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Reviews:

The PSB Stratus Silver^{i}

Onkyo's Home Theater Receiver

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CONTENTS

VOLUME 32 NUMBER 11 NOVEMBER 2001

FEATURES

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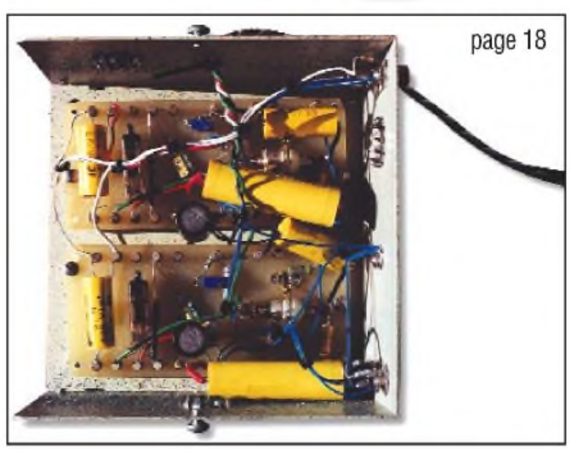
REVIEWS

page 40



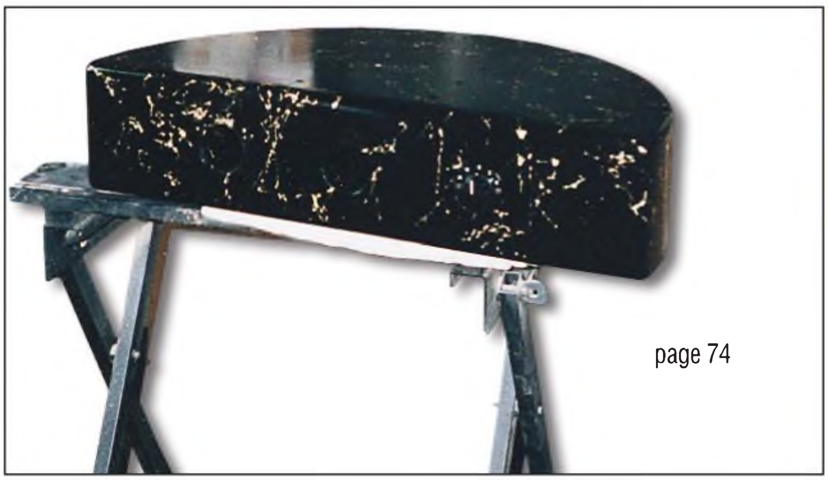
page 18

page 69

DEPARTMENTS

page 32

IN EVERY ISSUE



page 74

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

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

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Audio News

NEW PRODUCTS FROM BOSTON ACOUSTICS®

Boston Acoustics introduces the Boston Bravo® high-performance loudspeaker and the Dsi250 in-wall, flush-mount speaker. The Boston Bravo is basically one quarter of a cylinder, allowing mounting in any corner, where a wall and ceiling meet, or on a flat surface such as a table or shelf and offering consumers the Boston Sound™ in applications such as theater, stereo, commercial, or whole-house. This loudspeaker features Boston's 1″ VR tweeter with AMD™, both ferrofluid and extruded-aluminum heatsinks on the tweeter, and a 4½″ woofer extending low-frequency response to 80Hz (–3dB). The Dsi250 is a flush-mount in-wall speaker featuring a two-way design involving a 5¼″ woofer (with a lightweight, low-mass cone) and ¾″ ferrofluid-cooled tweeter. This speaker is finished in soft white and can be painted to match any wall or ceiling. Contact Boston Acoustics, Inc., 300 Jubilee Drive, Peabody, MA 01960, 978-538-5000, www.bostonacoustics.com.

AUDIO GOLD RESISTOR LINE

Ohmite Manufacturing Company announces the new Audio Gold resistor family, designed for high-end loudspeaker and amplifier applications. These resistors feature welded construction, high-quality resistance wire wound on a ceramic core, and flameproof Centohm® coating. Ohmite offers leaded configurations from 3–10W, tubular 12–50W sizes with tab terminals, and resistance values from 0.47Ω to 100kΩ in ±1%–±5% (std) tolerance. Contact Ohmite Manufacturing Company, 3601 Howard Street, Skokie, IL 60076-9724, 847-675-2600, FAX 847-675-1505, kschwiebert@ohmite.com.

TRIBUTARIES® SUBWOOFER CABLE

Tributaries introduces their new subwoofer cable with integrated mono/Y output, allowing subwoofers with two audio inputs to connect to A/V receivers with a single mono output. The cable features twin Oxygen Free High Conductivity (OFHC) multi-stranded copper conductors, OFHC braided shield surrounding center conductors, a protective PVC covering, and more. These cables are available from 1m–8m. Contact Tributaries, 1307 East Landstreet Road, Orlando, FL 32824-7926, 800-521-1596, www.tributariescable.com.

NAD COMPACT MUSIC SYSTEM

NAD Electronics is offering a new compact music system featuring the L40 CD Receiver (with optional PSB Alpha Mini Speakers), the first in a proposed series of products called LifeDesign; these products are designed to reduce clutter and extra connections and concentrate on performance, simplicity, and value. The L40 CD Receiver is an integrated CD player, FM tuner, and amplifier in a single chassis no larger than an average CD player. The CD section employs digital-to-analog conversion (supported by a four-pole analog output filter) and separate power regulators for analog and digital sections. Contact J.B. Stanton Communications, 860-542-1234, FAX 860-542-0005.

BRYSTON POWERPAC 250

Bryston Ltd. added the PowerPac 250 to their PowerPac line of modular, single-channel amplifiers designed for on-speaker or near-speaker mounting. The PowerPac line now consists of three amplifiers that deliver 60W (PowerPac 60), 120W (PowerPac 120), and 250W. Each ships with 5-way banana jack outputs, RCA (unbalanced) and XLR (balanced) inputs, and a ground lift switch to mitigate any system hum; the PowerPac 120 and 250 have 12V triggers for remote turn-on. Contact Bryston LTD., 677 Neal Drive, Peterborough, Ontario K9J 7Y4, CANADA.

CD TRUSTEE

CD Trustee is a new Windows program that catalogs your CD collection by automatically accessing an internet database to gather the artist name, album title, song titles, music genre, and more, when you insert CDs into your computer; this takes only seconds per CD. You can manually add other information, such as location, your ratings, or whether you loaned it to someone. You can then print reports, quickly find an artist, album, or song, or print cover images or jewel case inserts containing song lists. Contact Mike Paulick, 303-838-8549, Mike@Base40.com.

RUSSOUND CUSTOM LOUDSPEAKERS

Russound introduces four new models, the SP652.1, SP632.1, SP622.1, and SP522.1, for their line of custom-installation loudspeakers. The SP652.1 and SP632.1 are uni-axis point source designs including 6.5″ injection-molded graphite laminated cones with rubber surrounds and uni-axis 0.5″ Mylar dome tweeters with high-performance neodymium magnets. The SP622.1 (6.5″ woofer) and SP522.1 (5.25″) include injection-molded woofer cones and twin Mylar dome tweeters. Because the drive materials are all moisture-resistant, all four models can be used in a wide variety of indoor applications, including showers and saunas. The SP652.1 and SP622.1 are square designs to flush-mount in a wall or ceiling, while the SP632.1 and SP522.1 have a round bezel and grille and are intended for use in ceilings. Contact Russound, 603-659-5170, FAX 603-659-5388, www.russound.com.

Editorial

aX: The First Year

Our new, combined periodical, *audioXpress*, will be just one year old with publication of the December issue. It seems appropriate to take some stock of what has happened during the months since we launched this publication dedicated to hands-on audio technology.

In one sense, *audioXpress* is a return to our roots in *Audio Amateur*, whose first decade covered audio hardware of all kinds. Only with the launch of *Speaker Builder* did the magazines begin to become more specialized. *Glass Audio* in 1988 was a response to the growing interest evident in many manuscripts about the renaissance of tube audio.

Our goal for *aX*, resulting from a year of planning, was a monthly which was balanced in presenting all three technologies: solid state, loudspeakers, and vacuum tubes. Article ratios were 2,3,2 for each issue, so that over the course of the year we intended to publish 24, 36, 24 articles, respectively, for the three interest categories.

We redesigned the magazine, upgraded the paper, used more color, and set a 100-page per issue target, which we maintained for the first ten issues. We expected circulation to go to somewhere between sixteen and seventeen thousand—although we had no guarantees from our news dealers, for whom this was a "new," untested magazine. I am glad to say we met that target.

In advertising, we had great hopes that advertising clients from the three previous titles would stay with us. If they had done so, we ought to have had some 50 pages of advertising per issue. On average we have been running 27 to 30 per issue. Now the facts of life about ad ratios in monthly periodicals *requires* a level between forty and fifty percent. We raised ad rates, fairly modestly, because we were delivering almost twice the circulation level of any of our previous magazines.

Gross advertising revenue for *audioXpress* for 2001 looks as though it will be higher than it was for the three previous titles. This despite some problems we have experienced with sales agencies. I am very grateful to those advertisers who recognized that we were offering an enhanced market for their products and have taken advantage of this. I can understand the hard business decisions others have had to make by keeping budgeted amounts level, and going for smaller space contracts.

The new magazine's production is somewhat more expensive than the three previously published titles. Mailing and shipping costs are up significantly.

It is of primary importance for every reader to realize that his or her response to the advertisers is crucial to the health of this avocation. Without the advertisers and the products they offer, the audio components we build or buy for music reproduction will not be available. And the magazine will not exist.

> *As I write this we are only a few days beyond our nation's tragedy of September 11. Clearly we must all go on with our lives and work, while dealing with our grief. In these terrible days, music has helped with our grief as never before. And "God bless us every one."***—E.T.D.**

Audio reproduction is undergoing massive changes these days. Audio is now being regarded as just one ingredient in communication media: video and home theater, computers and internet music as well as music storage and editing. Some are prophesying that "stereo is dead."

Does a magazine committed to high-quality sound allied to craftsmanship still have a place in the world? We at Audio Amateur, Inc., believe that it does. It may become an "old" technology, but the interest in vacuum tube technology and the continuing popularity of long play disks confirms that older ways of doing things have a continuing life in surprising ways.

We believe that you have, in being part of *aX*'s life for the first year, confirmed that craft audio is a vibrant and encouraging success. Although for some, the change was a disappointment, and they have dropped subscriptions. Those few have been replaced, and others attracted, so that in each and every month we have registered net growth in circulation.

Despite the general downturn in business, we are seeing a small growth in advertising month by month. Given such economic facts, this is undoubtedly a success story. I am grateful to you as subscribers for your response. I thank our advertisers for their participation and believe they are benefiting from the change.

I thank our wonderful staff who have given generously of their intelligence and remarkable skills to make *aX* a reality month by month. It would be impossible to overstate the contribution authors have made to the publication. I am just as amazed, pleased, and delighted day after day to receive the amazing offerings of the authors whose work fills our pages.

The year 2002 will bring some new adventures for the company. We are instituting a subscription rate increase from $29.95 to $34.95. This makes the subscription price less than 50% of the cover price, just under $3.00 per month. The change will go into effect January 1, 2002, and old rates will be in effect through December 31, 2001. In addition to our two annuals, *The World Tube Directory* and *The Loudspeaker Industry Sourcebook*, we plan a special for tube lovers: *A Glass Audio Special*. This will feature 12 articles on tube designs and will be available in mid-April.

A year from now, we will publish a new, comprehensive equipment directory covering loudspeakers, wire, crossovers, and equalizers. In 2003, we will publish Part 2, covering all other audio hardware equipment offerings.

We are looking forward to a new year and fresh opportunities for audiophiles and for continued improvement in the quality of reproduced sound—of every kind. I hope you are as encouraged as we are about what is becoming possible in this changing environment. One thing is certain: our love of music is immortal and our commitment to keeping it as beautiful and untainted as possible is strong and resolute.**—E.T.D.** ❖

Choosing and Using Electronic Parts: A Survival Guide, Part 1

Everything you need to know about how parts function, and how to select and maintain them. **By Charles Hansen**

You have just purchased a new solid-state power amplifier for your audio system. The owner's manual admonishes: "Provide adequate ventilation to allow for dispersion of the heat which is generated by this amplifier. Leave at least two inches above and behind the amplifier. Do not place books or other objects on the unit so that the free flow of cooling air through the slots and openings in the cabinet are obstructed. Do not place near radiators, hot-air vents, or other sources of heat.

"Do not place on a bed, sofa, rug, or other similar surface that may block the cooling slots and openings. Do not install the amplifier in a confined space, such as a bookcase or built-in cabinet, unless proper ventilation is provided. Fan cooling may be necessary in some cases."

TIMES CHANGE (OR DO THEY?)

Why is this? Isn't modern solid-state equipment supposed to run cool and be highly reliable? Why is the life of electronic equipment so drastically shortened when it becomes hot?

The damaging effects of high temperatures generated by vacuum tube equipment seems a little more obvious than they are with solid-state devices. In the "Olde Days," you had wax-impregnated paper capacitors. Wire insulation was made of rubber, linen, cotton, and other low-temperature materials. The life of tube filaments decreased as temperatures soared. The need for adequate circulation of cooling air was obvious. You could feel the heat.

So why hasn't the use of modern high-temperature materials and the decreased amount of heat generated by solid-state equipment made these warnings obsolete? Some people suspect that electronic equipment manufacturers design their products for a limited lifetime—one day past the expiration of the factory warranty, for instance. While the "planned obsolescence" conspiracy theory has been around for a long time, major name brands have reputations to protect. A bad showing in the repair history bar graphs that accompany the product tests in *Consumer Reports* magazine and its adverse effect on future sales are reason enough for manufacturers to strive for a long operational life.

WHY THINGS FAIL

You are much more likely to experience obsolescence from format changes than from the failure of electronic parts. The hardware for erstwhile formats such as Betamax VCRs, reel-to-reel tape recorders, and digital compact cassettes may still perform well, but readily available support for these machines is sometimes difficult to find.

Failures in equipment with moving parts (CD players, VCRs, hard drives) are much more common than in those that are purely electronic (amplifiers, computer sound cards, modern tuners). Mechanical wear is inevitable and is accelerated by dust, vibration, humidity, and other factors against which few consumers bother to take preventative measures.

The primary enemies of electronic circuitry are heat, the less-than-perfect structure of all materials used in electronics, and the electrochemical reactions that occur to degrade materials as their temperatures increase. Higher temperatures accelerate the failure mechanisms in all electronic components in an exponential fashion. All components have a basic reliability, which has its baseline at 25°C.

Above this temperature, the components follow the "10°C" rule, which states that the life of the component decreases by half for every 10°C rise in temperature. If a transistor has a reliability of 1 million hours at 25°C, its expected life will be only 250,000 hours at 45°C. This will decrease to 125,000 hours at 55°C. Few components exist that can withstand more than 125°C.

The temperature in electronic components will rise if you increase the power through the device, or increase

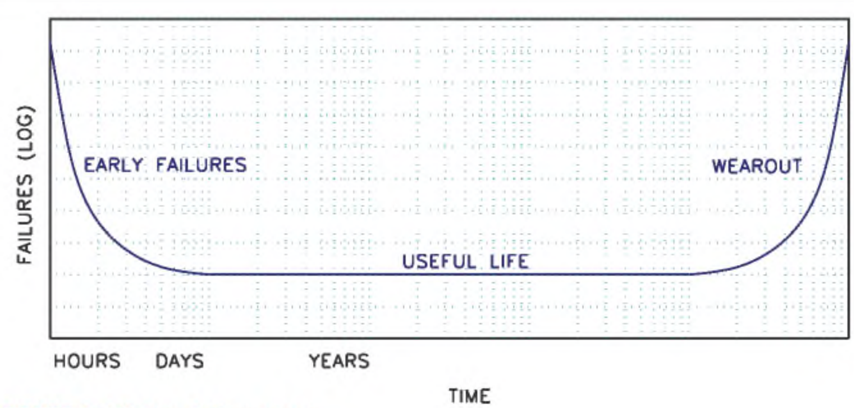

FIGURE 1: Lifetime failure curve.

the ambient temperature at which you operate it, or both. The ambient temperature for any component will be increased by other heat-producing components that are in close proximity, so the probability of a uniform temperature within a piece of electronics equipment is low.

The failures for any electronic device over time follow a typical "bathtub" curve (*Fig. 1*). There is an initial period of early failures during which defective parts expire. Some manufacturers try to weed out these early failures by means of a limited operational burn-in, and/or incoming device testing, so failures are less likely to occur after the customer purchases the equipment. Depending on the complexity of the equipment, it may take from 2 to 68 hours, sometimes at elevated temperature, to statistically eliminate these early failures.

The flat low-failure portion of the curve represents the useful life of the product. After a certain number of years, the curve begins to slope upward again as wear-out failures begin to occur. All the components have been operating for the same amount of time, so any repairs may be short-lived since additional components may begin to fail as additional "rising-curve" hours are accrued.

MEAN TIME BETWEEN FAILURE (MTBF)

MTBF is, by definition, the total running hours divided by the number of failures, and is typically expressed in hours. MTBF is a statistical value determined by theoretical calculation or field statistics (see sidebar). Its purpose is to estimate a random value, in hours, of the probability that a failure will occur within a population of equipment, for the environmental and operating conditions under which the data was obtained. Failure rates apply to equipment with power applied while it is performing its normal intended function.

Say a resistor has a failure rate (λ) of 1×10^9 hours. If 50% of rated power is applied, and the case temperature is 40°C, statistically it will not fail for a billion hours. If a piece of equipment uses 1000 of these resistors, you can expect a statistical resistor failure every million hours. Taken to absurdity, a piece of equipment with a billion resistors will statistically see one resistor failure every hour (or 10,000 resistors fail after 100,000 hours).

You can see from this that the published statistical failure rates do not predict reliability or guarantee the operating hours you can expect. It provides the standard by which electronic equipment manufacturers can estimate failure rates using a common set of rules that provide a basis for MTBF comparison. There is a statistical variation for the wear-out life of a piece of equipment as well. In fact, the standard deviation for wear-out life is much greater than the standard deviation of MTBF. As they say in the auto business, "your actual mileage may vary."

COMPONENT TOLERANCES

All electronic components have a specified allowable deviation from the "nominal" value at 25°C. This deviation is called its "tolerance." Electrical parameter changes due to environmental variations and aging must be taken into account when designing an electronic device.

A properly designed product will allow for all the worst-case tolerance variations that may occur over the life of the product. Once a component exceeds its maximum tolerance range, it may or may not cause failure or degradation in the operation, depending on the criticality of the component in the overall function.

MECHANICAL ENVIRONMENT: SHOCK AND VIBRATION

Most electronic products are subject to vibration and shock at least during shipping, and components that are not securely mounted can short-circuit against adjacent components, or leads and connections may be weakened or broken. Additional steps are needed if components that are designed to be mounted flat on the circuit board are stood on end. Components should also be mounted to preclude dirt and moisture build-up between conductors that might result in short circuits.

Failures can also be the result of electrical, mechanical, or environmental overstress or manufacturing defects. One of the most frustrating failures is erratic operation, which can result from intermittent opens or shorts, insulation breakdown, internal connection defects, component value drift, or instability with temperature. Many TV sets have spent their later years in the repair facility, waiting for the customer's complaint to repeat itself.

CAPACITORS

Despite their simple appearance, capacitors are complex devices[1,2]. In addition to the obvious parameter of capacitance, they also have series resistance (ESR), inductance (ESL), and parallel leakage resistance (Rp). Capacitors generally come in two types (fixed and variable) and two generic categories (electrostatic and electrolytic). Electrostatic capacitors use inert dielectrics. Electrolytic capacitors have a very thin dielectric oxide film that is chemically formed on the surface of the electrodes.

The capacitance-voltage (CV) product for a given size capacitor is determined by the dielectric constant, or relative permittivity (ε_r), of the insulating material. This is the ratio of the permittivity of the material to the permittivity of a vacuum (free space). The dielectric constant is directly proportional to the capacitance and the leakage current.

A higher dielectric constant increases the electric flux density, enabling a capacitor to hold its charge for longer periods of time, and/or hold large quantities of charge. Electrolytic capacitors have very high dielectric constants, while film capacitors have very low dielectric constants.

Failures in capacitors are related to operating temperature, peak voltage stress on the dielectric, peak current in solid tantalum and metallized film capacitors, and the maximum hot-spot temperature. The most prevalent failure mechanism is dielectric breakdown. Most dielectric materials deteriorate with time and temperature and undergo a slow aging process by which they become brittle and more susceptible to microscopic cracks. Dissipation factor (DF) increases with temperature as well.

Insulation resistance (measured in megohm-microfarads, or MΩ-μF) decreases exponentially as the temperature increases above 25°C because of increased electron activity. Low insula-

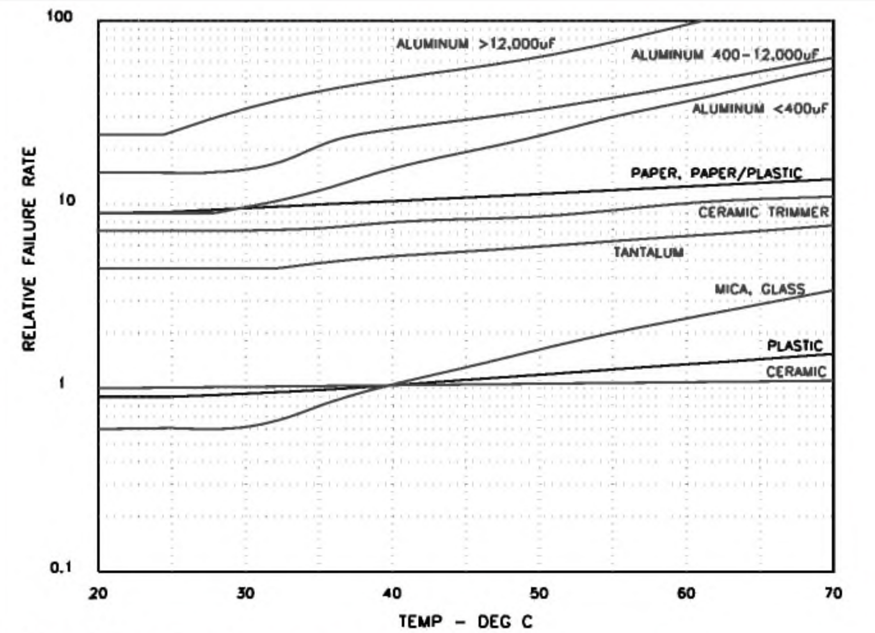

FIGURE 2: Capacitor failure rate versus temperature.

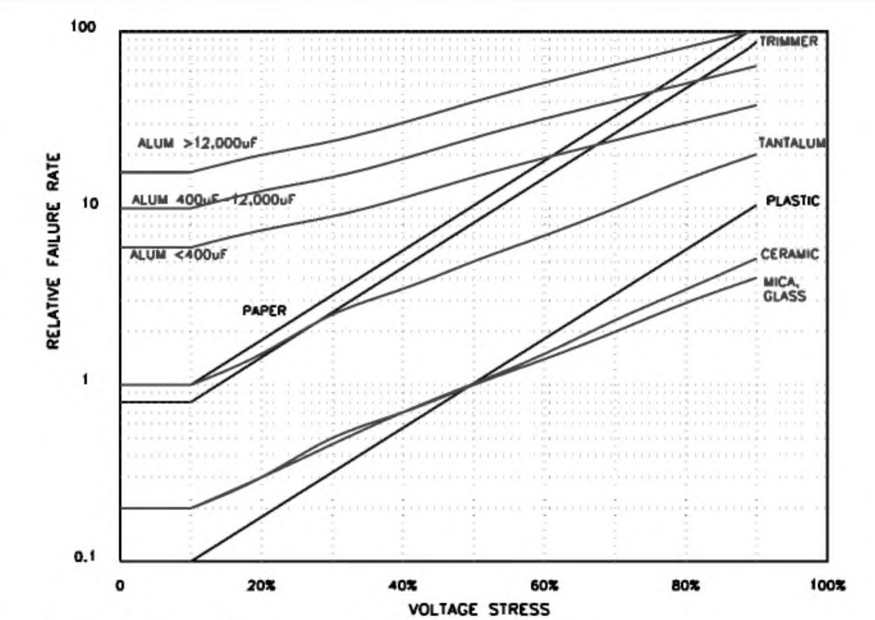

FIGURE 3: Capacitor failure rate versus applied voltage.

tion resistance can also result from trapped moisture as a result of prolonged exposure to excessive humidity, or from moisture trapped during the manufacturing process.

