Veneer Technologies in Newport, North Carolina. I called Veneer Technologies and spoke to Elaine Avery, an in-house sales representative. She told me that a consumer without access to a local distributor may purchase their products directly from them through anyone in their in-house sales staff.

Address:
Veneer Technologies, Incorporated
PO Box 1145
Newport, NC 28570
US toll free number: 1-800-593-5601
FAX 1-252-223-3511
e-mail: vti@veneertech.com,
sales@veneertech.com
website: www.veneertech.com
Their product line is listed in detail on their website.

DAMPING FACTORS

I was interested to see the variable damping factor amplifier in the September 2001 issue on page 93 (Glass Shard). I always wondered how this was accomplished. I'm into low damping factors and I wondered how low it can be adjusted. A certain kind of speaker distortion can be reduced with a low damping factor.

If you connect a very low damping factor amplifier to a single driver—or a few drivers connected in series—without a crossover, the amplifier will control the current through the voice coils.

The current will be forced to follow the input voltage of the amplifier accurately in spite of changing voice-coil temperature, which affects the resistance of the coil. It is the current, not the voltage, through the coil that produces its output. Controlling the current may actually reduce the variations in the speaker's sensitivity.

The speaker will need more acoustic damping since the voice coil won't do this anymore. You may use an EQ or adjustable low-level, high-pass crossover instead to bring down the main resonance. A parametric EQ would be ideal for this purpose.

A difference between low and high damping factor amplifiers is that with a high damping factor amplifier the output voltage is controlled by the input and the current depends on the speaker—and the current peaks can become pretty high with a complex crossover and vented box, because of impedance variations with frequency.

In a low damping factor amplifier, driving a single driver without a crossover and a well-damped resonance, impedance variations are not so severe. So required amplifier overkill is not as big. The current is controlled by the input, and the voltage peaks depend on the speaker.

At a high voice-coil temperature, the resistance of the voice coil increases, and this has two effects. The peak voltage of the amplifier increases, and the power to the voice coil increases, instead of decreasing, so the speaker power needs to be derated.

William C. Cross
Denver, Colo.

THE PARTS CONNECTION

By now readers are probably aware of the demise of The Parts Connection. This is sad news for all readers of this magazine. Over the years, TPC has been an important supplier of high quality parts for audio equipment designers and builders, and they also produced many superb products in their Assemblage line of audio kits. According to a note posted on what is left of their website (www.partsconnection.on.ca), the parent company Sonic Frontiers International has pulled the plug on TPC in order to concentrate on their Anthem line of home theater equipment.

Since TPC's web page makes no mention of continued service and support for Assemblage products, many readers are probably wondering whether they have been left out in the cold. Paul Fliss of Sonic Frontiers International has assured me that this is not the case, and that SFI will provide service, including honoring warranties, for Assemblage products. If anyone is in need of Assemblage support, contact SFI via e-mail or telephone-contact information is posted on their website (www.sonicfrontiers.com).

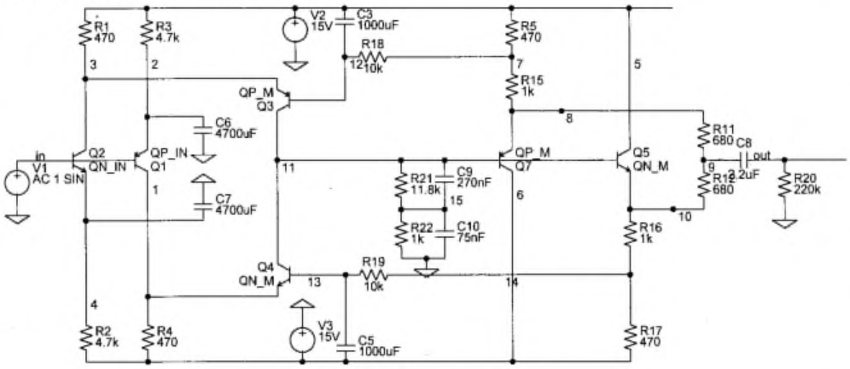
Gary Galo
Potsdam, N.Y.

A successor company which will carry on much of the parts supply functions of The Parts Connection is the new company, Parts Connexion at www.partsconnexion.com or 1-866-681-9602. E-mail at info@partsconnexion.com. - Ed.

PHONO CIRCUIT

To answer the cry from Art Day in *audioXpress*, September 2001 (p. 78), I offer this phono-circuit (*Fig. 1*). I designed it to accompany an article by M. J. Hawksford (*Hi-Fi News & Record Review*, Sept. 1984) on how the bipolar transistor is supposed to work.

FIGURE 1: MM/MC circuit.



It uses no exotic or magical components, but can be constructed from off-the-shelf parts. Transistors 3–6 should have a beta of 300 or more. Transistors 1 and 2 will benefit more from a lower beta, such as 100. Transistors 1 and 2 can even be 8W drivers for better results and are best matched in betas.