The dielectric withstanding voltage decreases as the temperature increases due to the chemical activity of the dielectric material, which causes a change in the physical or electrical properties of the capacitor. Dielectric breakdown may also occur as a result of high voltage transients (surges), causing premature failure. Operating DC-rated capacitors at high AC current levels can cause localized heating at the end terminations and separation of the end-connection of the capacitor. This is why care must be taken when selecting a capacitor for AC applications.

Mounting a capacitor by its leads in an environment where it is exposed to high shock or vibration levels may also cause it to open. Military specifications require that components weighing more than one-half ounce not be mounted by their leads alone. The lead wire may fatigue and break at the body connection, so the capacitor body must be fastened into place with a clamp or conformal adhesive.

Figure 2 shows the relative failure rates for capacitors versus temperature. Since the baseline specifications for all electronic components are defined at 25°C, the reliability standards do not provide any reduction in failure rate below this temperature. In fact, components such as aluminum electrolytic capacitors may be adversely affected by very low temperatures. Many consumer-grade components are rated for only 0°C to 70°C operation, while Mil-spec parts can be used from –55°C to +120°C.

Figure 3 shows the relative failure rates for capacitors versus applied voltage (DC plus peak AC). Note that there is no credit for operating a capacitor (or any other component) at less than 10% of its rating. Even a component with no voltage across it still has a statistical failure rate. Above 90% rated voltage, the chance of failure increases by many orders of magnitude, so none of the reliability standards project failure rate data into this area of operation.

ELECTROSTATIC CAPACITORS

A paper capacitor consists of aluminum foil and a kraft paper dielectric rolled into a cylinder and impregnated with oil, wax, or epoxy resin to exclude moisture. Paper-plastic capacitors add polymers to the kraft paper. Oil-filled paper capacitors generally have higher voltage ratings than other capacitors, from 400 to 5000V.

When a dielectric fault breakdown does occur, the paper is self-healing and leaves little carbon residue that could decrease the insulation resistance. Metal cases are used to contain the oil, and the high voltage types have terminals that are brought out through ceramic stand-offs. They are often used in AC applications such as motor starting and power factor correction.

Since the 1970s, oil-paper capacitors have used mineral oil or synthetic polymer oils. Some of the early paper-oil capacitors are filled with toxic PCBs—be wary of what you buy surplus. Seal leakage is common in poorly made oil-impregnated capacitors. Mechanical failures are caused by fracture of the electrode tab at the point of attachment

WHEN WILL FAILURE STRIKE?

Two standards are used for calculating MTBF. The Defense Department authorized the first issue of Military Handbook 217 (MIL-HDBK-217A), *Reliability Prediction of Electronic Equipment*, in 1965. The latest issue is MIL-HDBK-217F Notice 2, Feb. 1995 (www.dodssp.daps.mil).

The former Bell Communications Research (Bellcore) took MIL-HDBK-217 as a starting point and then modified and simplified the failure rate models to better reflect their own field experience. In 1985 they issued Technical Reference 332 (TR-332), *Reliability Prediction Procedure for Electronic Equipment*, now also known as Telcordia GR-332 (www.telcordia.com). The latest revision is Issue 6, Dec. 1997. Bellcore developed TR-332 for telecommunications companies, but it has now been adopted for many other commercial electronic products.

MTBF is the inverse of the sum of the failure rates of all the electronic and electromechanical parts (λ) in a piece of equipment.

$$MTBF = 1/\Sigma(\lambda_1 + \lambda_2 .... + \lambda_n)$$

The failure rate models for each individual part are estimates based on available data, and are the product of the base generic failure rate λ and a number of modifying π factors. The failure rates in TR-332 have fewer modifying π factors than MIL-HDBK-217. For TR-332, the steady-state part failure rate is:

$$\lambda_{SS} = \lambda_G \pi_Q \pi_S \pi_T \pi_E$$

where: λ_{SS} is the steady-state part failure rate (failures in 10^9 hours)
λ_G is the generic failure rate
π_Q is the quality factor
π_S is the electrical stress factor
π_T is the temperature stress factor
π_E is the environmental conditions factor
π_{FY} is the first year multiplier

For MIL-HDBK-217, the part failure rates λ_p (equivalent to TR-332 λ_{SS}) are the products of the base generic failure rate λ_b (equivalent to TR-332 λ_G) and a larger number of modifying π factors, some with added coefficients. As a minimum:

$$\lambda_p = \lambda_b \pi_T \pi_S \pi_Q \pi_E$$

where: λ_p is the part failure rate (failures in 106 hours)
λ_b is the base part failure rate
π_T is the temperature stress factor
π_S is the electrical stress factor
π_Q is the quality factor
π_E is the environmental factor

Other π factors used in the handbook are:

π_A is the application factor (certain semiconductors)
π_C is the construction factor (magnetrons and semiconductors)
π_{CF} is the configuration factor (vacuum capacitors)
π_{CYC} is the cycling factor (mechanical switches and relays)
π_{CV} is the capacitance factor (capacitors)
π_F is the function factor (meters)
π_K is the mating/unmating factor (connectors)
π_{LS} is the load stress factor (switch and relay contacts)
π_L is the learning factor (number of years since a vacuum tube's introduction to field use)
π_M is the matching network factor (RF components)
π_P is the power degradation factor (lasers, laser diodes), or active pins (connectors)
π_R is the power rating factor (semiconductors) or resistance range factor (resistors)
π_S is the size factor (synchros and resolvers)
π_{SR} is the equivalent series resistance factor (electrolytic capacitors)
π_U is the utilization factor (magnetron tubes, switches)
π_V is the voltage factor (variable resistors and capacitors)

The failure rates for LSI chips (Large Scale Integrated Circuits) such as microprocessors and gate arrays are calculated by gate count or transistor equivalents, and tables are provided in both reliability specs for determining the generic failure rates. There are also tables for estimating the junction temperatures of these complex parts.

As you might expect, electromechanical parts have higher failure rates than electronic parts. The λ_{SS} = 10 for switches is

to the electrode or to the external lead.

Paper capacitors have always been given higher failure rates than the other electrostatic types. This is due to the relatively uncontrolled variations in the kraft paper insulation. There are a number of suppliers who hand-build special paper and paper-oil capacitors for high-end audio. To my knowledge, their long-term reliability has not been established in any recognized standard at this point, but this does not automatically declare them unreliable.

PLASTIC FILM CAPACITORS

The plastic film capacitor is similar in construction to the paper capacitor. Plastic is much denser than paper, and contamination particles are virtually nonexistent. Plastics can withstand higher temperatures and are considerably more stable than paper.

A number of suitable plastics are available for use as the dielectric in film capacitors. The conductive plates of the capacitor can be either a separate thin metal foil (film-foil), or aluminum can be sprayed directly on the plastic film (metallized). Film-foil capacitors have lower ESR and higher insulation resistance than metallized types, but metallized film capacitors are smaller for the same CV product, and are self-healing if the plastic insulation breaks down.

If a voltage breakdown occurs, and if sufficient current is available, the metallization surrounding the insulation breakdown will burn away, restoring normal operation. In the film-foil types the foil will easily carry the fault current, which will carbonize the plastic and cause a permanent short-circuit. The residual carbon may reduce the insulation resistance somewhat.

The plastics used in film capacitors that are of interest to the audio enthusiast are polyethylene terephthalate (PE, polyester or Mylar™), polycarbonate (PC), polypropylene (PP), and polystyrene (PS). Other less-familiar film types are polyphenylene sulfide (PPS), Parylene (paraxylene), polysulfone (PSO), polytetrafluoroethylene (PTFE or Teflon™), and polyvinylidene difluoride (PVDF).

For a long time, plastic capacitors were given the same high failure rates as paper dielectric capacitors. In the last 20 years this has changed with the improved processing of polymers and the introduction of advanced engineered materials. Specialty manufacturers have introduced new film and film-foil designs aimed at the high-end audio market.

Polyester film has the highest dissipation factor (DF) of the films commonly used for capacitors. Also, its DF increases exponentially with temperature and directly with frequency, which can lead to thermal runaway in AC voltage applications. Polyester film absorbs moisture (it is hygroscopic), which can decrease insulation resistance and result in unstable capacitance.

Polycarbonate, polypropylene, polystyrene, and Teflon™ capacitors are very stable and much less influenced by moisture and temperature. PS and PP types are widely used in audio applications.

CERAMIC CAPACITORS

Disc ceramic capacitors consist of special ceramic dielectric materials applied over two or more silver electrode layers, fired at high temperature and encapsulated for protection. Another type of ceramic capacitor is the monolithic type, which consists of silver plates molded with ceramic into a single block. There are three principal ceramic types, with widely different temperature characteristics.

NP0 or COG types are made from non-ferroelectric ceramics. They are the largest ceramics for a given capacitance, but they have very low temperature coefficients (excellent stability). X7R or BX types are made from a ferroelectric ceramic called barium titanate, which changes its crystalline form at elevated temperatures. This structural change causes a change in the dielectric constant that alters the capacitance.

Z5U types have even lower stability, but better packaging efficiency than X7R. Temperature coefficients for NP0 types are very low: ±30ppm/°C. The X7R types vary 15% with AC voltage and up to −25% with DC voltage. The Z5U types vary as much as +22%, −56% over their full temperature range.

X7R and Z5U ceramic capacitors have additional undesirable characteristics. Their capacitance decreases exponentially with time. As the applied voltage increases, the capacitance decreases (this is the ferroelectric effect), from 10% for X7R to 50% for Z5U. This characteristic makes them a poor choice for audio, where a varying signal voltage across the capacitor can cause nonlinear distortion.

In DC decoupling applications, the partial polarization of the ferroelectric ceramic persists even after the voltage is removed. General-purpose ceramic capacitors also exhibit piezoelectric effects, in which they can transform mechanical vibration into electrical noise, similar to an electret microphone element. I have also seen large ceramic capacitors develop internal fissures when subjected to high inrush currents.

Dielectric absorption occurs when a voltage reappears across a ferroelectric ceramic capacitor that has been fully discharged. Charge is absorbed into the dielectric due to its dipole moment and the migration of free electrons to its surface.

Ceramic capacitors are generally quite reliable. Most failures are caused by cracks in the materials used to protect the capacitor and lead assembly. Electrical degradation will occur if moisture and contaminants are absorbed between the coating and capacitor plates. Ceramic capacitors do not self-heal like metallized film capacitors.

MICA CAPACITORS

Mica is one of the best known natural insulators, and mica capacitors are available with voltage ratings as high as 30kV. Silver-mica capacitors have a thin layer of silver deposited on the surface of the mica, resulting in excellent stability and tight tolerance. They are among the most reliable of capacitor types. Like ceramics, most failures are caused by cracks in the capacitor case.

GLASS CAPACITORS

Glass was first used as a capacitor dielectric in the early 1950s, and is available with voltage ratings up to 6000V. Capacitors of this type are characterized by extremely low losses and excellent stability, although the glass cases are fairly fragile.

about the same as power semiconductors. Small fans have $\lambda_{SS} = 50$. Relays have $\lambda_{SS} = 70$ or more.

Things are also difficult in microwave and high-frequency switching applications. The λ_{SS} for HF switching devices, detectors, mixers, and amplifiers are all above 100.

The stress factors extend to 90% of the particular parameter's rating. Once the applied stress exceeds this limit (overstress), the part failure rate is no longer valid, and catastrophic part failure is likely to result.

Given the severe penalty for earth orbit, how do satellites last as long as they do? Currently, it is impossible to repair a satellite in geosynchronous orbit, so to ensure long life, the electronics are built with the highest reliability (Hi-Rel) Military specification (Mil-Spec) parts available:

1. Established Reliability (ER) passive parts,
2. Joint Army-Navy Tested-Extra 100% internal Visual inspection (JAN-TXV) discrete semiconductors,
3. MIL-M-38510; monolithic, multichip, and hybrid microcircuits and their quality and reliability assurance requirements.

The finished equipment is then given a burn-in with full environmental qualification. Redundant circuits are used for critical functions. While redundant circuits do add to the total parts count, properly designed fail-safe/fail-operational redundancy earns some significant credit factors. Part failure rates are additive, while redundancy reduces the failure rate exponentially, lowering the overall equipment failure rate well in excess of the added penalty due to the increased parts count.

There are two specific techniques used to evaluate MTBF. During the product development process, engineers calculate predicted MTBF using the failure rates of the components used in the design. Initially, the design is calculated using a parts count method that assigns an operating temperature of 40°C, and an electrical stress of 50%. Once component values and ratings are selected and actual junction and hot-spot temperatures are known, the predicted MTBF can be calculated with a higher accuracy.

The preferred calculation method for MTBF requires data that documents actual failure rates for electronic components under normal operating conditions. Manufacturers or users analyze their field return rates to determine the actual field-demonstrated MTBF. Using this data, they can calculate the overall field MTBF using a set of assumptions based on accurate sampled information.

It is also important for a manufacturer to have good design rules, careful component selection, quality control and test procedures, and a controlled production process that builds products with the highest possible life expectancy.

As mentioned in the article, one of the most important design tools is the Failure Mode and Effect Analysis (FMEA). The FMEA takes each mode of failure for a given component (short, open, degradation, value shift, etc.) and then determines the effect on the overall equipment. A failure in one device can propagate and lead to subsequent failures in other components. Whenever an undesirable failure effect is found, the engineers will evaluate the compensating conditions, and perhaps make a design change to make the failure effect more benign. An FMEA is complementary to the design process of defining positively what a design must do to satisfy the customer, but it is most useful when applied with failure mode distribution data.

The military failure mode distribution reference is published by the Reliability Analysis Center (RAC), Failure Modes/Mechanism Distributions, document FMD-97 (www.rac.iitri.org). It lists the percentages of failure mode probability, obtained from repair facility repairs for electronic and electrical components. Some examples (not, unfortunately, from the latest version of FMD) are:

Carbon composition resistor: 46% open, 3% short, 51% value drift.
Film resistors: 45% open, 15% short, 40% value drift.
Variable resistors: 70% open, 10% short, 30% mechanical binding.
Ceramic caps: 30% open, 40% short, 30% value drift.
Metallized caps: 43% open, 43% short, 14% value drift.
Film-foil caps: 20% open, 70% short, 10% value drift.
Tantalum caps: 10% open, 85% short, 5% value drift.
Aluminum caps: 60% open, 10% short, 30% value drift.
Signal diode: 40% open, 60% short.
Power diode: 30% open, 70% short.
Zener diode: 40% open, 10% short, 50% value drift.

Failure modes and their probabilities become more complicated with transistors, ICs, and larger scale devices. The concept of an individual transistor "failure" in a 100,000 transistor monolithic microprocessor is not easy to fathom in a practical sense.

VARIABLE CAPACITORS

Air variable capacitors generally have one set of plates meshing between a second set of plates. The capacitance is altered by turning a shaft to vary the overlap area of the movable and fixed plates. The greater the common area, the larger the capacitance. The fixed plate assembly is called the stator, and the movable portion is called the rotor.

Another type of variable capacitor is the trimmer, or padding capacitor. This type of capacitor consists of two or more plates separated by mica dielectric. Tightening an axial screw compresses the plates against the dielectric, increasing the capacitance. Ceramic trimmer capacitors are similar to the air types, except a ceramic-coated metal disc is used as the movable plate.

The plates of a variable capacitor are fragile, and short-circuits can occur due to mishandling.

ELECTROLYTIC CAPACITORS

Electrolytic capacitors have electrochemical as well as the usual mechanical failure mechanisms, making them inherently less reliable than electrostatic capacitors.

ALUMINUM

Aluminum capacitors are used primarily for power-supply filters. The AC ripple current passing through the capacitor produces a power loss (heating) in the ESR. Aluminum capacitors are also used as coupling capacitors in audio circuits. The ESL makes electrolytics ineffective as capacitors at high (radio) frequencies, where the inductive reactance is very large. There are low-leakage aluminum capacitors available that have been specially designed for audio applications.

The dielectric in an aluminum electrolytic is a thin layer of aluminum oxide that develops on the aluminum foil electrodes during the high-voltage “formation” process. The foil is then wound with a thin paper separator and inserted into the aluminum capacitor case. Finally, an electrolyte of ethylene glycol and ammonium borate is added, which allows the oxide to renew itself during operation.

This chemical nature of aluminum electrolytics causes them to have a limited shelf life—the amount of time the capacitor is stored without any voltage applied, whether in a piece of electronics equipment or in a distributor’s inventory. The electrolyte gradually evaporates through the case seal. The capacitance and internal resistance continues to drop until the DC leakage current eventually exceeds the limit value for the capacitor. This is the primary wear-out mechanism for aluminum capacitors. If hydrocarbon PC board cleaners are used, they can enter the capacitor and attack the aluminum foil through the end seal.

Modern electrolytic capacitors are more sensitive to ambient temperature than they are to voltage or ripple current (*Figs. 2* and *3*). It is important to avoid placing high power dissipation near electrolytic capacitors. The internal hot-spot temperature can be quite high in large diameter filter capacitors, so ripple current limits are set by limiting the internal rise to a fairly low 10°C.

It is not unusual for power amplifier

manufacturers to connect a number of smaller filter capacitors in parallel to obtain a higher total reservoir capacitance. The I^2ESR power loss heating effect is the limiting factor for each individual capacitor in the filter bank. The usual practice is to use capacitors with the same voltage and capacitance so the total allowable ripple current is the sum of the ripple current ratings of the individual capacitors.

Tube amplifier designers, on the other hand, often use filter capacitors in series in order to achieve a specific capacitance at a higher voltage rating than is available in an individual capacitor. The permissible ripple current for the series combination is limited to the lowest ripple current rating of the individual capacitors. Unfortunately, the wide ESR and capacitance tolerance of aluminum electrolytics does not result in equal distribution of voltage across the series capacitors, so they must be shunted with resistors. The current through the shunt resistors should be ten times greater than the rated capacitor leakage current at the temperature at which the capacitors actually operate. Both ESR and leakage resistance will decrease with increasing temperature.

Aluminum capacitors can be reconditioned (the dielectric "re-formed")[3] by applying the rated working DC voltage (WVDC) through a 10W current-limiting resistor over an extended period of time.

The predominant failure mode for aluminum capacitors is a short circuit caused by flaws in the oxide film or separator paper that lead to dielectric breakdown. Another life-limiting failure in electrolytic capacitors results from the leakage of the electrolyte due to high internal temperature. Electrolyte leakage is a mechanical failure and is most commonly caused by a deficient compression seal, leakage at the weld on the bottom of the case, or leakage around the aluminum terminals in plastic seals.

REFERENCES

1. "Considerations for a High Performance Capacitor," R. Marsh, www.capacitors.com/consider/consider.htm.

2. "Picking Capacitors," W. Jung, R. Marsh, *Audio Magazine*, Feb. 1980, pp. 52–63/Mar. 1980, pp. 50–63.

3. "Electrolytic Capacitor Reforming Unit," C. Hansen, *Glass Audio* 6/97, pp. 22–27.

Application of a high (greater than 1.5V) reverse voltage will result in a loss of capacitance. An aluminum capacitor can absorb higher reverse voltages for a short period of time, but their failure mode can be explosive when internal pressure builds up after an excessive internal temperature rise.

TANTALUM

The solid tantalum capacitor is another common type of electrolytic capacitor. Tantalum powder is pressed into a cylindrical shape, then sintered at very high temperatures, resulting in a porous anode. This tantalum surface is electrochemically converted to tantalum pentoxide, which serves as the dielectric. The anode is then coated with manganese dioxide, which serves as the electrolyte in solid form.

Thermal runaway can occur if minute contamination particles such as tantalum carbide are present, preventing the complete anodization into tantalum pentoxide. Local current flow in these oxide faults can increase over time until total failure occurs.

Solid tantalum capacitors usually require current limiting of about 3Ω per applied volt (1/3A) to prevent failure due to scintillation effects. This can also be accomplished by paralleling the tantalum capacitor with a large aluminum cap that limits the rate of voltage rise, thus limiting inrush current into the tantalum capacitor.

Wet slug tantalum capacitors with silver packages are notorious for major and frequent failures due to silver migration across the dielectric. The silver eventually shorts to the tantalum anode slug. If sufficient current is available, a catastrophic failure can occur due to boiling of the wet electrolyte. Wet slugs should never have a reverse voltage applied to them.

Solid tantalum capacitors can withstand limited reverse voltage: up to 15% of rated voltage at room temperature, decreasing to 5% or rated voltage at high or low temperatures.

Next month I'll continue this survey of electronic components, and offer suggestions about how to prevent failure of parts. ❖

A Planar-Triode Phono Preamp

An unlikely looking tube is the star performer in this smooth-sounding preamp. **By Eric Barbour**

PHOTO 1: 6S17K-V tube.

After having worked in tube audio as a "professional," i.e., getting paid by an actual employer that sold tubes, I've gained some useful insights into tube design. Some of this knowledge actually applies to the common circuit designs that see frequent use. More to the point, I learned the personal motivations of many designers of said equipment.

You might be surprised to learn how much of the use of a given tube is based upon pricing, not on technical merit. All of the largest OEMs of high-end audio equipment seem unwilling to pony up a large sum for development of a tube to meet their strict requirements. Instead, they prefer to constantly call tube distributors and factories and try to wheedle them into selling for a lower price. If a fraction of this effort was used to redesign their products using available tubes, all of their problems would go away.

The hobbyist has a tremendous advantage here, by simply keeping an open mind. Instead of chasing down a price that is one penny lower for Russian 6922s, he can use tubes that are readily available, and are vastly better performers than the lowly 6922. Ironically, some outstanding Russian tubes are never used in audio equipment, yet happen to be excellent for audio use.

HISTORY

The 6922/6DJ8 has been used for years in audio, because of its low cost, detailed sound qualities, and (alleged) low noise figure. I find this ironic, because it is a frame-grid triode, and there are far more sophisticated and quieter frame-grid triodes. The first frame-grid triode, in fact, was a very odd-looking tube in a metal canister—the Western Electric 416A of 1948. In spite of its very high Gm (equating to very low noise), audiophiles have no interest in it or its metal-cased cousins. Instead, they seek frame-grid types in conventional glass envelopes.

While a WE 437A can sell for as much as $600 on eBay, a 416D (a far superior device) usually sells for less than

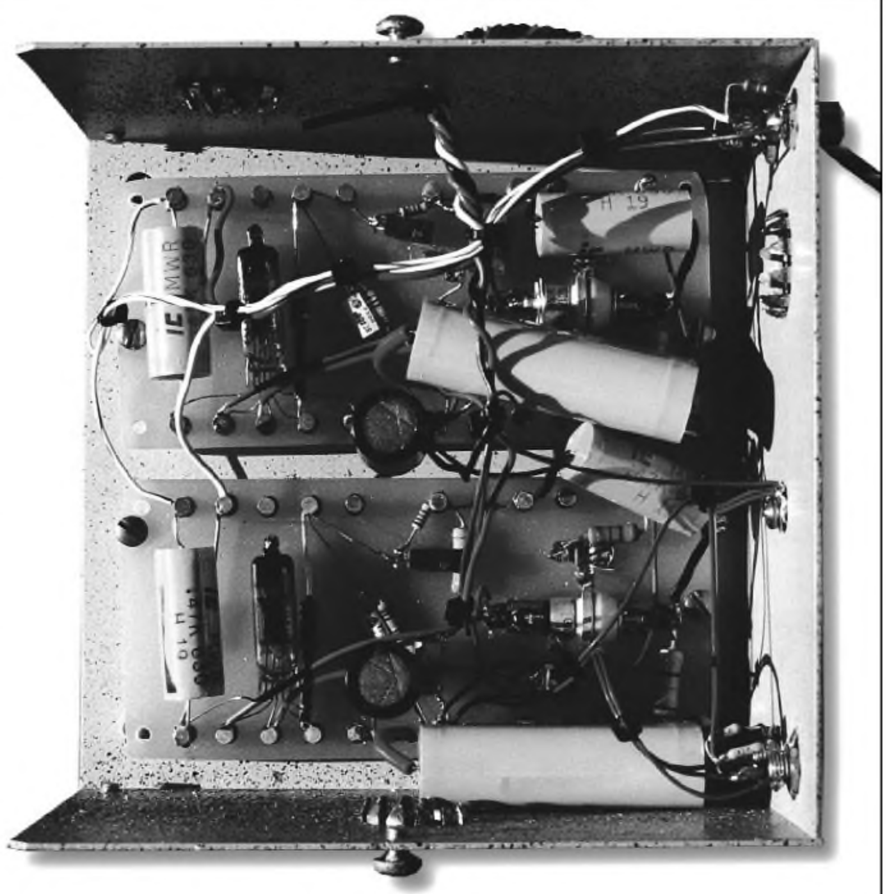

PHOTO 2: Interior of planar-triode phono preamp.

$100. Reasons? Apparently, the difficulty of finding 416-series sockets or other hardware is a factor, as well as the simple and childish fact that you can see the glow of the 437A's heater.

The 416A led to several series of ceramic-metal triodes, all having a planar structure rather than the conventional glass-tube coaxial design. A planar triode is a "sandwich" of parts, brazed together in a vacuum chamber using special tooling. For example, General Electric developed the 7077 in 1951 for low-noise RF amplifiers. In spite of the "obsolescence" of vacuum tubes, the 7077 and its descendants are still manufactured in Owensboro, Ky., by MPD, the firm that took over GE's tube manufacturing.

These planar triodes are made almost exclusively for the front ends of military pulsed radars, due to the high cost of manufacturing a receiving tube in the US, especially an exotic one. Few audiophiles have ever seen a 7077, and I enjoy their expressions when they are handed one. It is the size of an aspirin tablet, and is made of white ceramic and gold-plated metal contacts. Some people have called it a "funny looking transistor."

I can guarantee that you will never see 7077s or their many derivatives in audio equipment, due to wholesale prices in the US $600–1000 region. As I said, most manufacturers of high-end audio equipment usually are only interested in reducing manufacturing costs, not producing amplifiers that sound or perform better.

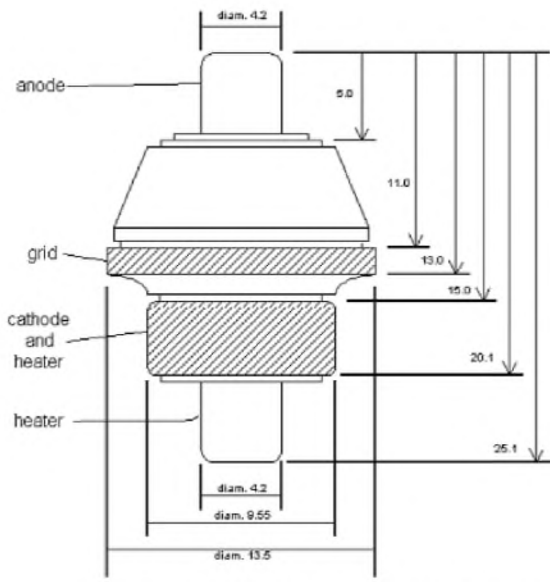

FIGURE 1: Mechanical dimensions of 6S17K-V planar triode. All dimensions in mm (not to scale).

AVAILABILITY

The high cost of such planars need not be an impediment to the home constructor. Many tube factories made their own planars over the years. A series of unique planar triodes was made at the Novosibirsk factory in Russia—these are similar internally to the American 7077 family, yet quite different in outward appearance. I have recently discovered that surplus Russian planars are available at very reasonable prices.

The easiest one to obtain is the 6S17K-V (usually called 6C17bB by Russian dealers). This device resembles a rocket nose-cone (*Photo 1*), not a vacuum tube. Intended for microwave frequencies, it has no base or contact pins—electrical contacts are smooth surfaces to minimize path lengths, thus reducing parasitic inductances and transmission-line effects (*Fig. 1*).

I have learned that surplus Russian tube dealers usually have a supply of 6S17K-Vs for sale for about $30 apiece. These tubes are apparently no longer manufactured, yet I understand there are tens of thousands

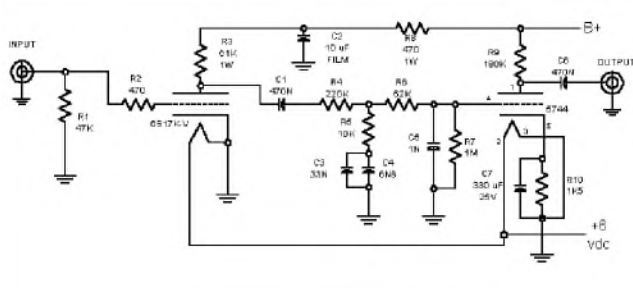

FIGURE 2: Schematic for planar-triode phono preamp. One channel shown. Other channel uses resistors R10–R20, capacitors C10–C17.

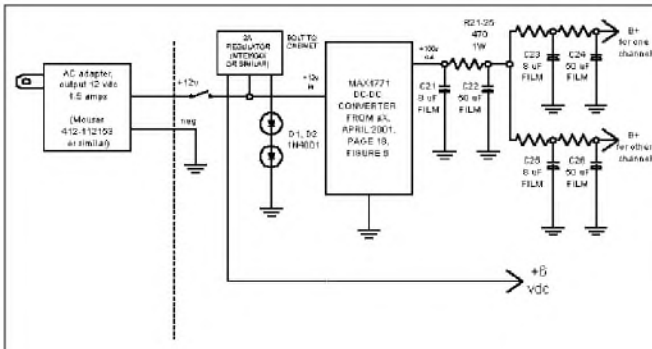

FIGURE 3: Power supply for planar-triode preamp.

available in Russia in surplus form. The price is so attractive that the tube lends itself to low-noise preamplifier design, in spite of the need to fabricate a special mounting assembly with spring contacts.

PREAMP

I constructed a basic zero-feedback phono stage with a 6S17K-V as the input amplifier, feeding a passive equalization section (*Fig. 2*). Because the 6S17K-V has its cathode connected to one side of its heater, a clean source of DC heater power is recommended. The output stage is made with a subminiature triode, type 5744 (also available in Russian form under the number 6C7bB).

Examination of the 6S17K-V plate curves indicated a plate resistance in the center of the curve area of approximately 13.2kΩ. The data indicates 2kΩ as the usual load, for RF amplifier service. In audio design, you need to minimize distortion, and commonly this happens at the expense of output impedance. For small-signal triodes, a load of approximately four times the plate resistance is a standard practice. Thus, a suitable anode load impedance would be 51kΩ.

The heater-cathode connection seems to complicate biasing. However, after running the tube in a breadboard, I found that it generates so much self-bias that its cathode could be directly grounded. The grid, with a 47k resistor to ground, then floats at approximately 0.5V below ground, and with a B+ of 140V and a 51k resistor for anode loading, the anode idled at approximately 50–60V—one-half the B+ supply. This is apparently caused by secondary emis-

sion and other grid effects, sometimes collectively called "contact potential." Thus, no biasing network or negative bias voltage was needed.

The EQ circuit is a classic which has been used many times. My previous experience with it was in my nuvistor phono preamp (*Glass Audio* 4/93, p. 6). Because of the very high voltage gain of the 6S17K-V, you can dispense with one whole gain stage. The output subminiature-tube stage provides moderate gain and a reasonably low output impedance.

CONSTRUCTION

Each channel was constructed on a custom IAG terminal board. The special mounting requirements of the 6S17K-V required some ingenuity.

I drilled holes in the IAG boards next to the location of the tube, as it is laid down on the board. A #4 screw and nut at each location holds a small contact finger cut from brass shim stock. The contact is bent to firmly contact the appropriate ring on the tube. If this is done properly, all the contacts will be solid, and the tube will be very firmly held in place. The submin tubes had long wire leads, which were wrapped around convenient turret terminals.

I mounted the two channel boards in

6S17K-V CHARACTERISTICS

Heater: 6.3V 300mA
Anode voltage, maximum: 200V DC
Cathode current, maximum: 11mA
Anode dissipation, max: 2W
Grid dissipation, max: 0.1W
Transconductance: 14mA/V (14ms)
Mu, approximate: 80–185
Noise figure at 3GHz: 16.5dB
Microphony noise, max: 30mV (with anode load 2kΩ)
Input capacitance: 3pF
Output capacitance: 0.015pF
Grid-anode capacitance: 1.5pF

SUPPLIERS

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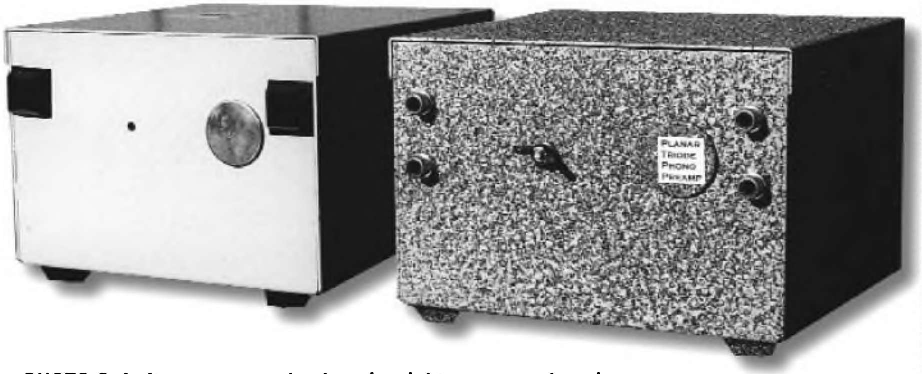

PHOTO 3: Left, power-supply chassis; right, preamp chassis.

a massive mild-steel box, which appears to be a surplus electrical enclosure (*Photo 2*). In spite of past claims that aluminum enclosures result in better sound, you need proper magnetic shielding here, so a steel cabinet is called for.

The power supply, which provides a great deal of filtering for low noise, is in a separate cabinet (*Photo 3*). This was a good place to apply one of the circuits developed for my switching-supply article (April 2001 *aX*), so I used Fig. 6, the supply based around a Maxim MAX1771 and a small transformer intended for making high voltage for electronic automotive ignition (*Fig. 3*). This supply easily makes about +100V at 15mA, plenty for this circuit. By using heavy filtering and decoupling, the need for separate voltage sources for the two channels and for active regulation (often an additional source of noise) is eliminated.

I measured the B+ supply under load and found that with no signal into the preamp, noise and hum on the B+ line was less than 5µV. A three-terminal regulator drops the +12V to approximately 6V for heater power. This IC is a 5V device—by using the old trick of forward-biased diodes between its return line and ground, the actual voltage is boosted to about 6.2V DC.

The +12V is obtainable from any suitable AC-DC plug-in power pack, or from a bench supply. This scheme even raises the possibility of a tube audio system powered from a 12V battery, making its use in an RV reasonable. (Provided the turntable is not used while the RV is in motion, of course.)

How does it sound? Very clean and transparent, much like my *GA* nuvistor preamp...and with a major improvement: the planar triodes are very non-microphonic, while nuvistors are notorious for having microphony problems. I mounted my previous nuvistor preamp's PC board on rubber shock mounts for a very good reason: high amplifier volumes easily gave a feedback howl. This circuit does not appear to need such mounting considerations.

I believe that this preamp gives some very costly and finicky commercial preamps a serious run for their money. Building this project will both save a large sum of money and permanently eliminate the audiophile's need to plug 6922s into a preamp and tap them each to find one that is less microphonic than the others. ❖

PARTS LIST

(R1–R10 and C1–C7 are for one channel; R11–R20 and C11–C17 for the other. All resistors ½W metal film unless noted; all capacitors 150V or greater metalized film unless noted.)

R1, R11	47k
R2, R12, R8, R18, R21–R25	470, 1W
R3, R13	51k, 1W
R4, R14	220k
R5, R15	10k
R6, R16	62k
R7, R17	1MΩ
R9, R19	180k
R10, R20	1k5
C1, C10, C6, C16	470n
C2, C12	10µF
C3, C13	33n
C4, C14	6.8n
C5, C15	1n
C7, C17	330µF 25V electrolytic
D1, D2	1N4001
Regulator	2A large TO220 package (NTE1934X or similar)
DC-DC converter	see text

Misc: (2) steel enclosures, AC adapter (Mouser 412-112153 or similar), SPST power switch rated 2 amps or more, (4) RCA panel-mount connectors, (2) IAG custom breadboards, wire.

A First-Order 3-Way, Part 1

This audio veteran presents a relatively simple speaker construction project using first-order crossovers. **By J. L. Markwalter, Jr.**

I made this speaker to replace a multichannel horn system I built in the 1950s. The old speaker still sounds fairly good to me but, of course, does not measure up to present standards. A lot of progress has since been made in components and in design methods.

With this new design (*Photo 1*), I have paid as much attention as I could to matters within my control that influence performance. Judging how a loudspeaker sounds is undeniably subjective, but attention to detail can be expected to pay off in better sound. I am quite pleased with the results, but also had some of my young friends (doubtless with better hearing than mine) do critical listening. The speakers received high grades from their "tests."

DESIGN CONSIDERATIONS

I decided not to build another horn speaker largely for technical reasons. Horns excel in stadiums and auditoriums, for example, where large-scale sound is needed. In some present situations where limited audio power is available but high sound output is needed, the high-efficiency characteristic of horns might be the answer. Electroacoustic efficiency is now of little consequence, however, as high-quality high-power audio is relatively cheap. About 10W was the norm yesteryear, now 50 or 100W or more is common.

ABOUT THE AUTHOR

J. Laurence Markwalter received bachelor's and master's degrees in Electrical Engineering from Georgia Tech. He was a career electrical/electronics engineer in industry and pursued aspects of audio as a hobby. He is a Life Senior member of the IEEE and a registered professional engineer in Maryland.

Horns designed for extended low-frequency response are relatively large compared to direct radiator speakers. Low-frequency horns for home use are usually folded to make them an acceptable size. Unfortunately, folding gives rise to response irregularity, and some designs also require room corner placement for good low-frequency performance.

The path length of a low-frequency horn is typically 5–7′, making it difficult, or impossible, to equalize time delays among the channels in a multichannel system to prevent phase and delay distortion.[1] It has been established that phase and delay distortion affect the quality of speech, music, and sound effects.[2] It is partly for these reasons that direct radiator drivers are most widely used today.

In a multichannel speaker, all drivers would ideally occupy the same space, which is, of course, impossible. The ideal speaker would have only a single driver that could cover the entire frequency range, have perfect transient response, and provide broad, constant polar coverage, among other things. Unfortunately, no such driver is known to exist, at least not to me. Perhaps good headphones come close in many ways, although this would be inconvenient for general listening. The usual approach is to apportion the audio spectrum to be reproduced into channels handled by separate drivers—three in my design.

FIRST-ORDER PERFORMANCE

First-order crossover networks are featured in this design. I do not recall seeing a speaker project in *Speaker Builder* that uses first-order crossover networks. Why first-order crossovers? The theoretical advantage is perfect transient response. Since music and speech are largely transient by nature, realization of first-order performance should result in better sound quality.

A first-order system requires first-order electrical crossovers[3] and a suitable choice of drivers, which, unfortunately, is somewhat limited. System transient response depends upon the transient response of the drivers themselves as well as the crossovers. Good drivers are indispensable.

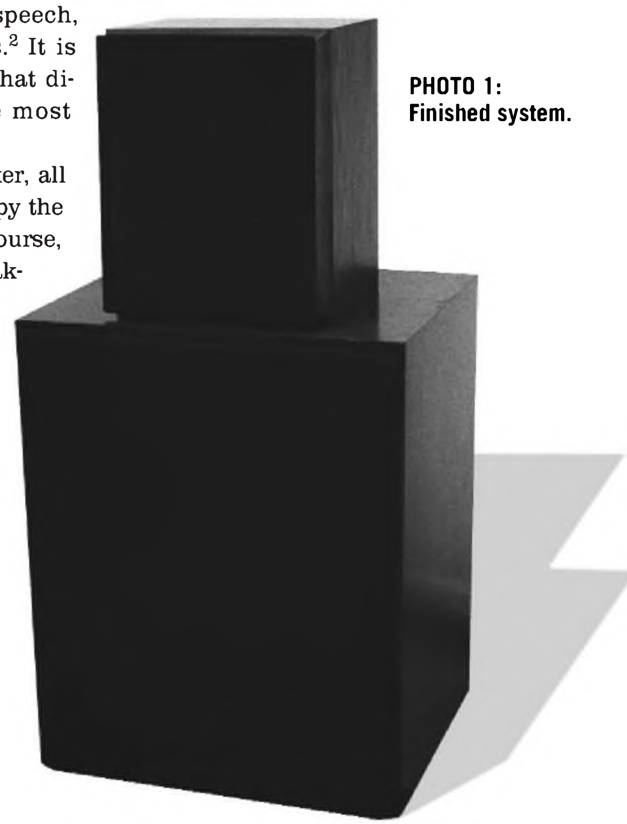

PHOTO 1: Finished system.

In a first-order system the signal to each driver is characteristically down 3dB at its crossover frequency, and approaches a falling rate of 6dB per octave further out of band. The output of the low-pass network will be .707, −45° phase, at the crossover frequency, decreasing toward 0.0, −90° phase, as the frequency increases. The output of the high-pass network will be .707, +45° phase, at the crossover frequency, increasing toward 1.00, 0° phase, as the frequency increases.

The two signals will combine at all frequencies to 1.00, 0° phase. This may not be apparent but is mathematically true. This is why there will be no distortion of the audio electrical signal using first-order crossover networks, thus approaching perfect system transient response with improving drivers.

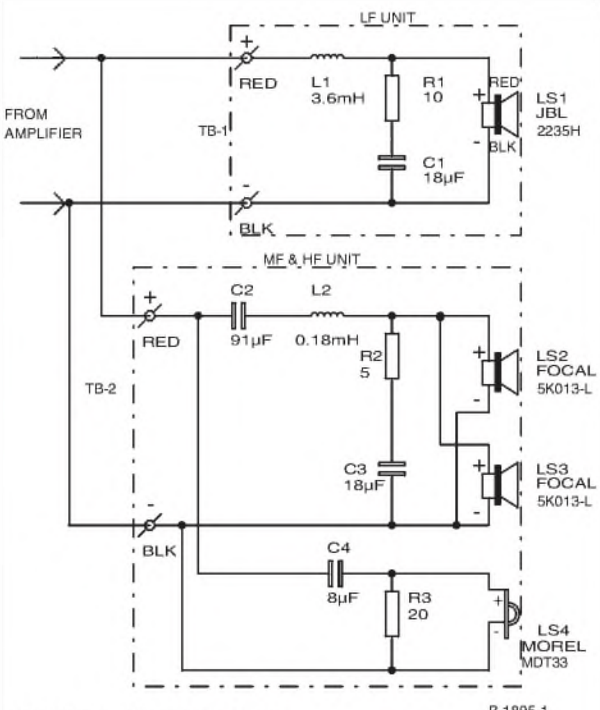

FIGURE 1: Speaker system.

Since the signal versus frequency to each driver falls at a rate approaching 6dB/octave, there is substantial overlap on each side of the crossover frequency of an octave or more. The drivers themselves must have flat response to this extent and must be capable of handling greater signal input beyond crossover than would be required in systems having higher-order crossovers. Relatively few suitable drivers are available.

To minimize response ripples, it is important that the path length from all drivers to the listener be as nearly equal as possible and the phase response of the drivers themselves match closely. These requirements can be met sufficiently well so that good results are obtainable. Note that response ripples are likely to occur in any listening situation with any system due to multipath reflections from walls, ceiling, floors, and objects. In fact, this and reverberation characteristics contribute importantly to the character of the listening environment and are not bad in themselves.

The wavelength at the lower crossover frequency (400Hz) is 33.75″, and that at the higher crossover (4kHz) is 3.38″. You should locate the drivers close to each other in the same plane, especially the mid- and high-frequency drivers, because of the short wavelength at 4kHz. It was reported from work done in motion picture sound systems[2] that a time delay of less than 2ms between adjacent channels could not be perceived in reproducing impulsive sounds. Two milliseconds corresponds to a path difference of 27″. From the time delay consideration, there should be no difficulty arising with the configuration of the subject design.

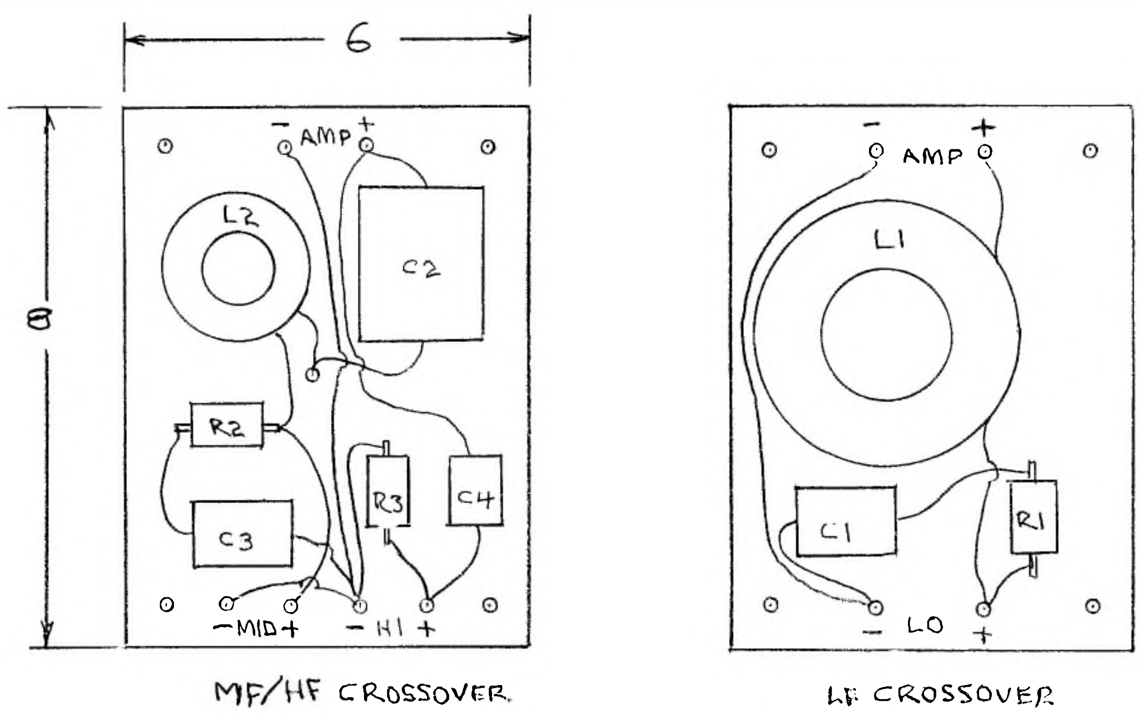

FIGURE 2: Crossover networks.