Eddy E.H. Folkers
The Netherlands

SUPPLIES FOR A SIMPLE AMP

It is laudable that Larry Lisle would offer a simple circuit for a beginner to attempt ("A Beginner's Push-Pull or Single-Ended Amp," Jan. 2002 *aX*,

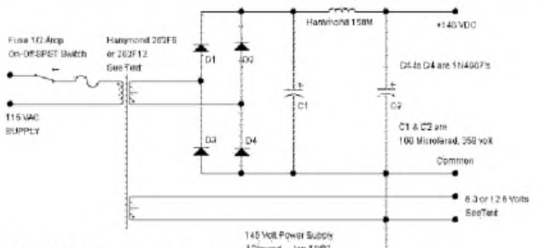


FIGURE 2: 140V power supply.

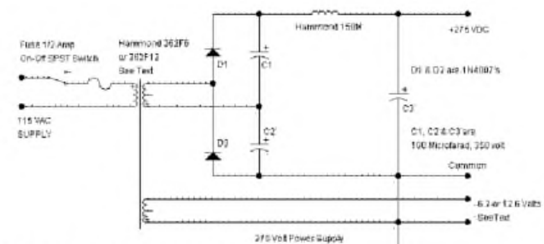


FIGURE 3: 275V power supply.

p. 24). However, there are two items that need to be addressed.

First of all, Larry did not include a power supply for his circuit. Here are two inexpensive alternatives that would work well (*Figs. 2* and *3*). One of them uses a bridge rectifier hookup and provides an output of about 140V. That would be good for about 400mW audio output from his circuit in the push-pull (PP) mode.

The other uses the same parts in a full-wave doubler. Output from this arrangement would be about 275V. Audio power output of the amp in the PP mode would be about 2W.

All of the parts used in the two versions of the power supply are interchangeable. Most are available through Antique Electronic Supply.

Second, the Hammond 808, while being suitable for this amp, is overkill. I'm sure Larry used it here because he had one available close at hand. I would have done the same. A good alternative at one-third the cost would be the Hammond 124E. That leaves enough money to buy the parts for the power supply. I believe we should offer the experimenter low-cost alternatives.

Using the Hammond 124E as the input transformer, you will need about 2V to drive the amp to full power.

A further savings is possible by using a 12SN7 rather than the now-expensive 6SN7. The power-supply parts list includes part numbers for both 6V and 12V versions of the power transformer. They are both the same price.

John L. Stewart
Ontario, Canada

Larry Lisle responds:

Thank you for writing! I'm glad you're experimenting and building. I've used transformers such as the Hammond 124E with good results. I've even written a couple of articles about projects using them. While they will work well in some applications, they don't have the voltage step-up and frequency response of the Hammond 808. As you pointed out, several times more input voltage is needed with the 124E. If you have the voltage available and like the sound, go for it!

Incidentally, I compared the 808 to some

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(from page 69)

of the classical transformers from the golden age by UTC and others and found it to be just as good. Electra-Print also makes an outstanding input transformer for single-ended circuits. Excellent transformers usually cost more, but you get what you pay for.

The 12SN7 is a fine tube, but you may need to use DC on the filaments to avoid hum problems.

I chose a voltage of 120–150V for the amplifier because this is essentially a beginner's circuit and the amplifier sounds very nice at this level. Feel free to experiment with the power supply or tubes or transformers as long as you keep safety in mind. Remember that changing one parameter may necessitate other changes, and the final sound will almost certainly be different. It may be better or it may be worse, but keep experimenting until you find the sound you like! That's what DIY audio is all about!

DRIVER REPLACEMENT

In the November 2001 issue of *audioXpress*, J. L. Markwalter wrote a speaker construction article that I really enjoyed and would like to construct ("A First-Order 3-Way," p. 24). I am having trouble finding the Focal 5K013-L drivers. Do you know of a source for the Focal 5K013-L driver or a substitution that would provide equal performance. If a substitution is used, what crossover modifications are necessary?

Dave Paulson
Carroll, Iowa

J. L. Markwalter responds:

I appreciate Dave Paulson's interest in my speaker construction article. I have contacted Zalytron from whom I purchased my Focal 5K013-L drivers and Speaker City USA concerning their current availability. Both dealers informed me that this particular driver has been discontinued and recommended the Focal 5K4211 as its replacement.

I have reviewed the parameters of the Focal 5K4211, which appears to be a good candidate to replace the 5K013-L, although I have not used it. Electrically the two are quite similar, but there are differences in their mechanical parameters, such as the 5K4211 having a slightly higher f_s and differences in Qs, among other things. Their frequency responses, polar coverage, and sensitivities are similar.

I cannot say that you can make a substitution without some modification of the mid-frequency crossover network values. If you decide to proceed with your project and use the Focal 5K4211 drivers, I recommend you order them and make impedance measurements on them in the enclosure before ordering the mid-frequency crossover parts, as you may need to change some of the component values. I would be interested in how this turns out and may be of some help if needed. ❖