CHOOSING THE DRIVERS

The system drivers are:

JBL 2235H, low frequency, covering 30–400Hz
Focal 5K013-L, mid frequency, covering 400–4kHz
Morel MDT33, high frequency, covering 4kHz–20kHz

The JBL 2235H is a 15″, 8Ω driver having a rated sensitivity of 93dB, 1W, 1m. The 2235H is designed expressly as a low-frequency driver. The manufacturer's curves show the frequency response to be essentially flat to 1kHz up to 45° off-axis.[4] Its Thiele/Small parameters permit design for operation to 30Hz in a moderate-size enclosure. A notable feature of the 2235H is a flux stabilizing ring about the pole piece to reduce production of second harmonic distortion.

The Focal 5K013-L is a 5″, 8Ω midrange driver having a published sensitivity of 88.6dB, 1W, 1m. The frequency response curves are essentially flat from 150Hz to 8kHz up to 45° off-axis.[5] The 5K013-L has a phase plug at the cone apex mounted to the center pole piece to improve the high-frequency polar coverage.

Since its sensitivity is about 4.4dB less than that of the 2235H, I used two drivers in parallel, placed close together, in the midrange. The system electrical impedance is therefore 4Ω over the midrange. The parallel drivers increase the voltage sensitivity 3dB over a single driver, and being placed close together further increases their sensitivity.[6] I would have preferred a single 8Ω driver for the midrange but did not find one having sufficient sensitivity, broad enough frequency response, and sufficient polar coverage to be compatible with the 2235H and MDT33.

The Morel MDT33 is an 8Ω soft dome tweeter. Published sensitivity is 92.5dB,

TABLE 1
LOW-FREQUENCY CHANNEL CALCULATED ELECTRICAL RESPONSE

FREQUENCY	DRIVER IMPEDANCE	CORRECTED DRIVER IMPEDANCE	OUTPUT (VECTOR)	OUTPUT (dB)
20Hz	8.92 –j6.43	8.66 –j6.50	1.03, –1.97°	+0.26
30	9.02 –j3.51	8.79 –j3.73	1.03, –3.85°	+0.26
40	22.04 +j19.33	18.50 –j19.32	1.02, –1.38°	+0.17
50	18.03 –j25.13	13.84 –j23.03	1.04, –1.29°	+0.34
70	8.29 –j8.06	7.31 –j7.95	1.11, –6.35°	+0.91
100	7.76 –j3.60	7.08 –j3.98	1.11, –15.70°	+0.91
150	7.97 –j1.69	7.29 –j2.49	1.05, –25.92°	+0.42
200	8.19 –j0.52	7.51 –j1.73	0.96, –33.39°	–0.35
250	8.22 +j1.75	8.08 –j0.096	0.82, –35.22°	–1.72
300	8.44 +j1.94	8.13 –j0.30	0.78, –40.71°	–2.16
400	8.92 +j3.53	8.78 +j0.064	0.69, –45.64°	–3.22
500	8.97 +j4.54	9.41 +j0.58	0.62, –48.10°	–4.15
600	9.17 +j5.43	8.91 +j0.11	0.55, –56.25°	–5.19
1kHz	10.87 +j6.86	8.17 –j0.54	0.35, –73.50°	–9.12
1.2	11.40 +j9.64	8.61 –j0.33	0.31, –74.41°	–10.17
2	13.04 +j15.72	8.70 +j0.054	0.19, –78.77°	–14.42

1W, 1m, which is very nearly that of the 2235H low-frequency driver and paired 5K013-L drivers. The curves show the frequency response essentially flat from 1kHz to 15kHz 45° off-axis, flat to 20kHz on-axis.[5]

The MDT33 is totally enclosed at the rear by a metal cup sealed to the body, which is convenient for operation in an enclosure shared with open back drivers as in this case. The resonant frequency is 500Hz, well below the 4kHz crossover frequency and capable of handling a considerable signal input well below the crossover frequency. It is termed an "extremely fast" tweeter, having a rise time of 10μs.

TABLE 2
MID-FREQUENCY CHANNEL CALCULATED ELECTRICAL RESPONSE

FREQUENCY	DRIVER IMPEDANCE	CORRECTED DRIVER IMPEDANCE	OUTPUT (VECTOR)	OUTPUT (dB)
50Hz	7.47 +j7.16	8.10 +j7.09	0.37, +115.0°	–8.56
70	24.84 –j4.53	22.72 –j8.49	0.59, +35.45°	–4.58
100	6.72 –j8.07	5.63 –j7.74	0.37, +23.41°	–8.64
150	4.37 –j2.98	3.93 –j3.08	0.33, +36.83°	–9.63
200	4.03 –j1.92	3.66 –j2.12	0.38, +40.94°	–8.40
250	3.97 –j1.07	3.66 –j1.40	0.44, +44.84°	–7.13
300	4.03 –j0.53	3.75 –j0.98	0.52, +45.26°	–5.68
400	4.04 +j0.60	3.97 –j0.11	0.70, +43.86°	–3.10
500	4.09 +j0.0	3.71 –j0.75	0.72, +33.39°	–2.85
600	4.23 +j.63	3.93 –j0.41	0.83, +28.01°	–1.62
1kHz	4.57 +j1.74	4.16 –j0.23	0.98, +8.36°	–0.18
1.2	4.78 +j2.40	4.34 –j0.17	1.00, +1.32°	0.00
2	5.29 +j3.13	3.95 –j0.26	0.96, –19.70°	–0.35
3	5.89 +j4.75	3.97 –j0.074	0.81, –35.34°	–1.83
4	6.59 +j6.04	4.00 +j0.026	0.70, –45.40°	–3.10
6	7.65 +j8.32	4.09 +j0.18	0.52, –55.95°	–5.68
8	8.87 +j10.17	4.15 +j0.24	0.42, –62.10°	–7.54
12	10.99 +j13.23	4.27 +j0.29	0.30, –68.80°	–10.46
16	12.93 +j15.71	4.35 +j0.31	0.23, –72.60°	–12.77

TABLE 3
HIGH-FREQUENCY CHANNEL CALCULATED ELECTRICAL RESPONSE

FREQUENCY	DRIVER IMPEDANCE	CORRECTED DRIVER IMPEDANCE	OUTPUT (VECTOR)	OUTPUT (dB)
1kHz	11.91 +j0.00	7.46 +j0.00	0.35, +69.43°	–9.11
1.2	9.72 +j0.00	6.54 +j0.00	0.37, +68.47°	–8.64
2	8.18 +j0.87	5.82 +j0.44	0.52, +62.83°	–5.68
3	7.21 +j0.00	5.30 +j0.00	0.62, +51.36°	–4.15
4	6.82 +j0.00	5.09 +j0.00	0.71, +44.36°	–2.97
6	7.14 +j0.55	5.27 +j0.30	0.87, +33.04°	–1.21
8	7.36 +j2.17	5.47 +j1.15	0.99, +25.60°	–0.09
12	8.08 +j3.35	5.95 +j1.67	1.04, +10.67°	+0.34
16	8.92 +j3.95	6.42 +j1.85	1.04, +10.67°	+0.34

TABLE 4
CALCULATED COMBINED ELECTRICAL RESPONSE OF ALL CHANNELS

FREQUENCY	LF CHANNEL	MF CHANNEL	HF CHANNEL	COMBINED	DB
20Hz	1.03, –1.97°			1.03, –1.97°	+0.26
30	1.03, –3.85°			1.03, –3.85°	+0.26
40	1.02, –1.38°			1.02, –1.38°	+0.17
50	1.04, –1.29°	0.37, +115°		0.94, +19.45°	–0.54
70	1.11, –6.35°	0.59, +35.45°		1.59, +7.9°	+4.03
100	1.11, –15.70°	0.37, +23.41°		1.42, –6.22°	+3.05
150	1.05, –25.92°	0.33, +36.83°		1.22, +11.49	+1.73
200	0.96, –33.39°	0.38, +40.94°		1.12, –14.39°	+0.98
250	0.82, –35.22°	0.44, +44.84°		1.00, –9.41°	0.0
300	0.78, –40.71°	0.52, +45.26°		0.97, –8.28°	–0.26
400	0.69, –45.64°	0.70, +43.86°		0.99, –4.80°	–0.09
500	0.62, –48.10°	0.72, +33.39°		1.02, –3.68°	+0.17
600	0.55, –56.25°	0.83, +28.01°		1.04, –3.72°	+0.34
1kHz	0.35, –73.50°	0.98, +8.36°	0.35, +69.43°	1.20, +6.44°	+1.58
1.2	0.31, –74.41°	1.00, +1.32°	0.37, +68.47°	1.22, +3.22°	+1.73
2	0.19, –78.77°	0.96,–19.70°	0.52, +62.83°	1.18, –2.30°	+1.44
3		0.81,–35.34°	0.62, +51.36°	1.05, –0.86°	+0.42
4		0.70,–45.40°	0.71, +44.36°	1.00, –0.11°	0.0
6		0.52,–55.95°	0.87, +33.04°	1.02, +2.44°	+0.17
8		0.42,–62.10°	0.99, +25.60°	1.09, +2.97°	+0.75
12		0.30,–68.80°	1.04, +15.55°	1.11, +0.05°	+0.91
16		0.23, –72	1.04, +10.67°	1.09, –1.41°	+0.75

LOW-FREQUENCY ENCLOSURE

I chose the drivers based on their published performance characteristics and some personal familiarity with the JBL 2235H and Focal 5K013-L. I chose the Morel MDT33 solely from published data, having never seen or heard one before. I employed Boxmodel 96 for calculations using the Thiele/Small parameters for both the low-frequency and the mid/high-frequency enclosures, and then calculated the low-frequency performance versus box volume for them. The MDT33 enclosure volume is that of the cup enclosing the rear of the driver, so you have no choice here. Its published performance characteristics appeared good for the application, however.

I chose a bass reflex enclosure for the JBL 2235H since that will provide good response to 30Hz in a moderate-size enclosure. I made the following calculations using Boxmodel from the manufacturer's Thiele/Small data:

f_S	20Hz
V_{AS}	16.2ft^3
S_D	138 in^2
Q_{ES}	0.28
Q_m	2.50
X_{MAX}	0.33″
V_{AB}	5.0ft^3
f_B	25Hz

I calculated frequency response using various combinations of assigned box volume and box resonant frequency. The combination giving maximally flat response is 3.237ft^3 and 32.7Hz. This combination provides very flat response to 60Hz but rolls off to –9dB at 30Hz, which is greater than that desired.

The combination I judged best is 5ft^3 and 25Hz box resonance. Response with this combination is down about 1.5dB at 60Hz and approximately 5.5dB at 30Hz. I chose 5ft^3 and 25Hz box resonance for the design. The measured electrical data herein applies to this combination.

The box resonance is set by the vent in the front of the driver baffle, which is made of plastic pipe 4″ ID × 7.84 (use 7

27/32) length. I computed the length with Boxmodel from the chosen diameter. The internal dimensions of the enclosure are related to each other proportional (approximately) to the cube root of 2, i.e., 1 × 1.26 × 1.59, to spread the standing waves within the enclosure evenly within the spectrum.[7]

I mounted the driver to the inside surface of the baffle, but you can mount it outside if you prefer, making sure to increase the spacing of the grille frame from the enclosure accordingly to prevent possible contact of the cone with the grille cloth during a large excursion. The manufacturer recommends a 13 31/32″ cutout for front mounting. Inside mounting should have no effect on response smoothness in the operating frequency range of the low-frequency channel due to panel thickness.[8]

MID/HIGH-FREQUENCY ENCLOSURE

The mid- and high-frequency drivers share a common enclosure. The enclosure volume is operational with only the Focal 5K013-L mid-frequency drivers because the Morel MDT33 high-frequency driver is enclosed at the rear, and is thus acoustically isolated. The enclosure is not vented.

I made midrange response calculations up to the lower crossover frequency using Boxmodel 96, whose upper limit of frequency analysis calculations is 400Hz, which unfortunately is also the lower crossover frequency of the midrange. It would have been nice to have the analysis range extended to a frequency at least an octave above cutoff. Fortunately, both the low- and mid-frequency drivers are in their respective response flat ranges at 400Hz. Below 400Hz the calculated acoustic response of the 5K013-Ls begins to fall off, and is down about 7.5dB at 70Hz, which should reduce the "hump" in the mid-frequency crossover calculated electrical response.

I calculated the Focal 5K013-L performance using the Thiele/Small published data:

f_S	40.6Hz
V_{AS}	0.671ft^3
S_D	12.1 in^2
Q_{ES}	0.27
Q_m	3.14
X_{MAX}	0.217″
V_{AB}	0.849ft^3

I sized the enclosure to be large enough to mount the three drivers close together on a common baffle, while the inside dimensions are related to each other proportional (approximately) to the cube root of 2, i.e., 1 × 1.26 × 1.59, to spread the standing waves within the enclosure evenly within the spectrum.[7] The actual dimensions are shown on the drawings. I mounted the drivers to the front surface of the baffle and beveled the edges of the baffle 45°. I beveled the enclosure body as a continuation of the baffle bevel to minimize diffraction effects.[9] Boxmodel could not calculate the acoustic performance of the Morel MDT33.

THE CROSSOVER NETWORKS

In this section I discuss in detail the influence of driver phase response on frequency response in a multichannel system. A demonstration of it is described in reference 10. System response is the

PARTS LIST

PART	VALUE	DESCRIPTION	SOURCE	PART #
C1, C3	18μF ±5%	250V polypropylene	Zalytron	
C2	91μF ± 5%	250V polypropylene	Zalytron	
C4	8μF ±5%	250V polypropylene	Zalytron	
L1	3.6mH ±1%	14 ga. air core coil	Zalytron	
L2	0.18mH ±1%	14 ga. air core coil	Zalytron	
LS1	8Ω	low frequency driver	JBL	2235H
LS2, LS3	8Ω	midrange driver	Focal	5K013-L
LS4	8Ω	high frequency driver	Morel	MDT33
R1	10Ω ±1%	25W, metal cased	Mouser	71-RH25-10
R2	5Ω ±1%	25W, metal cased	Mouser	71-RH25-5
R3	20Ω ±1%	25W, metal cased	Mouser	71-25RH-20
TB1, TB2		terminal assembly	Parts Express	260-303
perfboard		6 × 8″ pre-drilled board	Radio Shack	276-1396
vent		4″ plastic pipe	Home Depot	
flange		4″ plastic flange w/mtg holes	Home Depot	

combined result of driver and crossover network amplitude and phase responses.

The impedance of the drivers must be measured operating in their enclosures prior to designing the crossover networks. I used the method of measuring impedance without a bridge as I described in *Speaker Builder*.[11] This method provides resistance and reactance values at the measurement frequencies. As is well-known to speaker builders, speaker impedance is not constant, not purely resistive, and the published value is only nominal.

The frequency response of drivers is customarily measured with constant voltage applied to the unit under test as the frequency is varied,[12] and is understood to apply to published curves. Driver acoustic outputs should be flat beyond the cutoff frequency(ies) of their associated crossover networks. In a first-order system the network for a particular driver must supply voltage reduced to .707 (–3dB) at 45° (plus or minus as appropriate) at the cutoff frequencies, approaching 1.000 or 0.000 away from cutoff.

The design of a first-order crossover network would be a simple matter if the driver impedance were a pure resistance and constant with frequency. For the low-frequency driver a series inductance could be used whose reactance at the cutoff frequency is made equal to the driver resistance; for the high-frequency driver a series capacitance could be used with reactance at the cutoff frequency made equal to the driver resistance. Likewise, for the midrange driver a series inductance and capacitance combination could be used in series with it with net reactance made equal to the driver resistance at the two crossover frequencies.

For real-world speakers using first-order crossovers, an impedance correction network must usually be used with each driver to make the voice-coil impedance appear essentially resistive at the crossover frequency and relatively constant an octave or so beyond. This is usually only approximately attainable with a simple R-C network across the voice-coil terminals.

In an ideal system each driver might be supplied from separate power amplifiers, providing near-zero source impedance to the drivers, with the crossover filters at the inputs of the amplifiers. In this way, driver impedance would have no influence on response and driver damping would be fully preserved. This is the best way to go, but it is also the most expensive. Cost, no doubt, is one reason passive crossovers are the most popular.

Figure 1 shows the schematic diagram of the system. The three networks are contained in the two system enclosures as indicated by the diagrams. *Tables 1*, *2*, and *3* present the measured driver impedances, the impedances as corrected by the components in parallel with the voice coils, and calculated vector outputs at the drivers. Each crossover network is an equivalent L-filter having the corrected driver impedance as its shunt arm and a reactive circuit element as its series arm.

The driver impedances and corrected values are expressed in rectangular coordinates; i.e., in resistive and reactive components. The outputs are expressed in vector values—magnitude and phase angle. You must add the individual outputs vectorially to obtain the combined electrical output. *Table 4* presents the vector outputs of the three channels combined.

I found simple R-C networks to adequately compensate the low- and mid-frequency driver impedances, and only a resistor across the high-frequency driver was necessary to compensate (i.e., stabilize) its impedance. It is important that the corrected driver impedances be as nearly purely resistive as possible at the crossover frequencies and an octave or so beyond.

More sophisticated (and complex) networks that stabilize driver impedances over much wider frequency ranges have been derived by an associate and fellow hobbyist.[13] The corrected impedance, in conjunction with the series reactance of the crossover network, should make the frequency response .707, 45° phase, at the crossover frequencies. This has been realized to a satisfactory degree using the simpler networks, however.

CONSTRUCTION

The crossover networks are constructed on 6 × 8″ perfboard. *Figure 1* shows the electrical connections, and *Fig. 2* shows the parts arrangement. The resistors are metal encased and have mounting ears, and should be attached to the board with small bolts or tied to it. You should attach the inductors and capacitors to the boards with cable ties or lacing cord. I used small brass bolts as terminals and all connections to and from them are made with ring solder lugs. The perforations in the boards are too small to be used for mounting and must be drilled larger as needed. However, the hole pattern is handy as a kind of location grid.

TESTING

I performed the acoustic testing in my living room. Testing was difficult because of room reflections, but I found that I obtained best results with the microphone about 40″ from the MID/HF enclosure centered on its baffle. I chose close testing in hopes the incident sound would predominate sufficiently over the reflections to minimize the difficulty. With this arrangement the distance to the low-frequency driver center from the microphone will necessarily be about 4″ greater but probably have negligible effect because this distance corresponds to the half wavelength frequency well above the low-frequency channel.

I used an electret microphone. Although not calibrated, I regarded the microphone as reliable in probing for response irregularities in the crossover frequency regions. Even so, its response is likely relatively flat from low audio frequencies to 10kHz or higher as observed in catalog data.

I have relied on published data on the drivers themselves as having been taken in proper test environments and having been machine run. I expect the published data (curves) to be an indication of the expected performance of the speaker system itself except where response variations may be caused by the crossover networks and enclosure volume effect around the crossover frequencies.

The measured data is:

Frequency	Relative dB
30Hz	–6.2
60	+2.0
80	–2.1
150	+2.4
300	–0.8
400	–3.8
600	+0.7
1kHz	0.0
2	–4.0
4	–2.4
6	+1.7
8	–1.1
12	–0.2
16	–3.1

The calculated electrical phase at each driver at their crossover frequency is very nearly 45° and the response 0.707 as required. The response dips seen at the crossover frequencies, 400Hz and 4kHz, are very likely due to phase differences in the driver acoustic responses.

Next month, I'll show you how to construct and finish the boxes. ❖

SOURCES

Focal SA, JMlab
42013 Saint-Etiene CEDEX 2
France
011-33-477-43-1616
FAX 011-33-477-37-6587
Midrange driver
(Drivers available from several U.S. dealers)

JBL-Harmon Consumer Group
250 Crossways Park Drive
Woodbury, NY 11797
516-496-3400
FAX 516-496-4868
Low frequency driver

Morel Acoustics USA Inc
414 Harvard Street
Brookline, MA 02446
671-277-6663
FAX 617-277-2415
www.morelusa.com
e-mail: sales@morelusa.com
High frequency driver

Mouser Electronics Inc
958 N Main Street
Mansfield, TX 96063
800-346-6873
FAX 817-483-6899
www.mouser.com
e-mail: www.mouser.com
25W metal-cased resistors

Old Colony Sound Lab
PO Box 876
Peterborough, NH 03458-0876
603-924-6371, 888-924-9465 (US/Canada)
FAX 603-924-9467
www.audioXpress.com
e-mail: custserv@audioXpress.com
BoxModel

Parts Express
725 Pleasant Valley Drive
Springboro, OH 45066-1158
513-743-3000
FAX 513-743-1677
www.partsexpress.com
e-mail: sales@partsexpress.com
terminal assembly, grille guides, and grille cloth

Zalytron Industries Corp.
469 Jericho Turnpike
Mineola, NY 11501
516-747-3515
FAX 516-294-1943
250V polypropylene capacitors, air core coils

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A Switching Power Supply for a Mike Preamp

Set aside your misgivings about switching power supplies and see how you can use them in your tube setup. **By Stuart Rubin**

Some time ago, I began designing a high-end vacuum tube microphone preamp for recording studios. I did not want to use a traditional power supply and incur the size, cost, and heat penalty associated with the various forms of linear supplies. I also wanted an added element of safety and reliability.

I ended up designing a switching power supply for the plates, heaters, and "phantom power." I was quite proud of this design, which turned out to be very inexpensive, small, and versatile. Unfortunately, when I told colleagues and customers that I had taken a modern "switcher" approach, they turned up their noses. I had apparently adulterated a holy tube paradigm.

BACKGROUND

It turns out that people's apprehension about switching power supplies in high-end audio applications is based on their use in low-end, low-quality designs. These supplies generally produce low voltages (well below the typical plate voltages needed for tube designs), and are poorly regulated (if regulated at all). Needless to say, I was pleasantly surprised to see Eric Barbour's article, "Switching Power Supplies for Tube Preamps," in the April 2001 issue of *audioXpress*. Mr. Barbour does an excellent job demonstrating several ways to use "switchers" in tube preamp applications.

ABOUT THE AUTHOR

Stuart Rubin holds a BS in Electrical Engineering from Drexel University, Philadelphia, Pa. He is currently a hardware engineer for Digital 5, Inc., a supplier of DSP solutions for portable audio devices. He is also the founder of Zocat Audio, which builds custom high-end audio equipment for recording studios.

I admit that I felt somewhat vindicated that someone else has been thinking outside of the audiophile box and proudly bucked the use of linear supplies. I was prompted to publicize my particular application and expound upon his article. I will not go into the details of how switching power supplies work, but refer you to Mr. Barbour's fine work.

First, I would like to attempt to explain why people believe that switchers are bad for audio, and explain why this is simply not so.

1. At the heart of listeners' issues about switchers is the belief that they are noisy. Actually, they do not produce "noise" (which is random) in the strictest sense; they do produce ripple in the output that is harmonically related to the switching frequency. This is no different than a bridge-rectified linear supply generating ripple in harmonics of 60 or 120Hz (depending on the use of a half- or full-wave rectifier).

As any beginning audio designer knows, eliminating that "hum" can be

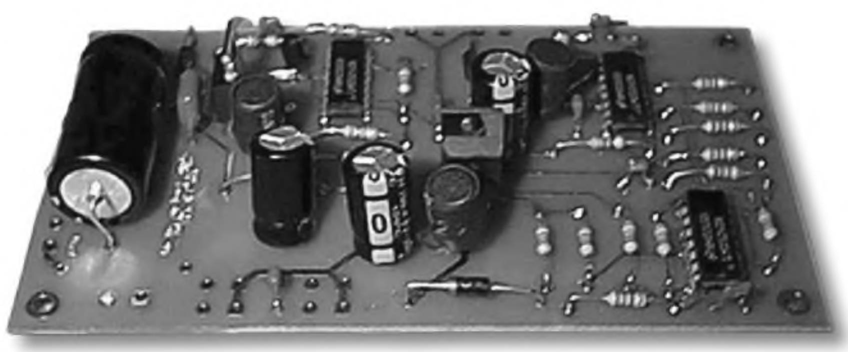

PHOTO 1: Power-supply board.

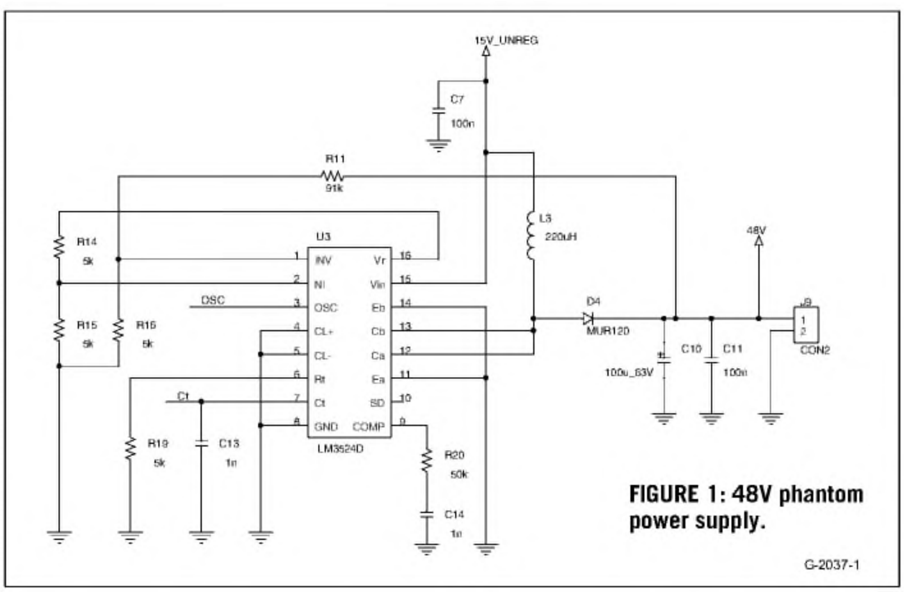

FIGURE 1: 48V phantom power supply.

extremely difficult, even with the use of many large capacitors. Not only is the 120Hz clearly audible, but also so are its second through approximately 166th harmonics. This ripple, including all of its harmonics, "beat" against the audio content and blur the detail of the music.

In my particular application, the switching frequency is approximately 300kHz, many octaves above the audio spectrum. Not only is the switching frequency "noise" out of the audio band, but also these high frequencies are much easier to filter with reasonably sized capacitors.

2. A second issue falsely associated with switchers is that they are poorly regulated. I agree that there are many poorly or unregulated designs out there, but it doesn't take much to produce one with regulation that surpasses a linear design. The regulation depends extremely little on temperature, load, or line voltage, but more on resistors that can be chosen to precision of fractions of a percent and a precision internal reference. There is only approximately ±2% variation on the internal reference of the LM2524D IC used in this design, regardless of the load or line.

3. Many people are apprehensive about switching power supplies because of their complexity. I hope that Mr. Barbour's article and mine will show designers how easy they can be.

DESIGN

The design goals of my power supply are relatively straightforward.

1. Bias four 12AX7 tubes at 150V, for a total of approximately 2mA.

2. Run the heater for each tube at 12.6V, 300mA, for a total of 1.2A.

3. Generate a 48V "phantom power" supply that is used by many micro-

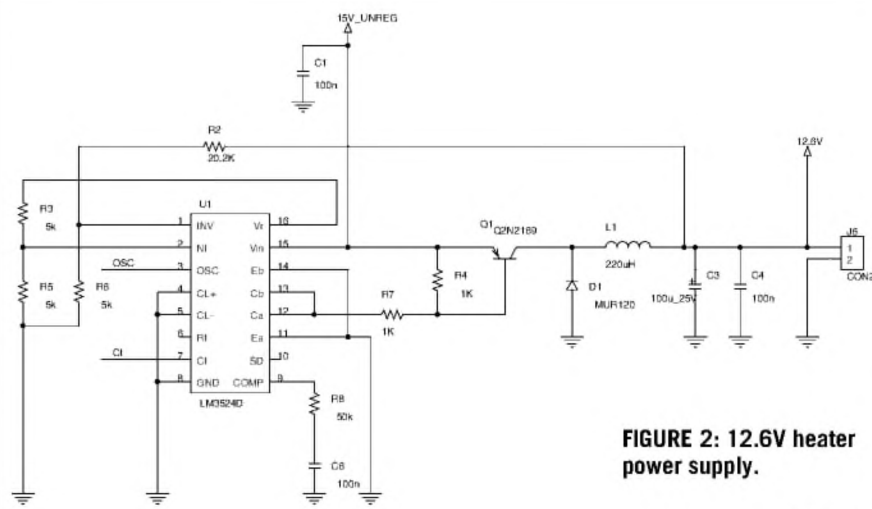

FIGURE 2: 12.6V heater power supply.

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phones for biasing condenser elements, running small internal electronics, and so forth.

4. Power the entire system from a single low-cost, low-power "wall-wart" supply.

5. Use inexpensive, highly available components.

6. Maintain a small physical size.

One problem I worried about was the possibility of the three individual switching supplies "beating" against each other. For example, if one supply ran at 300kHz, and another at 310kHz, there could be audible ripple in the system at 10kHz. This issue steered me to the selection of National Semiconductor's LM3524D Regulating Pulse Width Modulator (see sidebar). The LM3524D has the ability to slave the oscillators of several ICs to a single one so the switching of each is completely synchronized.

This chip turned out to be one of the most flexible such devices on the market, giving you access to many of its internal components. I highly recommend reading the data sheet for this device (available from National's website: www.nsc.com) for anyone building this or another "homebrewed" switching power supply. Several other manufacturers including Maxim and Linear Technology make excellent switching power supply ICs and have outstanding general information on their designs available on the web.

My design (*Photo 1*) features three separate architectures, all using the LM3524D chip in different configurations, a 48V step-up (which basically came straight out of the application note), a 150V step-up using external transistors, and a 12.6V step-down using an external transistor. From a superficial read of the data sheet, it would seem that I am using the LM3524D out of its specified operating range—60V at 200mA.

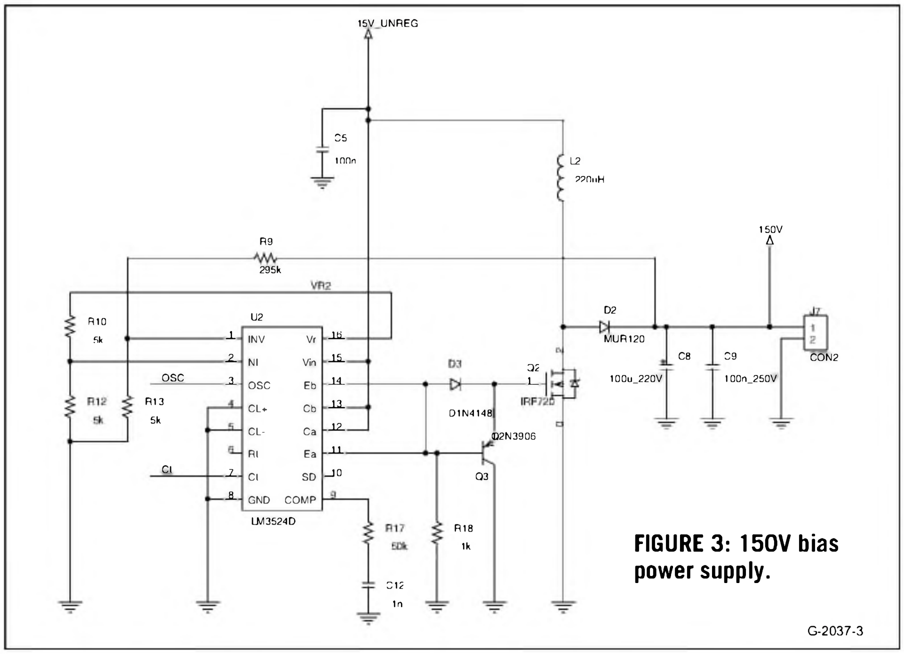

FIGURE 3: 150V bias power supply.

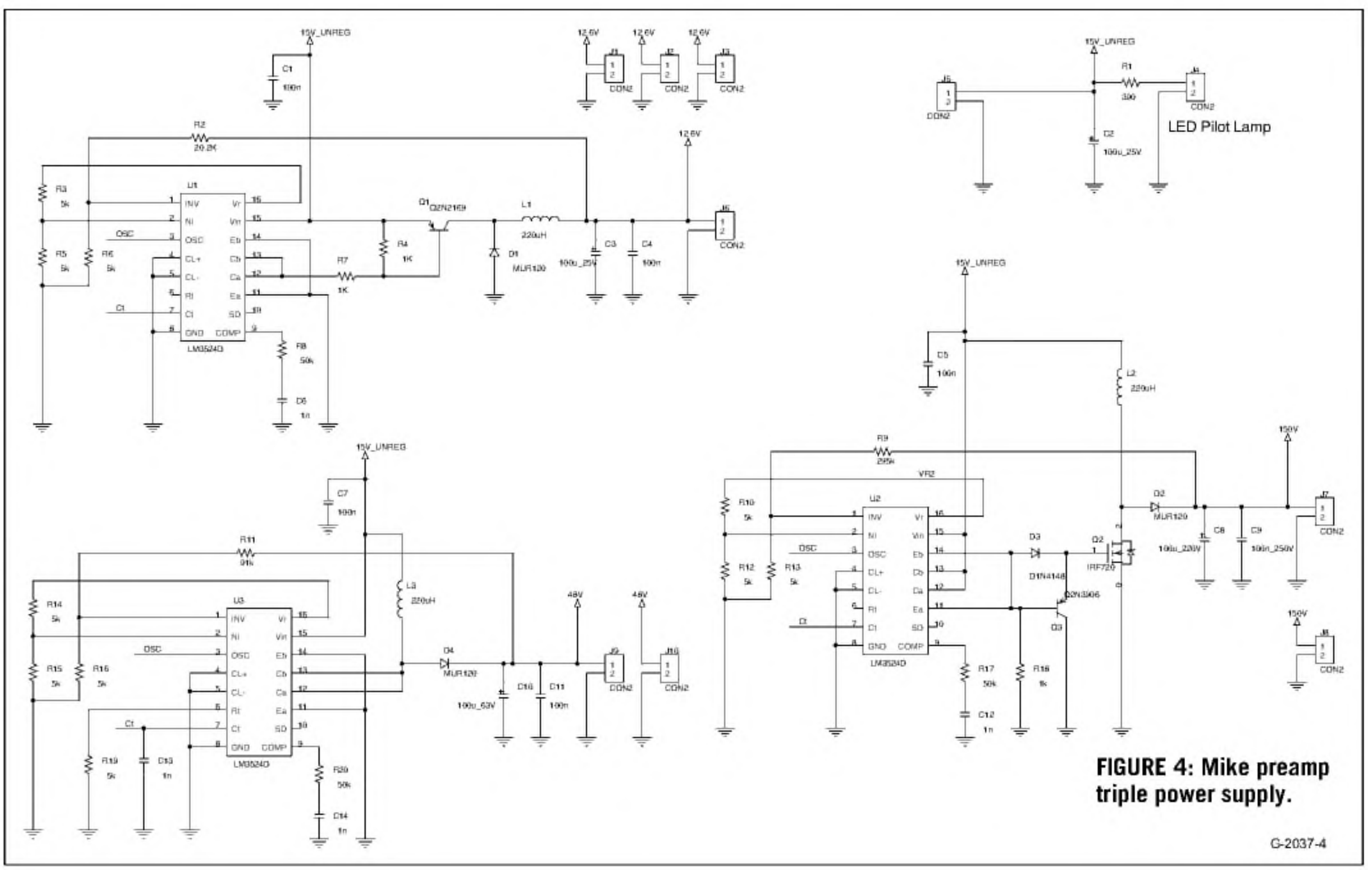

FIGURE 4: Mike preamp triple power supply.

NATIONAL SEMICONDUCTOR LM3524D

The LM3524D is a very versatile integrated circuit for use in switching power supplies, pulse width modulators (PWM), and other applications. This is somewhat of a legacy part and has been manufactured by several companies throughout its history, but it is still widely available from National and perhaps others. There are five main blocks contained within the 16-pin package:

1. Oscillator. Connecting a resistor and capacitor to pins-6 and 7, respectively, sets the frequency of the oscillator. This determines the switching frequency for the PWM.
2. Pulse Width Modulator. Two flip-flops take the clock generated from the oscillator and the output of the error amplifier to determine the duty cycle (ratio of on time to off time) of the output switch. The duty cycle can vary from approximately 49% maximum to 0% minimum (never on). The duty cycle determines the amount of power transferred in a particular cycle of the oscillator.
3. Reference. A precision 5V reference is generated internally by the IC. The supply voltage must be greater than 8V to ensure proper regulation. The regulator can be overridden if less than 8V is used.
4. Error amplifier. The error amplifier takes input from the reference voltage, which is typically externally divided, and a divided version of the produced output voltage. The error is turned into a logic level to determine whether the output is too high or too low. This logic level feeds the input to the PWM.
5. Output transistors. Two NPN transistors are switched on and off by the PWM. The collectors and emitters are left "uncommitted," allowing the designer to connect them to an external inductor, ground, power, other transistors, or any application specific device. The collector-emitter breakdown voltage is specified as 40V, and emitter output voltage 18V. Each can deliver 200mA. Any use beyond these limits will require external switching devices. Two transistors, switched at 180° out-of-phase, are provided rather than one for maximum flexibility and improved switching characteristics.

There is also a current-limiting apparatus in the IC. Current flowing through an external resistor generates a voltage that measures the output current. If a threshold is passed, the current-limiting section turns off the PWM.

Using external transistors controlled by the chip allows for almost arbitrary voltage and current if used carefully.

"PHANTOM POWER"

Many microphones need a low-current 48V DC power supply that is superimposed on the audio going back into the amp. There could theoretically be several milliamps drawn on this supply, but it is typically in the microamp region. Either way, 48V at less than 200mA is easily attainable with no special tricks with the LM3524D.

Figure 1 shows the first and most basic of the three supplies. You program the output of the supply by feeding back a resistor-divided ratio of the output voltage into the LM34524D's error amplifier. Following the equation from the data sheet, R_f = 5K(Vo/2.5–1), setting R11 to 91kΩ yields the desired 48V output when R14, R15, and R16 all equal 5kΩ.

This same equation is used for all three supplies, regardless of the external components to the switcher IC. For this particular supply section, 220µH is probably overkill since such little power is actually transferred from the supply to the microphone. Since 220µH is used in the other supplies, I chose part duplication to be more important than efficiency.

All inductors used are high-current devices capable of delivering considerable power before saturating. The saturation current can dramatically lower the efficiency and deliverable power if it is too low. All diodes are MUR120, high voltage, high current, and ultra-fast recovery (25nS). You can substitute other diodes as long as their forward voltage, peak-inverse voltage (PIV), and current specification are not violated.

Recovery time of the diodes is also a key parameter. In some cases, using a Schottky diode, ultra-fast recovery, general purpose, or some other variation may result in better or worse performance.

HEATER POWER

The four heaters of the 12AX7 triodes used in the amplifier are run from a 12.6V supply. *Figure 2* shows the schematic for the heater supply. Since the circuit is powered from a 15V "wall-wart" supply, the input to the regulator is stepped down to achieve 12.6V. Notice that D4 is connected to ground and "in front of" the inductor. This differs from the step-up configuration, where the diode is wired to the output and "behind" the inductor.

Each heater requires 150mA, for a total of 600mA. This is much higher than the 200mA rating of the LM3524D. To get around this limitation, Q1, a high-current PNP, is inserted as a pass element. Now, the IC must provide only enough current to switch on the transistor, not the actual current needed by the output. R4 and R7 bias the transistor for simple full-on and full-off switching.

R2 is set to 20.2kΩ to satisfy the equation in determining the output. You could change this to produce a 6.3V or any other supply less than 15V needed without any other modifications to the circuit.

In my application, I also use the 12.6 supply to actuate relays used in the amplifier circuit. Compared to the 600mA consumed by the heaters, the relay magnets add negligible load to the supply.

BIAS VOLTAGE

The 150V bias supply (*Fig. 3*) is the most complicated of the three. Although the current is low, 150V is clearly higher than the LM3524D can withstand. Again, I used external components to protect the internal transistors of the IC.

Rather than use a bipolar device, I used an IRF720 N-channel power MOSFET for maximum efficiency. It is important that the MOSFET be turned completely on and completely off so that power is not lost in the transistor itself. In this case, that could mean the difference between using and not using heatsinks on the device. As in CMOS devices, the FET consumes the most power in its transient state between on and off.

To assist the LM3524D in turning the FET on hard (quickly and fully on and off), I employed a 2N3906 transistor, which is connected to the emitter outputs of the IC, not the collector as in the other cases. This is because the use of the 2N3906 inverts the polarity of the switch. Connecting it to the emitters again inverts the switch so that the polarity going to the MOSFET is

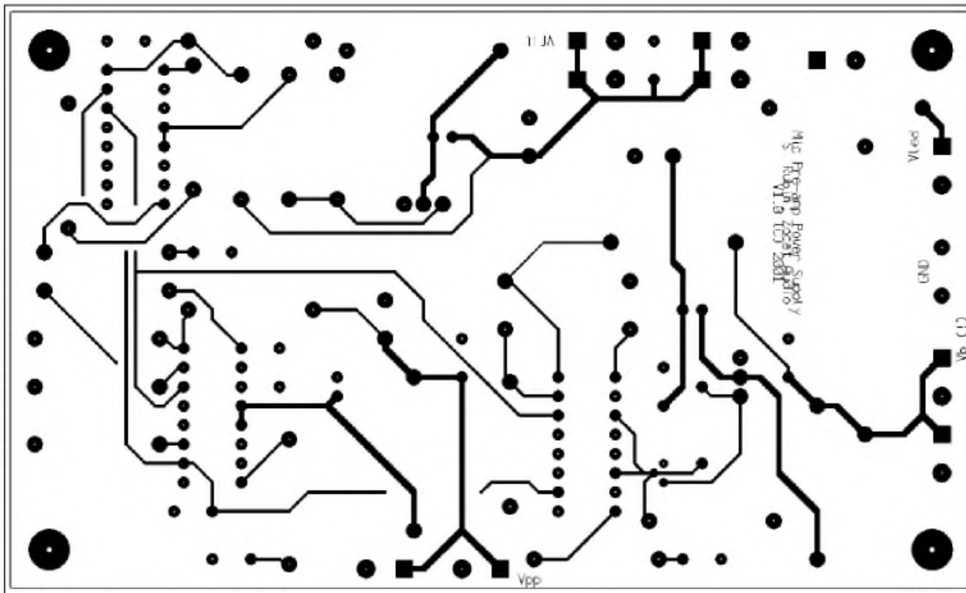

FIGURE 5: Power-supply PCB. G-2037-5

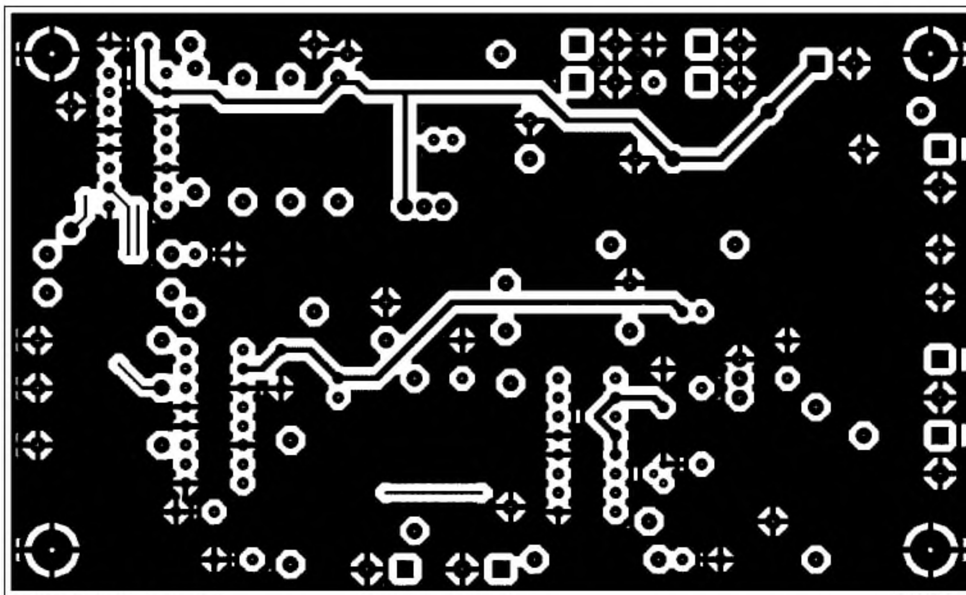

FIGURE 6: Solder side. G-2037-6

correct. D3 protects additional current from flowing back into the emitters of the IC during the off stage of the switching cycle.

Again, R9 determines the output voltage of 150V. Changing this value could produce any reasonable bias voltage. The IRF720 has a voltage rating of 400V, so you could experiment with voltages up to that point.

There are many suitable power MOSFETs that you could substitute for higher or lower rating, cost, or other reasons. Keep the voltage rating of the filter capacitors in mind if you make any changes, and always leave some margin of error and safety.

OSCILLATOR SYNCHING

To prevent the adverse effect of three oscillators free-running with respect to each other, all share a common clock source. U3 serves as the master. Slaving is easily accomplished by tying all pin-3s (oscillator outputs) and pin-7s (timing capacitors) together, respectively, and leaving unconnected pin-6s (timing resistors) of the slaved ICs. R19 and C13 set the switching frequency according to the chart in Fig. 2 of the LM3524 data sheet. (In practice, I discovered considerable variation in the oscillator frequency, which should have been 200kHz according to this chart, but there is no harm in this deviation.)

PUTTING IT ALL TOGETHER

Figure 4 shows the entire design including all three supplies, connectors, and so forth. The 15V wall-wart is connected to J5, where it is filtered by C2. An additional connector, J4, is provided for direct connection to a power-on LED indicator.

Most of the connectors are doubled so that you can use one power-supply board for two channels. You can build a prototype on a blank PCB or etch your own from the artwork provided in *Figs. 5–7*. If there is enough demand, Old Colony may be willing to sell professionally built, unpopulated PCBs for hobbyists. *Photos 2* and *3* show the completed unit.

PERFORMANCE

I took all of my performance data on the 150V supply, which is the most demanding and difficult of the three. The load was a variable resistor, which allowed me to easily change the current while measuring the output level and noise. I varied the operating current between 0.5–6.0mA, which I considered to be a reasonable operating range that worked with my available test equipment. Also, I did not use any external local bypassing capacitors across the load.

For all currents, the load regulation was nearly perfect. That is, the output voltage had no measurable change whatsoever across the entire 0.5–6.0mA range.

The line regulation also turned out to be better than my equipment could measure. To measure the line regulation, I replaced the 15V wall-wart supply with a variable bench-top power supply. Keeping a steady 2mA load, I varied the power supply.

There was no drop in the output until the input voltage reached 9V. There was also no increase in the output when the input voltage was raised to 20V. I did not try to measure beyond a

20V input because I was concerned about power dissipation in the 12.6V step-down section of the circuit.

Noise was my highest concern in the application of this power supply. Again, I use the term "noise" loosely to describe all AC content in the output, which is composed of switching noise, 60Hz hum, and true random noise. There was no measurable 60Hz noise in the output, no matter how hard I tried to find it. The wall-wart provides an additional layer of 60Hz rejection, because it is a switching power supply itself, not a linear regulator.

As expected, the dominant noise source was the 200kHz switching noise. These pulses in current were modulated by an approximately 4MHz "ring," which is probably the resonant frequency of the inductor, filter capacitors, and so forth. The peak noise was approximately 1V (–43.5dB with respect to 150V), but in very quick pulses. In the amplifier application, this noise is easily suppressed well beyond any audible level with small filter capacitors.

One interesting problem continually crept into the mix throughout the design and application of this power supply in various stages. Under the right conditions, the 150V supply can "motorboat." Motorboating is a term used for low-frequency oscillation and can be caused by loop instability of the switching supply, the controller IC turning off and on under low-current conditions, or many other undesirable (and often unknown) circumstances.

In my design, the motorboating is caused by diodes whose recovery time is too slow. When the diode goes from conducting (forward-biased) to not conducting (reverse-biased), there is a short period when current flows in reverse through the diode. The

TABLE 1
PARTS LIST

QUANTITY	REFERENCE	VALUE	DESCRIPTION	MANUFACTURER
LESS COMMON PARTS				
3	D1, D2, D4	MUR120	Ultra fast rectifier diode	General Semiconductor
1	Q2	IRF720	Power MOSFET	International Rectifier
3	U1, U2, U3	LM3524D	Regulating Pulse Width Modulator	National Semiconductor
3	L1, L2, L3	220μH	High-current radial	Digikey PN M6023-ND
COMMON PARTS				
5	C1, C4, C5, C7, C11	100n	Ceramic	
2	C2, C3	100μ	25V, Electrolytic	
4	C6, C12, C13, C14	1n	Ceramic	
1	C8	100μ	220V, Electrolytic	
1	C9	100n	250V, Electrolytic	
1	C10	100μ	63V, Electrolytic	
1	D3	D1N4148	General-purpose diode	
1	Q1	Q2N2169	High-current PNP	
1	Q3	Q2N3906	General-purpose PNP	
1	R1	390		
1	R2	20.2k		
10	R3, R5, R6, R10, R12, R13, R14, R15, R16, R19	5k		
3	R4, R7, R18	1k		
3	R8, R17, R20	50k		
1	R9	295k		
1	R11	91k		

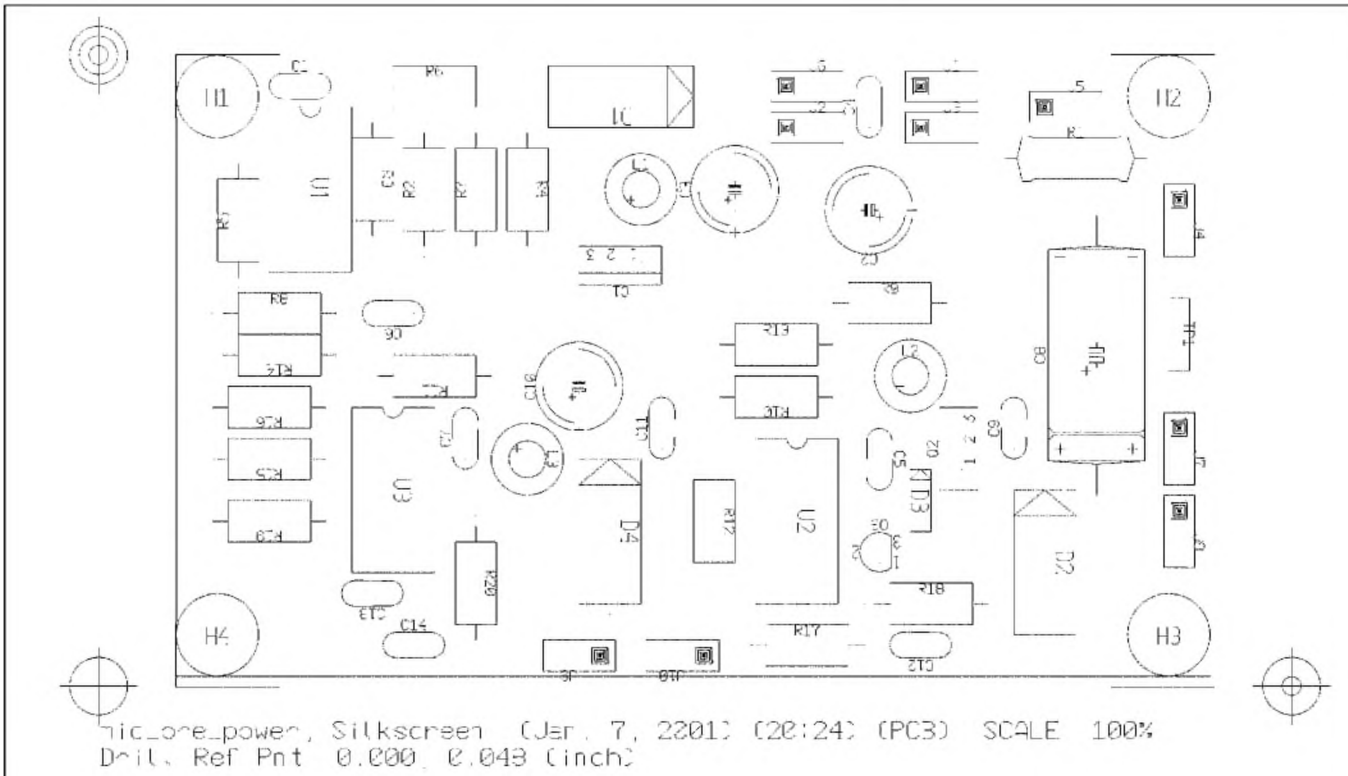

FIGURE 7: Component layout.

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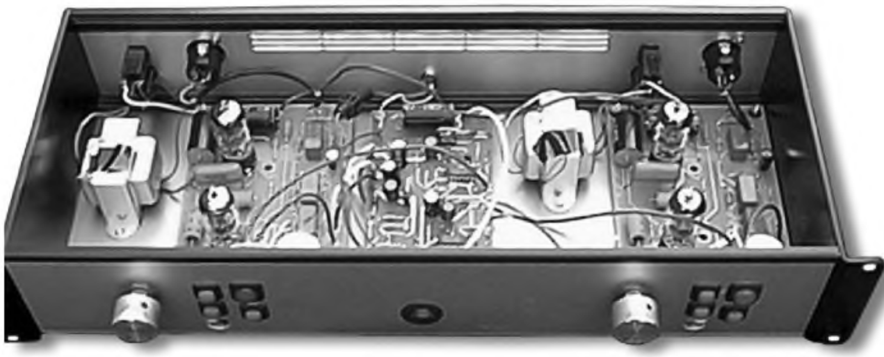


PHOTO 2: Vacuum tube mike preamp.

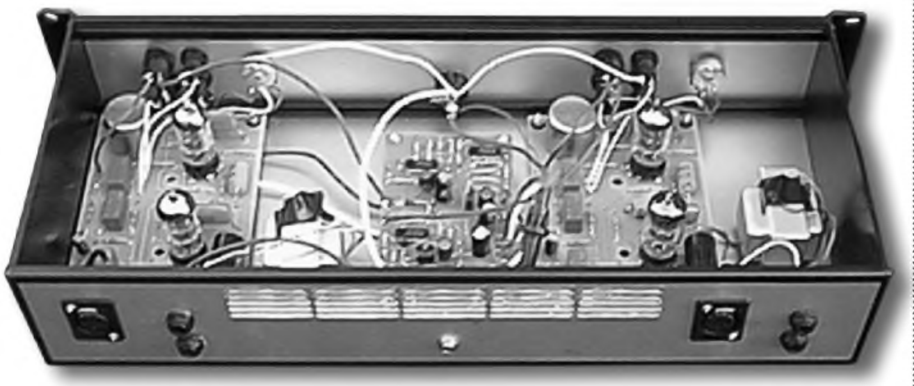


PHOTO 3: Rear view.

MUR120 diode has a recovery time rated at 25nS. This means that for 25nS for each switching cycle of 5µs (1/200kHz), current flows backwards through the diode and into the switching circuitry.

Under the right load and line conditions, this causes the switcher to turn off and on abruptly at an approximately 10–20Hz rate. This manifests itself as audible clicking in the inductor.

In my original prototype, I used an ESD1 diode, which has a much faster recovery time, and I experienced no motorboat problems. The motorboating is also usually stopped by raising the input voltage or lowering the load. If you build this supply and experience this problem, I strongly recommend using a faster diode, even if it means going to a surface-mount device such as the ESD1.

SAFETY

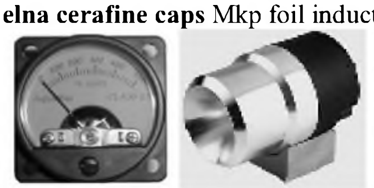
As with any project, safety should be the highest priority for any design and construction. Not shown in this design is a 1A, slow-blow fuse placed in series with the DC wall-wart supply. The wall-wart itself is UL approved and current-limited for 1A, adding a second level of protection from the AC line voltage.

Even when built correctly, lethal voltages are generated and should be handled with the utmost caution. The author of this article is not liable in any way for damages from this circuit or any modification thereof.

CONCLUSION

I met all of my design goals with little difficulty. I was surprised at the relative ease in bringing this circuit to life. It has proven to be reliable and flexible, and I plan to use the exact PCB, changing only component values, for future amp designs. The entire project fits neatly on a small 3″ × 5″ circuit board using all through-hole, standard parts. At a total cost of about $30, it is worthy of experimentation.

I hope that other people will give switching power supplies their chance in high-end audio applications—tube or solid state. Their benefits are rooted in technology and proven by application. Just because there is an IC in your tube amp doesn't make it any lower fidelity. ❖



The DR8 Horn

This author's latest speaker project features an improved design for better horn efficiency in a smaller cabinet. **By Bill Fitzmaurice**

PHOTO 1: DR8 horn, oblique view.

This past winter I went to a ski-industry party at which a local Vermont-based band played. The hotel function room where they set up was about 25′ wide, and perhaps 100′ deep; the attending crowd numbered around 200. To drive this room with PA, this five-piece band had a 32-channel board, two racks full of processors, and a six-cabinet speaker system totaling over $72ft^3$ in volume. Including power amps and monitor cabs, I figured the system must have cost at least $30,000.

For a 200-seat room it was overkill by a factor of ten. This monster system needed three roadies and a high-cube van to haul it in. Talking to the band, I found out that their paycheck for the night was $1,000, fully one-half of which they spent just on overhead. The price paid to feed one's ego can be astounding.

When I told this band that only two of my DR12 horns (*audioXpress*, Jan. '01) occupying less than $30ft^3$ would blow away their huge system, and do it with one-eighth the wattage input, they couldn't believe it. They would have been even more flabbergasted if they had seen the cabinet I would use for a 200-seat room: the DR8 (*Photo 1*). Smaller than most commercial stage monitors, an average SPL/2.83V of 105dB (±4) makes a pair of these suitable for PA chores in small to medium venues, where bass and kick drum reinforcement are not required.

Since low-frequency power, and thus a large cabinet, is not required, this very high SPL within the 120Hz to 16kHz passband can be achieved from a very small box. And while it'll handle 200W, for 100 seats or less a 100W/channel power amp is sufficient for room-filling sound.

HOW IT WORKS

Using the folding horn geometry introduced in the DR12, the DR8's rounded horn bend and curved horn panels virtually eliminate phase cancellation, extending bandwidth two octaves beyond previously predicted limits. The midrange performance is further enhanced by the dual flare DR (double reverse) horn arrangement. Single flare bass horns, which taper at the same exponential rate throughout their length, fail to load high frequencies.

The DR8 driver fires into a short, rapid flare "throat horn," which could serve as a stand-alone midrange horn. This horn properly loads midrange frequencies before it in turn feeds the dual slow-flaring mouth horns, which then load the lows. With this arrangement horn loading takes place across a full five-octave range, instead of perhaps three octaves as seen in older designs.

For improved bass, the rear wave of the driver is not enclosed in a closed box, a practice that causes loss of response below the horn passband. Instead, the rear chamber is vented by a duct, which exits at the horn mouth, giving usable bass response two full octaves below the horn-flare frequency. The sum result of these innovations is a new class of speaker—one with the bandwidth of a direct radiator and the efficiency of a horn—in a cabinet no larger than a traditional T/S box.

If you've built my MiniSnail (*SB* 8/98), you already know that loud sounds can come from small packages. Close in size to the MiniSnail, the DR8 is even louder. *Figure 1* shows a comparison between the two cabinets. The improved design of the DR8 gives wider bandwidth and flatter response (though its two piezo elements give a bit less high end than the four of the MiniSnail). The good news for MiniSnail owners who choose to upgrade is that the woofer is the same

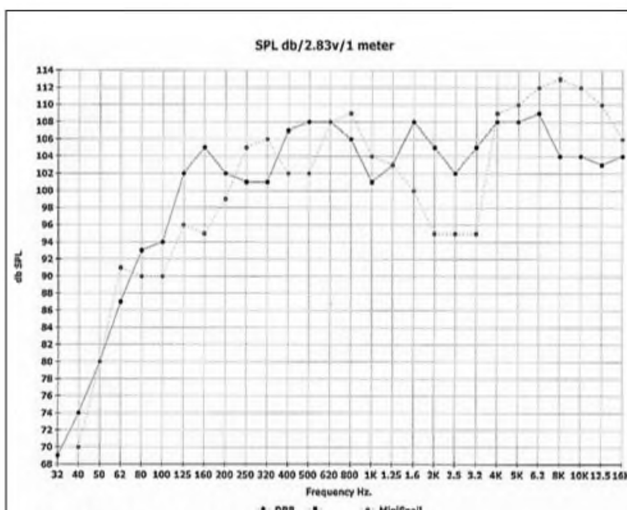


FIGURE 1: MiniSnail vs. DR8.

PHOTO 2: The throat-horn divider clamped to throat-horn side.

model, so you can swap that and the other hardware out of your old cabs.

DRIVERS

That woofer is a Carvin™ PS-8, an OEM from Eminence™ and a real bargain at about $50 (www.carvin.com). I've found more than one set of specs for this driver. The sheet that accompanied the ones I bought claimed an Fs of 80Hz, but my specimens measure out at 64 and 66. Frankly, I would have preferred 80Hz.

For maximum power in the desired passband, the horn Fs(h) [the driver Fs after mating it to the horn] should be between 60 and 70Hz. An 80Hz Fs would have allowed an Fs(h) within that range, and would have done so with a smaller throat than the 24 in^2 I ended up using, giving even better highs. As it was, the Fs(h) of my two prototypes are 53 and 55Hz, respectively. That enhances performance below 80Hz, where you don't want it, and probably costs 3dB at 100Hz, where you do want it (this can be seen as a slight flattening on the SPL chart at 100Hz). A larger throat would have raised Fs(h) to regain that 3dB, but at the expense of high-frequency performance.

The final size was the best compromise for the drivers being used. While this doesn't ruin the cabinet by any means, it shows the importance of not using a low Fs driver in a horn-loaded cabinet. I wouldn't recommend using

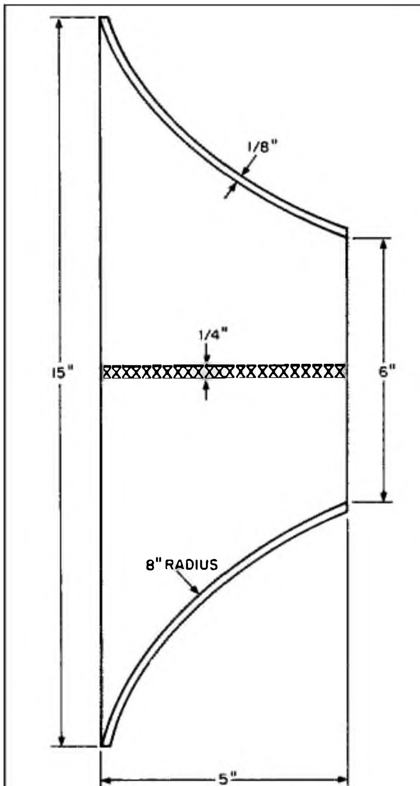


FIGURE 2: Throat-horn side view. Note throat-horn divider is offset slightly from center.

PHOTO 3: Clamping cauls for the throat horn.

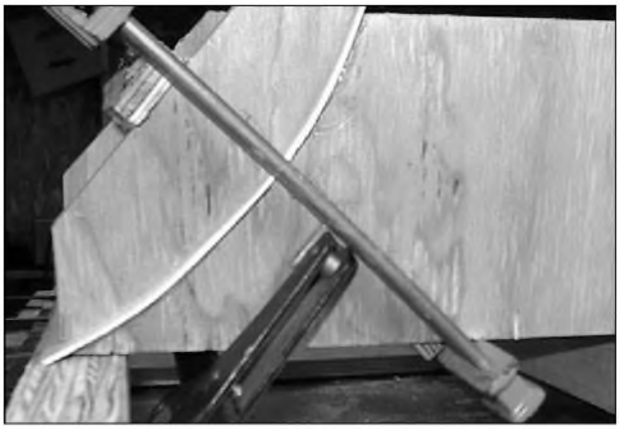


PHOTO 4: Assembling the throat horn with clamping cauls.

PHOTO 5: Trimming the throat horn on a cutting jig.

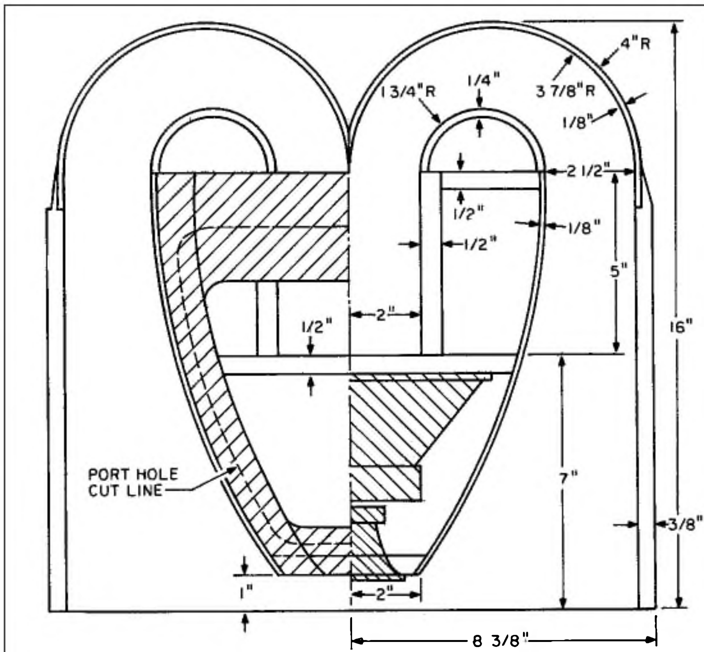


FIGURE 3: DR8 top/bottom view. Half of view shows drivers and most internal parts, half of view shows porthole cut-line and porthole flanges/horn braces. Not shown: Side-fire tweeters.

any driver in this box that measures with an Fs of less than 60Hz or more than 90Hz. As for other specs, my experience with horns has shown them to be noncritical, as long as you use a driver intended for high-power MI use.

The tweeters, alas, are not the same as those in the MiniSnail, which are too deep to fit inside this box. I used a pair of Motorola™ KSN-1167A piezos for the front array. You may substitute another model as long as it is no more than 3″ deep. The baffle is large enough so that you may use three KSN-1167As for a bit more high frequency "oomph."

I also used some "knock-off" piezos from Parts Express for side-fire tweeters. They seem to work well enough, and at only a buck apiece they are a steal. Time will tell whether they will hold up to the rigors of the road.

CONSTRUCTION

If you're familiar with my work, you know that my preferred building material is ½″ plywood for flat panels and ¼″ plywood for bent panels. I recently read an article in *SB* by one of the guys on the hi-fi side of the fence and was amused by his choice of materials, two layers of ¾″ MDF. I know that hi-fi cabs must be inert, and that where portability is not an issue sheer weight is the easiest path to sonic nirvana. Nonetheless, I'm not sure that making speaker cabinets able to withstand the same stresses as an Abrams tank using roughly the same construction is the best way to go. [An upcoming article will prove otherwise.–Ed.]

My construction methods are more akin to an aircraft's. Where strength is necessary, but excess weight intolerable, cross-bracing is far more effective than high density in eliminating spurious vibrations from cabinetry. Still, with a box of only 2ft^3, couldn't I deviate from my usual choice of ½″ ply? After all, the MiniSnail weighed in at only 27 lbs.

I thought about it long and hard before deciding that ½″ and ¼″ plywood were not the best choices for the DR8. Because I wanted this box to be even lighter than the MiniSnail, they were too thick. Thus, many of the DR8 parts are ⅛″ and ⅜″ plywood. If you wish, you may use thicker materials than those I recommend, but your DR8 won't sound any better, it will just be heavier.

Secure most joints with drywall screws and construction adhesive, preferably the urethane type. Use 1″ screws when attaching up to ⅜″ parts, 1¼″ when attaching ½″ parts. If the receiving part is less than ½″ plywood, however, screws are out for the most part, though you may be able to get some into ⅜″ ply if you are very careful. In these cases, use hot-melt glue, which cannot be used where high strength is required, but works fine where fast setting and gap filling are required.

When using hot-melt, chamfer well any edge that you intend to attach. This produces a trough on the joint to hold plenty of glue. Remember that every joint must be absolutely airtight, so too much glue is better than not enough.

Also, when cutting parts that are to be bent, always flex the plywood first to determine the more flexible axis. I recommend Baltic birch 1/8″ plywood, which bends easily but is very stiff after installed. It also tends to fray less than other woods when cut.

THROAT HORN ASSEMBLY

The first step in construction is to assemble the throat horn (*Fig. 2*). The throat horn sides are 1/8″ plywood. These are best cut at the same time, by rough-cutting two sheets of plywood, screwing them together, and then finish-cutting them to end up with two identical pieces. Save the scraps that are cut from the curve.

Glue the 4″ wide throat divider, which can be made of either ¼″ or 1/8″ plywood, in place between the two sides using hot-melt. Chamfer all the edges of the divider with a sander. You should slightly offset the divider from the exact center of the throat horn, so that the two resulting horn halves are not quite identical, and out-of-phase reflections in the two halves will occur at different frequencies. Accurate placement of the divider while gluing is easy using picture frame clamps to hold the divider in place while the glue sets (*Photo 2*).

Use the scraps cut from the curves of the throat horn sides to construct two clamping cauls (*Photo 3*). Cut the sheaths from 1/8″ plywood, making them at least an inch longer than necessary

(to page 46)

PHOTO 6: The throat horn attached to the horn supports.

PHOTO 7: Attaching the throat horn to the bottom with guideboard and clamps.

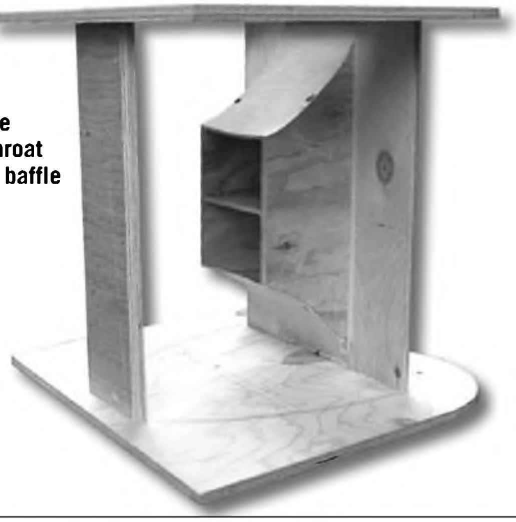


PHOTO 8: The top/bottom/throat horn/tweeter baffle assembly.

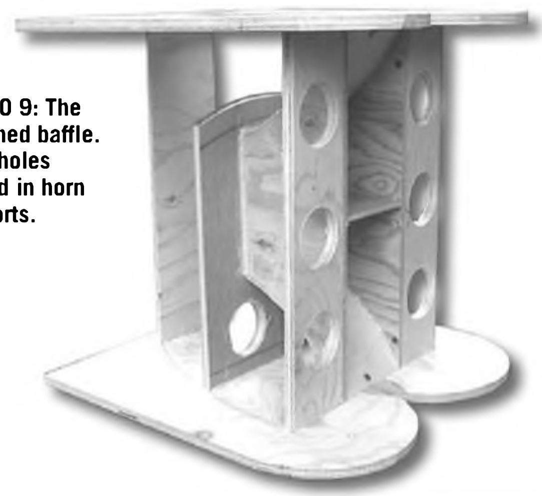


PHOTO 9: The attached baffle. Note holes drilled in horn supports.

(from page 45)

for trimming to size later. Put a good chamfer on the curved edges of the throat sides and clamp a piece of sheathing in place, using the caul to perfectly mate the sheathing to the sides (*Photo 4*). When you have made sure that the assembly is perfectly square and aligned, fill the troughs between the sides and the sheaths with hot-melt glue, inside and out. Don't glue the cauls to the assembly!

After the hot-melt has thoroughly set, repeat the procedure on the other sheath. For good measure, drive one screw apiece through the sheathing into each side to give the unit stability until you mount the throat horn into the cabinet. Trim the excess sheathing from the horn mouth by running the assembly across the table saw atop a panel-cutting jig (*Photo 5*). Sand the sheathing at the throat flush with a belt sander.

The throat horn supports are ½″ plywood. Cut these a bit long, overhanging each end of the throat horn when you attach them; you'll need either a stubby screwdriver or right-angle ratchet driver for the screws in the tight confines of the horn. Trim the excess length of the supports by again putting the entire horn assembly through the table saw atop a panel-cutting jig, which will keep everything square (*Photo 6*).

Cut the top and bottom from ½″ plywood (*Fig. 3*). Once again, it's best to rough-cut these pieces, screw them together, and then finish-cut them so that the two parts are identical. You may also want to sandwich a piece of plywood (of the same thickness as the throat horn divider) between the top and bottom to cut the back braces at the same time.

Draw on the top and bottom the locations of all mating parts, as well as the porthole in the bottom piece. Using a plunge cut to start, cut the porthole from the bottom with a saber saw. Attach the throat-horn assembly to the top.

This (and all joints) are best done by first drilling pilot holes through the top and clamping a straight guideboard (mine is made of 1½″ plywood) along the joint line. Trial-fit the horn in place to be sure of its exact alignment. Then glue the joint line and clamp the horn in place; the clamps will pull any flaws in alignment back to true (*Photo 7*). Finally, drill again from the outside through the previously drilled holes with a combination pilot/countersink bit and secure the joint with screws.

BAFFLE CONSTRUCTION

Cut the tweeter baffle from ½″ plywood; each side is cut at a 35° angle. Plan your cutting so that you have two strips of plywood left over, each 15″ long and about 1½″ wide, with a 35° angle on one

PHOTO 10: The assembly ready for sheathing.

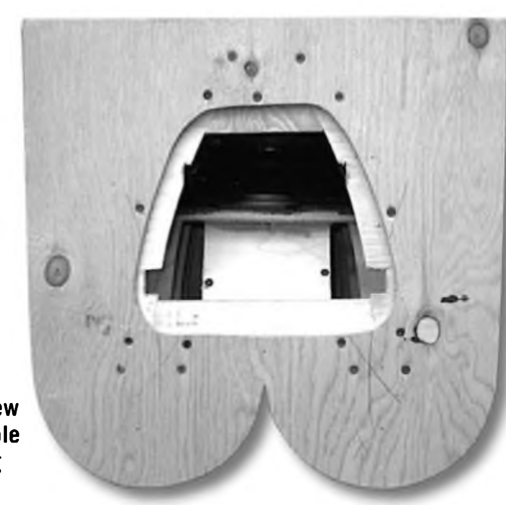


PHOTO 11: View through porthole after attaching the flanges.

edge. Attach the tweeter baffle to the top, and then the bottom to the assembly (*Photo 8*).

The baffle is ½″ plywood, with its side edges cut at a 15° angle. It intersects the top but does not extend to the bottom; cut it only as long as need be to fit the driver. Trial-fit it into the cabinet, tracing the location of the hole through the throat horn. Cut out the hole and round over the edges facing the driver.

Sound waves hate flat surfaces, so use a hole saw to drill a couple of 2⅛″ holes (the same size you'll need later for the tweeters and duct) through the free area on the baffle. Drill some holes through the horn supports as well. Install the baffle (*Photo 9*), again clamping everything in place and making sure all is square and true before driving any screws. Use plenty of adhesive where the ⅛″ sheaths intersect the baffle, for an airtight seal.

BRACES

Cut out a total of eight mouth-horn braces from ½″ plywood. The two braces attached to the bottom are installed whole, forming the bulk of the flange for the porthole. Complete the flange with scraps of ½″ ply.

Subdivide the remaining braces to accommodate the baffle. Mount two braces to the cabinet top, and four in the space between the top and bottom. Placement is not critical, but omit the two brace halves spanning the baffle and the tweeter baffle nearest the porthole (*Photo 10*). When installing the braces, temporarily place the driver on the baffle to be sure that the braces do not interfere with the driver or with access to the driver mounting holes.

Also trim the porthole flanging as required to allow the driver to pass through (*Photo 11*). Make absolutely sure now that you'll be able to get the driver into place. Cut the strips of plywood left over from the tweeter baffle to fit the rear of the tweeter baffle in between the braces, forming a 1″ thick panel at the baffle edges. Let the adhesive on the assembly set overnight before proceeding.

The mouth-horn sheath is ⅛″ plywood, which may seem too thin to do the job, but it is not. Once bent into a circular shape, the stiffness of ⅛″ plywood

PHOTO 12: Holding the sheathing in place. The 1/8″ sheathing bends easily, no clamps required.

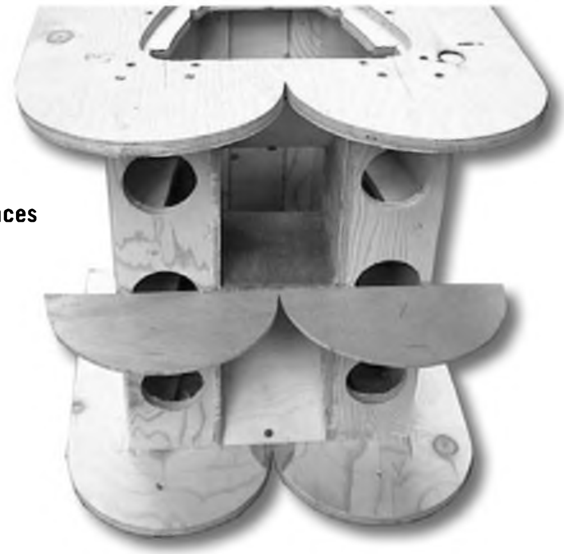


PHOTO 13: The back braces in place.

approaches that of 1/8″ plywood when flat. When further stiffened by both the horn braces and the side braces, the horn sheathing is rock-solid.

Attach it to the assembly starting at the horn supports, leaving about a 1/4″ overhang. Bend the plywood down into place, fastening with screws every 3″ or so (*Photo 12*). Hitting all the braces and the baffle will entail drilling a few "exploratory" holes, so make sure you fill in any misses later. After installing both sheaths, let the adhesive set overnight before sanding the excess flush to the tweeter baffle and the horn supports.

Attach the back braces to the assembly with hot-melt, gluing them to the throat-horn divider and the supports (*Photo 13*). Slice a 15″ piece of 4″ PVC Schedule 40 pipe in half lengthwise, clamping it to a jig and using an abrasive blade (*Photo 14*). Do not slice all the way through, as the PVC will close on the blade; finish the cut with a utility knife or hand saw. Cut four pieces of the pipe to fit the rear of the horn supports in between the back braces and the top and bottom. Fill these pipe sections with poly-fill before setting them in place and gluing and thoroughly sealing them with hot-melt (*Photo 15*).

Cut the side braces from the same thickness plywood you used for the throat-horn divider and back brace. Size them to extend to about an inch from the front of the tweeter baffle. Cut them about an inch too wide, then temporarily clamp them to the back braces with plywood stabilizing cauls. Trace the actual curve of the horn sheath on them, spacing the pencil if necessary with a sliver of plywood (*Photo 16*).

After cutting the curves, temporarily install the side braces again, this time running a straightedge from the cabinet top to bottom to mark the brace edges (*Photo 17*). When perfectly cut, attach them to the assembly, using hot-melt. If you made them from 1/2″ plywood, you may also drive a screw or two into them from inside the cabinet for good measure.

FINAL ASSEMBLY

The back sheath halves are 1/8″ plywood, which you should cut a good inch long and trim when in place. They should extend at least 1″ past the curvature of the back onto the flat of the sides. You should also cut the first half to be installed about 1/2″ wider than the other.

To install, first place the wider half in place and tack it with hot-melt. Slide the smaller half in next to the first and tack it in place, and then slather a good amount of hot melt on the overlapping joint of the two back halves (*Photo 18*). Gradually pull the sheaths into place, driving screws every 3″ or so into the top and bottom.

As you drive each pair of screws, reach inside the horn and hot-melt the halves to the top, bottom, and back brace where the parts have made contact. After you have driven all the screws, there will be a bulge in the

PHOTO 15: The halved PVC in place.

PHOTO 14: Halving PVC pipe on the cutting jig.

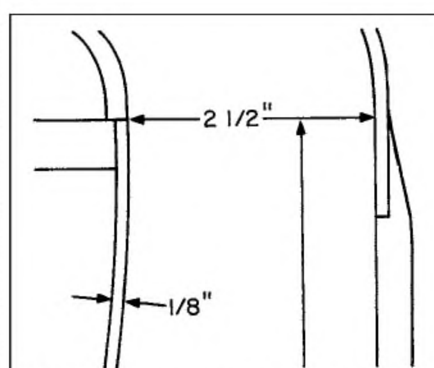


FIGURE 4: Closeup view of side/back joint, showing overlap of side over back sheath.

PHOTO 16: Attaching the side braces.

PHOTO 17: Truing the side braces with a straightedge.

halves where they won't quite make contact with the side braces; use long clamps, guideboards, and plywood cauls to push the halves into place for gluing (*Photo 19*). Also dribble a good amount of glue into the joint of the two halves from the rear, to thoroughly seal the junction.

The cabinet sides are made of ⅝″ plywood. They overlap the back halves by at least an inch, going all the way to where the curvature of the back begins. Using either a table saw or router, cut a rabbet in the sides to accommodate where they lap the back halves (*Fig. 4*). Install the sides using screws and construction adhesive on the top and bottom joints and hot-melt at the braces.

Use plenty of adhesive on the overlapping joint with the back halves. The sides will also bulge away from the braces, so use the same procedure as for the back halves to hold them in place until the glue and adhesive has set. When everything has cured, sand the back and sides flush to the top and bottom.

Drill or cut holes in the tweeter baffle for the tweeters and duct. The duct hole for 2″ PVC will be 2⅜″ d., as will those for the listed tweeters. For other tweeters make sure that they are at the top of the baffle, with the duct at the bottom. If you are using the optional side-fire tweeters, cut holes for them as well. Side-fires will require holes through the sheaths for their wiring, which you must seal tight with hot-melt.

Apply your finish. Install the driver, which you must screw to the baffle, with the screwdriver reaching through the holes in the tweeter baffle. The driver magnet will pull a screw off the screwdriver, so tack it to the driver with hot-melt before inserting the driver into the cabinet. Four screws will be sufficient to hold the relatively lightweight 8″ driver.

Lightly but thoroughly stuff the cabinet with poly-fill, working through the porthole and tweeter holes. Do not put poly-fill where the duct will hit it. Install the jack and speaker-stand top hat on the porthole cover. Make sure not to

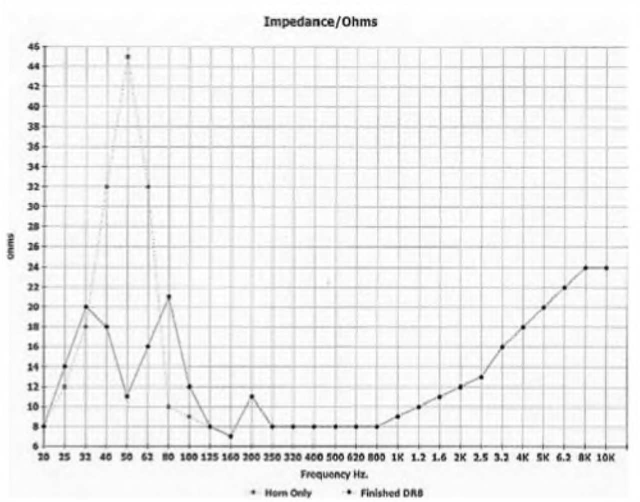


FIGURE 5: DR8 impedance measurements.

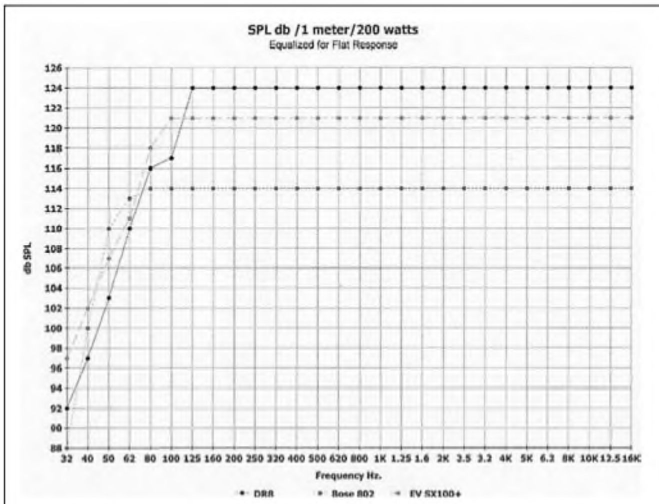


FIGURE 6: DR8 vs. Bose vs. Electro-Voice.

place the top hat where it can hit the driver magnet or baffle.

Wire and install the tweeters, either weather-stripping or caulking them in place. The tweeters are parallel wired, in-phase with the woofer. Wire the jack to the woofer and the tweeters. Rim the porthole flange with dense neoprene weather-stripping and install the porthole cover.

TUNE-UP

The prototype boxes use a 2″ × 2″ duct, which gives an Fb of 55Hz. To tune your cab to a different driver, first run an impedance sweep, driver installed, with the porthole cover off the cabinet and no duct or tweeters in place. Look for an impedance peak in the vicinity of 60Hz, which will be the Fs(h); on *Fig. 5* this is the dotted trace.

Install the tweeters and porthole cover. Sweep the impedance again, trial-fitting ducts of various lengths until the sweep shows an impedance minimum at the same frequency as the Fs(h); on *Fig. 5* this is the solid trace. The box is now tuned properly. Glue or caulk the duct in place.

You now are the owner of the best 2ft^3 PA speaker money can buy. Want proof? Look at *Fig. 6*. Here the top trace is the DR8, driven by 200W, EQ'd for the flattest possible response. (To EQ your DR8 for flat response with a 31 band EQ, set the sliders inversely to *Fig. 1*, using 105dB as the base line. For example: the slider at 160Hz is set at 0, the 125Hz slider set at +3dB, the 400Hz slider set at –2dB, and so forth. This gives a flat response, which must be further tweaked to accommodate room acoustics and feedback control.)

The lowermost plot is a Bose™ 802, also driven with 200W, EQ'd for flat response. The 802, slightly smaller than

TABLE 1
PARTS LIST

ALL SIZES NOMINAL, DEPENDING ON THICKNESS OF MATERIALS USED, SEE TEXT FOR DETAILS.

Throat-horn sides	15″ × 5″
Throat-horn divider	5″ × 4″
Throat-horn sheaths	5″ × 8″
Throat-horn supports	15″ × 3″
Top, bottom	16″ × 16″
Tweeter baffle	15″ × 4¼″
Baffle	9½″ × 12″
Horn braces	10″ × 1½″
Tweeter baffle braces	15″ × 1½″
Mouth-horn sheaths	15″ × 12″
Back braces	4″ × 8″
Side braces	11″ × 6″
Back sheaths	11″ × 16″

the DR8 but heavier at 35 lbs, is advertised as one of the finest small PA speakers available. A pair of these, along with the required System Controller (an active EQ), has a list price of $1,398, though you may be able to find them on sale for only $1,200. Save your money.

PHOTO 18: Beginning attachment of a back sheath.

Although the 802 has the advantage over the DR8 below 75Hz, the DR8 is fully 10dB louder than the Bose™ from 125Hz up, where it counts. A Bose would need 2,000W input to keep up with a DR8.

The middle trace is one of the newer small PAs on the market, the EV™ SX100+, also driven with 200W, EQ'd flat. This molded plastic enclosure uses a 12″ driver and HF horn, and also beats the DR8 below 100Hz. But from 125Hz the advantage again shifts to the DR8, this time by 3dB. The EV is larger than the DR8, weighs 8 lbs more, and has an average street price of $680 per pair.

While that may not be outrageous (a commercial version of the DR8 should come in at about that price), you can build a pair of DR8s as described for less than $250. For that much money you can easily afford to have a pair of DR10s or DR12s for the big gigs, and a pair of DR8s for those small jobs. Or you can blow half your gig checks paying for roadies, transportation, and overpriced commercial gear.

Take my advice. Stop feeding your ego by spending big bucks on overstuffed, underperforming gear and put the savings into an IRA. You'll thank yourself someday. ❖

Bill Fitzmaurice's collected speaker designs, including his work with Snail folded horns, are detailed in his book *Loudspeakers for Musicians* (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, custserv@audioXpress.com, $9.95).

PHOTO 19: Attaching a side.

PHOTO 20: The DR8, front view.

The Sessions Monitor, Part 1

This is the first of a two-part article describing the design of the Sessions Monitor System and the construction of the bass cabinets.

By Steve Philipczak

I had in mind the idea for this project for almost a year before I actually sat down to draw it up and contemplate building it. I had specific design and performance goals to meet, and I just did not want to compromise on any of them. The drivers available to DIYers at the time were not quite up to what I was looking for.

DESIGN THOUGHTS AND GOALS

I wanted to build a full-range, biamped system in a four-box configuration, with both the bass system and the satellite in sealed cabinets. I decided on an MTM—or D'Appolito, if you will—setup for the satellites, because I have had good experiences with them in the past. I also wanted to keep the low-frequency cross point at 100Hz or lower between bass and midrange for a more seamless blend.

"So, what's the problem?" you're probably thinking. I had one more idea that I wanted to stick to that was the real "fly in the ointment." I wanted to keep the satellite's bass/mid drivers less than 6″ in diameter, both for a narrow satellite profile and for the good percussive "attack" that 5″ drivers tend to provide. Although there were numerous 6″ mid-woofers available that would have filled the bill, I could not find a 5″ driver that could be expected to safely extend down into the 50–60Hz range in a sealed cabinet. With a fourth-order active crossover, I would need to attain 50–60Hz as a 3dB down point low extension to practically be able to cross over the bass system to the satellite at 100Hz.

So, the whole idea remained on the back burner until Scan Speak made my goals possible with one of their new "Revelator"-type drivers. This was the driver I was waiting for. I have lots more to say about it in the section on the drivers selected for the Sessions Monitor.

BEAUTY AND FUNCTION

I also had some aesthetic and ergonomic ideas for this system. I wanted to avoid the "smaller box sitting on top of a larger box" look that often goes with a four-box system. I thought that a triangular-shaped bass cabinet could blend into the narrow satellite profile, and also have the benefit of an irregular shape's reduction of internal standing waves. Finishing touches would include a radius on the satellite's front, a stand-off-type grille for the satellite, real wood veneer on all cabinets, and perhaps some wood inlay trim on the front of the bass cabinet.

In terms of ergonomics, I prefer to make my designs easily serviceable, and yet still solid and somewhat pleasing to the eye. This was a bit of a challenge for the Sessions Monitor. I wished to locate the satellite's passive crossover externally for three reasons: 1) the magnetic fields from the driver magnets would not be too close to the coils in the crossover; 2) the crossover could be easily modified or even swapped with a different design; 3) I would not need to concern myself with making a separate chamber for the crossover or making the crossover small enough to fit through the cut-out for a 5″ driver.

PHOTO 1:
Full view of completed system.

Making the bass cabinet very deep proved to be the solution to several problems. I could locate the satellite crossover on top of the bass cabinet behind the satellite. The added depth would provide the required internal volume for most 12″ woofers suitable for a sealed box, and also have the side benefit of giving the back wave of the woofer a little breathing room. All

ABOUT THE AUTHOR
Steve Philipczak is a graduate of Cleveland Institute of Electronics. He has been building speakers for approximately six years and has been a avid high-end audiophile for over 20 years. He is also experienced in luthiery, building and repairing stringed musical instruments. Steve is currently employed by Agere Systems (formerly the Microelectronics Group of Lucent Technologies) as a Process Analyst.

these ideas culminated into the basic shape of the final design as you can see from *Photos 1–5*.

BIAMPING DOES WONDERS

Readers of this publication are probably aware of the numerous and significant advantages of biamping full-range loudspeaker systems. The subject is too extensive to cover here; however, if you wish to know more, you can find some excellent information on Rod Elliott's website, www.sound.au.com/index.html (Bi-amplification—Not Quite Magic, But Close). I'm sure you can find many other articles on the subject as well.

I avoided both the expense and perceived complication of biamping on all of my previous designs. However, with a project of the caliber of the Sessions Monitor, it made a lot of sense. I believe a lot of the excellent performance of this system can be attributed to biamplification.

DRIVER CONSIDERATIONS

At this point, I pretty much had things worked out in my head, but I thought I would make the final decisions on drivers and then decide on final dimensions before drawing the design. Choosing drivers is obviously a very critical step in any speaker design endeavor. You must take care to avoid operating

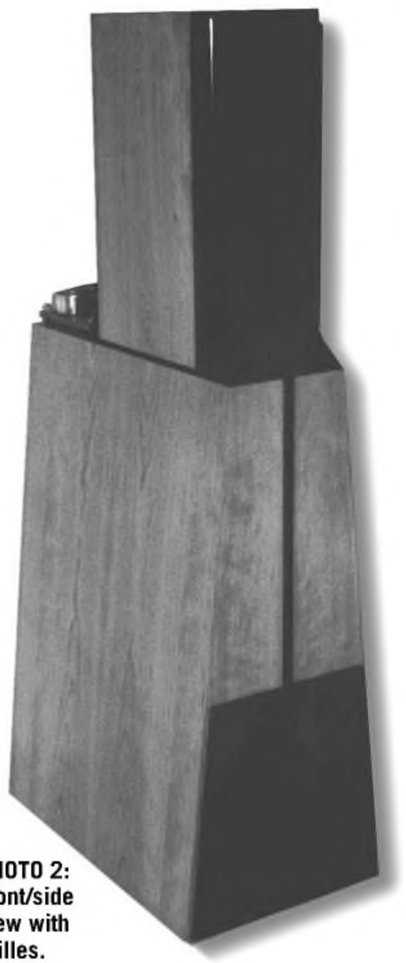


PHOTO 2: Front/side view with grilles.

PHOTO 3: Close-up of satellite with stand-off grille.

PHOTO 4: Close-up of satellite sans grille.

drivers too close to cone breakup modes or in frequency ranges that are beyond the normal capabilities of the drivers. You also need to address the issue of blending drivers with one another in a multiway design. I'm afraid there's no substitute for experience on this one except, perhaps, avoiding the known pitfalls of others!

If you're building the speakers for yourself or someone else you know, then personal preferences come into play here, too. And then, of course, there are economic considerations. My own preferences lean toward better-quality soft domes for tweeters, and paper or treated paper for midrange drivers. I've heard woofers that I've liked with several different types of cone material.

I was willing to spend a fair amount of cash to build the Sessions Monitor, but I wanted to keep the total cost within reason, too. All the materials required to build the project should total less than $2,500, including the active crossover. *Table 1* shows the building materials and parts list for all items required for the project, except for the satellite's passive crossover, which I will detail in Part 2.

You can save yourself a couple of bucks by ordering the Marchand XM-9 active crossover in kit form, as I did. All in all, it's not a bad investment considering what you'll have when it's finished. A commercially designed system of this quality would probably cost between four and five times as much...but hey, that's one of the reasons we're in this hobby, right?

Incidentally, it took me about six months of my spare time to complete the project. I may have more or less spare time and equipment than you do, so the time frame should only be considered a ballpark estimate.

MID-WOOFER

As I stated earlier, I needed an exceptional mid-woofer for this project. Enter the Scan Speak 15W/8530K series "Revelator." This driver comes in two incarnations, the 15W8530K00 and 15W8530K01.

This driver just exudes a feeling of quality! The basket is designed to expose the voice coil to outside air for cooling. You can tell by looking at the surround that it has serious excursion capability. The magnet system is beefy and utilizes their patented SD-1 system. The cleverly designed treated paper cone is actually cut diagonally and re-glued to (I assume) break up resonance. It is not cheap, but you do get what you pay for here!

Running a quick box model on my Harristech Bass Box Pro design program revealed the K01 version to be better suited to sealed boxes, while the K00 is probably better for vented boxes. Obviously, I chose to use the K01. Manufacturers specifications state an f_S of 30Hz and a X_{MAX} of 6.5mm. After running the box model, I was satisfied that this driver would have the low-frequency extension required for the project.

PHOTO 5: Full view, rear.

TABLE 1
BUILDING MATERIALS AND PARTS LIST

QUANTITY	ITEM AND USAGE	SUPPLIER(S)
DRIVERS		
2	SEAS T25CF002 (E-011) Millennium tweeter	Madisound
4	Scan Speak 15W8530K01 Revelator mid-woofer	Madisound
2	Shiva woofer	Adire Audio
BUILDING MATERIALS		
2	¾″ × 4′ × 8′ MDF sheets	Home Center
2	[illegible]/8″ × 4′ × 8′ particleboard (liners and backers)	Home Center
1	[illegible]/8″ × 4′ × 4′ particleboard (grilles)	Home Center
1	¾″ × 8″(W) × 5′(L) maple board (sat. cab. braces)	Home Center
1½	4′ × 8′ sheets of wood veneer	Home Center
1	6″ diameter "Quicktube" or "Sonotube" (sat. cab.)	Home Center
PARTS		
10	¼-20 threaded inserts (pin blocks & bass cab.)	Home Center
8	¼-20 hex head bolts (sat. crossover board mount)	Home Center
8	¼-20 knurled brass thumbnuts (crossover brd. mnt.)	Home Center or Reid Tool
8	¼-20 hex head bolts (mount "Small Tenderfoot")	Home Center
8	¼-20 × 1½″ (L) socket head bolts (pin block mounts and spikes)	Home Center
4	¼-20 × 1″ (L) socket head bolts (wedge mount)	Home Center
12	¼-20 hex nuts (pin block, wedge, "Small Tenderfoot" mounts)	Home Center
8	1″ diameter cone-shaped faucet washers (sat. cab. btm.)	Home Center
24	[illegible]/8″ diameter std. faucet washers (under sat. crossover board)	Home Center
8	½″ O.D. × [illegible]/8″ I.D. × [illegible]/8″(L) nylon spacers(stand-off grille mounts)	Home Center
16	Small "ball and socket" type grille fasteners	Meniscus
8	"Small Tenderfeet" (bass cab. to sat. cab. interface)	Audio Advisor
6	"Large Tenderfeet" (bass cab. btm.)	Audio Advisor
4	24″ long sections of "T tracks" (top of bass cab.)	Hartville Tool
10	Pair "BIG POSTS" binding posts	Madisound
6 lbs	"Acousta-Stuf" synthetic acoustic fill	Parts Express
ACTIVE CROSSOVER		
1	Marchand XM-9	Marchand Electronics

Note: Satellite passive crossover components detailed in Part 2

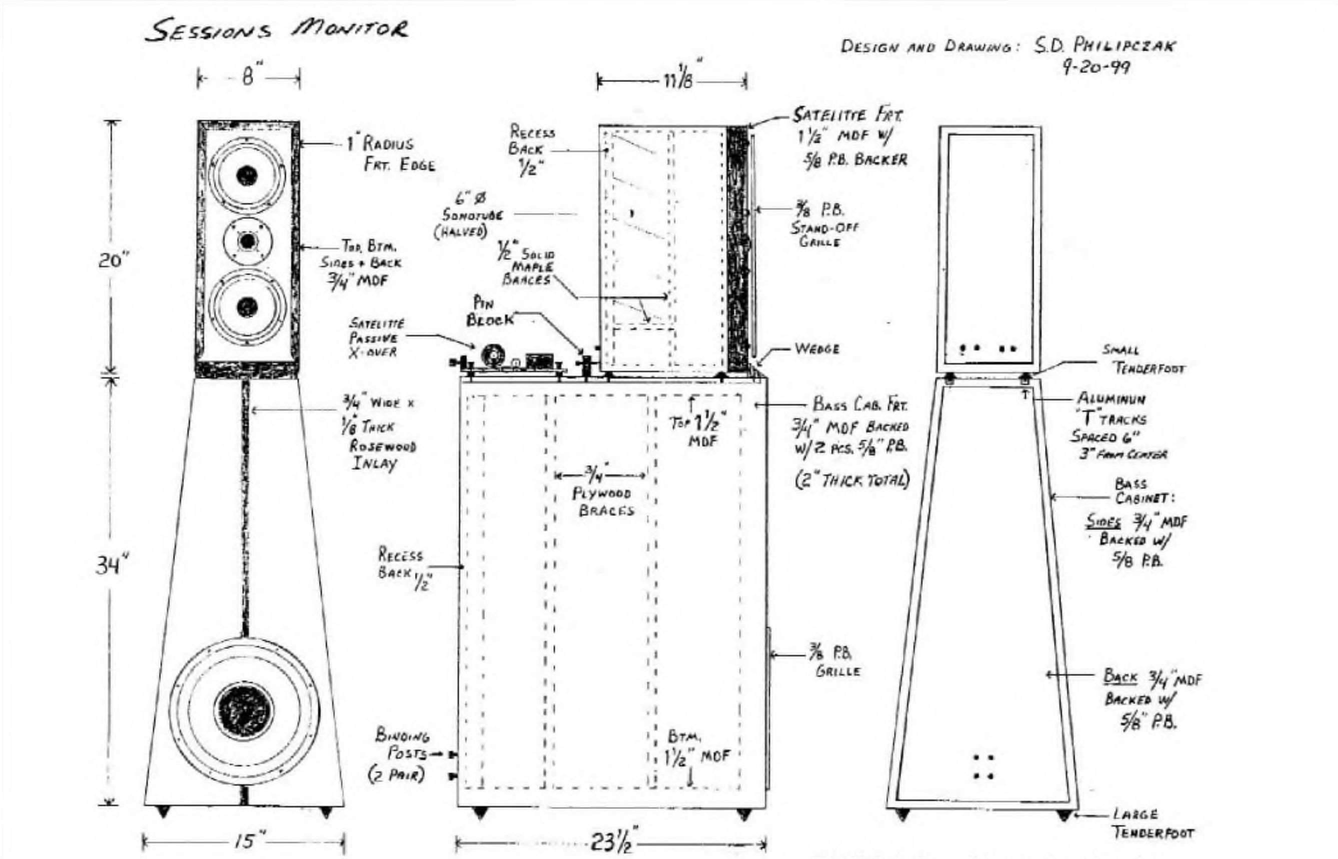


FIGURE 1: Complete system drawing.

WOOFER SELECTION

First, I started looking at 12″ woofers suited for sealed box designs. There are a number of 12″ woofers available that would probably work in the Sessions Monitor, especially considering the fact that the woofer doesn't need to do a whole lot of work above 100Hz. After much deliberation, I settled on Adire Audio's (formerly Avatar Audio) Shiva model. This very versatile woofer has dual voice coils, a "Kevlar impregnated paper" cone, a thick (1.6mm) UV treated foam surround, and a heavy (15AWG/B&S) stamped steel basket. One of its major claims to fame is a 15mm X_{MAX}.

The Shiva has a reputation for being a real thunderous beast. But, it can also do justice to music quite well, especially in a sealed enclosure. The Shiva worked out well considering that I did not choose to use a dual woofer setup, which would not have worked with the shape I wanted to use on the bass cabinet. You can find all the information you

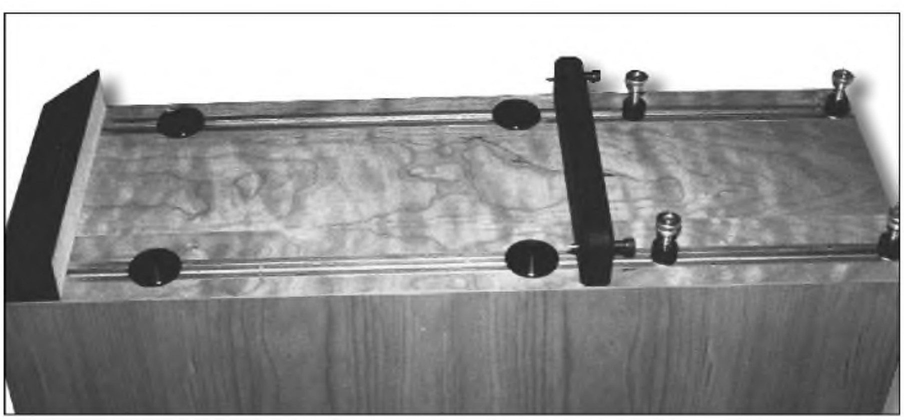


PHOTO 6: Top view of bass cabinet showing "T" tracks and all mounting hardware.

need on this driver through a "white paper" available on their website.

Another feature that attracted me to this driver is what Adire's white paper refers to as "RDO," or resistively damped operation. This involves using a known resistance across the second (passive, unused) voice coil to act as an electromagnetic brake to dampen the bass response. The smaller the resistance, the stronger the braking action. I surmised that between this feature and the controls on the Marchand XM-9 active crossover, I could tailor the bass response to fit my (or most other's) listening environment.

As far as low-frequency extension goes, I would be satisfied with an f_3 (3dB down point) of around 30Hz. Since my listening tastes lean mostly toward acoustic music, jazz, and some classical, this would be adequate, considering the 12dB per octave bottom-end rolloff that is theoretically produced with a sealed box. There were several suggested alignments in Adire's white paper that looked promising, so I had no reservations about this choice as well.

At any rate, I was going to test both the mid-woofers and woofers on my new toy...err-ah...tool, the Peak Instruments "Woofer Tester" (available from Parts Express) prior to final design of the enclosures. I used the manufacturer's Thiele/Small parameters to figure appropriate cabinet volumes for both the bass and satellite cabinets until I could actually measure my own samples. I understand that Adire has recently upgraded the Shiva to a MarkII status with a small improvement in the magnet system and another millimeter of X_{MAX} travel, along with a price reduction. These minor changes should be no problem with the Sessions Monitor.

TWEETER SELECTION

Which tweeter to use was the hardest decision of all, mainly because the choice was pretty much wide open. This is often the case with a three-way system, as the midrange drivers or even the mid-woofers used in the Sessions Monitor are capable of high-frequency extension into a range that will not require a low resonant frequency (f_S) from the tweeter. Due to my own preferences, I looked mainly at soft domes, which still gave me a lot of choices.

One tweeter that interested me was the SEAS T25CF002, or E 011 Millennium. This is also a fairly new driver, although I've seen it in at least one commercially designed system. This is a 1″ soft dome unit with a uniquely designed magnet system. The Hexadym magnet system is based on six rectangular-shaped, neodymium magnet blocks, which have been radially magnetized.

According to SEAS, there are several advantages to this configuration, including the elimination of internal cavities. More information is available on their website, of course. The only downsides were that it's rather pricey, and I hadn't heard it yet.

Since I was taking a chance on the other two drivers, it seemed only natural to take some risk here, too. Knowing the excellent reputations of these driver manufacturers, I was willing to take some risks on their latest offerings, especially since I have used some of their stalwarts with success in the past.

Now that most of the decision work was done, it was time to order drivers, start drawing, iron out some small construction details, and think about crossovers—both active from bass to mid-woofers, and passive from mid-woofers to tweeter. *Figure 1* is a drawing of the complete system. All the dimensions you should need to build the project are on the drawing on *Fig. 1*. A detailed drawing of the satellite will be provided in Part 2 of this article.

PHOTO 7: Top view of bass cabinet with crossover mounted.

TABLE 2
WOOFER TESTER RESULTS FOR ADIRE SHIVA
RAW DRIVER OR CLOSED BOX RESULTS:

f_S	3.9061Hz
Q_{TS}	0.8207
cone area	0.0507m^2
Q_{ES}	0.8971
V_{AS}	4.5976ft^3
Q_{MS}	9.6357
V_{AS}	130.2443 ltr
Re	5.4000Ω
Rm	63.4031Ω
R1	18.5034Ω
Voice coil inductance Le	1.9076mH
Efficiency	84.80dB 1W/1m
BL	10.53 Tesla meters
Cm	361.178 microns/Newton
Mm	122.7511 grams
Rm (mechanical)	1.914 mechanical ohms (Ns/m)

TABLE 3
BASSBOX PRO BASS CABINET BOX MODEL

BOX PROPERTIES
Description
Name: Sessions Monitor Bass Cabinet
Type: Closed Box
Shape: Prism, square (optimum)
Serial No: 1
Comment: Drivers tested on Woofer Tester
Finish: Hand rubbed wipe on polyurethane
Veneer: Natural Cherry
Core Material: ¾″ MDF
Joinery: Rabbet and dado
Bracing: (2) ¾″ plywood
Box Parameters

V_b =	85 ltr
Q_{TC} =	0.861
F_3 =	31Hz
Fill =	heavy

DRIVER PROPERTIES
Description
Name: Shiva 12″
Type: Standard one-way driver
Company: Adire Audio
Serial No: 557
Comment: T/S parameters taken from Woofer Tester
Piston: Kevlar impregnated paper
Suspension: Treated foam surround, flat progressive cloth spider
Frame: Stamped steel 15AWG
Voice Coil: 2″ wound on Kapton former
Terminals: Heavy tinsel leads
Magnet: 60 oz
Configuration

NO. OF DRIVERS 1
Mechanical Parameters

f_S =	23.9Hz
Q_{MS} =	9.64
V_{AS} =	130.2 ltr
Cms =	0.396mm/N
Mms =	112g
Rms =	1.744kg/s
P-Dia =	9.745″
Sd =	481cm^2

Electrical Parameters

Q_{ES} =	0.897
Re =	5.4Ω
Le =	1.9mH
Z =	6.48Ω
BL =	10.06Tm [0]

Electromech. Parameters

Q_{TS} =	0.821 [0]
1W SPL =	85.01dB
2.83V SPL =	86.72dB

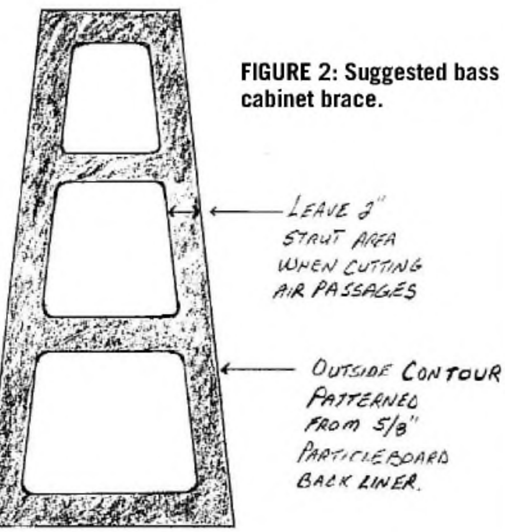


FIGURE 2: Suggested bass cabinet brace.

You'll want to add some rubber faucet washers under the crossover board to help damp vibration. I also put some stainless-steel washers under the faucet washers where they meet the "T tracks." The four low-profile cone points you see are "Small Tenderfeet" available from The Audio Advisor. They come with a small hole in the underside that you will need to drill out (#6 or #7 drill) and tap for ¼-20 threads. The other two items you see are the wedge and the pin block. I shall explain.

I decided that it might be feasible to obtain a physical time adjustment between the bass driver and satellite by setting the satellite back. The precise distance required would best be determined by using measuring equipment that I unfortunately do not possess. The next best thing is your ears and some pink noise.

I set the satellites on the cone points, which were threaded onto ¼-20 bolts, but were not tightened down completely to the "T tracks." Then, I played some pink noise from a test CD and enlisted help from some friends. Sitting directly between the finished speakers with the full system playing pink noise, I had my helpers slowly pull the satellites back as I listened to the pink noise. The point where the pink noise sounds smoothest is where you want the satellite to reside.

This is a subtle thing, but it is there. You must be patient and make several trials to see whether you can repeat the spot. This idea was suggested to me in an e-mail from Andre Perreault. You can check out his very interesting designs and ideas on his website, www.E-speakers.com.

After you decide where you want the satellite, you can determine how deep to make your wedge, which is simply two pieces of ¾″ pine glued together and cut at an appropriate angle to accommodate the depth. Mount the wedge to the cabinet with two ¼-20

I tried to make the external dimensions work with the cabinet volumes required, and still make good use of standard size sheet and wood veneer sizes. I will get into all of the particulars of how I constructed the cabinets later in this article, splitting the bass cabinet and the satellite into separate sections.

NEW USE FOR AN OLD SHOP HAND

Photo 6 shows the scheme I came up with to mount everything to the bass cabinet. I wanted everything to be solid, yet still be adjustable from front to back (you'll read why later). The rails you see routed into the top of the bass cabinet are called "T tracks," which are available from several sources; I obtained mine from Hartville Tool. I have these handy little devils in several of my home-made shop jigs, and one day it dawned on me to use them for this purpose.

In case you have not used them, they are basically solid bars of aluminum with an inverted T-shaped slot milled through them from one end to the other. The tracks utilize standard ¼″ hardware, so you can use any length bolt in them and hex and square nuts, too. There are four standard 1½″ × ¼-20 bolts with knurled brass "thumb nuts" on the back of the cabinet for mounting the crossover board. *Photo 7* shows the mounted satellite crossover.

socket head Allen bolts. The idea is to not have a 90° step from the top of the bass cabinet to the satellite front and to provide a solid stop for the pin block. I also covered the wedge with some soft material called "Foamtastic," which is available at most craft stores.

The pin block idea came about when I decided to use a soft mount from the cone points to the satellite bottom. I made four recesses approximately ⅛″ deep on the bottom of the satellites with a Forstner bit for 1″ cone-shaped faucet washers. These have a small center hole that fits nicely over the cone point tips and locate the satellite and provide a soft mount.

The potential problem is the possibility that the satellite cabinet will actually move front to back from the force of the mid-woofer cone movement. Even though the satellite cabinet is fairly heavy (approximately 37 lbs), I wanted to make sure this would not be an issue. The pin block prevents any possibility of this movement and potential source of distortion.

PHOTO 8: Assembled bass cabinet, prior to front baffle installation. Note horizontal braces used on prototype not necessarily recommended (see text and drawing).

The block is fabricated from ¾″ MDF (medium density fiberboard) and has threaded inserts for the spikes that pin the satellite cabinet against the wedge. I will detail this a little more in the construction section of the article.

ACTIVE CROSSOVER CONSIDERATIONS

With most of the details ironed out and a workable drawing on hand, I anxiously awaited the arrival of the drivers. I ordered the Shivas first, because I planned to build the bass cabinets first, then the satellites. This allowed me time to split some of my expenses and give some further thought to crossovers.

I chose to use Marchand's XM-9 for the active crossover. I purchased the "deluxe, balanced" kit version of the XM-9 because I wanted to be able to use amplifiers with either balanced or unbalanced inputs. The only way I could do this was to assemble the balanced version of the kit and add the RCA connectors afterward. This proved to be no problem, and it gave me the versatility I was looking for.

The design and flexibility of the XM-9 are very good, as are the parts quality and cabinet. The assembly instructions entailed a number of recent updates and were somewhat tricky to organize, but if you do any type of electronics assembly work, you'll have no problem assembling the kit. I spoke briefly with Phil Marchand to double-check myself on wiring the unbalanced connectors. He was very accommodating and a pleasure to deal with.

If you build the kit as per standard instructions, you will have fourth-order attenuation slopes on both the low- and high-pass filters. You can change that by following specific instructions in the manual. With the Sessions Monitor, I may have been able to use lower-order slopes than the standard fourth-order; however, I decided to use the fourth-order slopes as a starting point.

The crossover frequency is variable by way of a resistor block with four ¼W precision 1% metal-film resistors soldered in place. Changing the crossover frequency simply involves swapping the resistor modules. My kit worked flawlessly right after I finished it and has behaved well ever since. I highly recommend it.

GETTING STARTED

I started construction with the bass cabinets, and if you decide to build the system, I strongly suggest you do the same, for two reasons: First, you can split your expenses. The Adire Shivas are reasonably priced, and so are the MDF and particleboard you will need to build the cabinets. Later, when it's time to build the satellites, things are going to become a little more expensive.

Second, and more importantly, if the top width dimension of the bass cabinet does not come out to exactly 8″, you can adjust for that when you build the satellites. You need the top of the bass cabinets to match the widths of the satellites as closely as possible, and this is much easier to achieve if you build the satellites last because there are no angled dimensions to deal with.

I ordered my Shiva woofers direct from Adire Audio, and they arrived in a timely fashion. After unpacking and a quick visual inspection (all was well), I proceeded to break in both woofers on a signal generator set to 25Hz for approximately 24 hours. After break-in, I measured the woofers on the Peak Instruments Woofer Tester mentioned earlier. This marks my first project using this time-saving and helpful piece of equipment. It works rather well and is a whole lot faster than using the standard setup with generator, frequency counter, and meters.

The manufacturer provided driver measurements from a LAUD2 test; however, from the data listed it appeared that both voice coils were in parallel. Since I was going to be using only one voice coil in my design, this is the way I tested them. *Table 2* show the results from one of the drivers. The results from the second woofer were very close to the first sample, indicating good quality control from the manufacturer.

I then imported the Thiele/Small parameters into BassBox Pro. *Table 3* and *Fig. 3* show these results. I could not model the exact shape of the Sessions Monitor bass cabinet into the program, so I used a rectangular approximation.

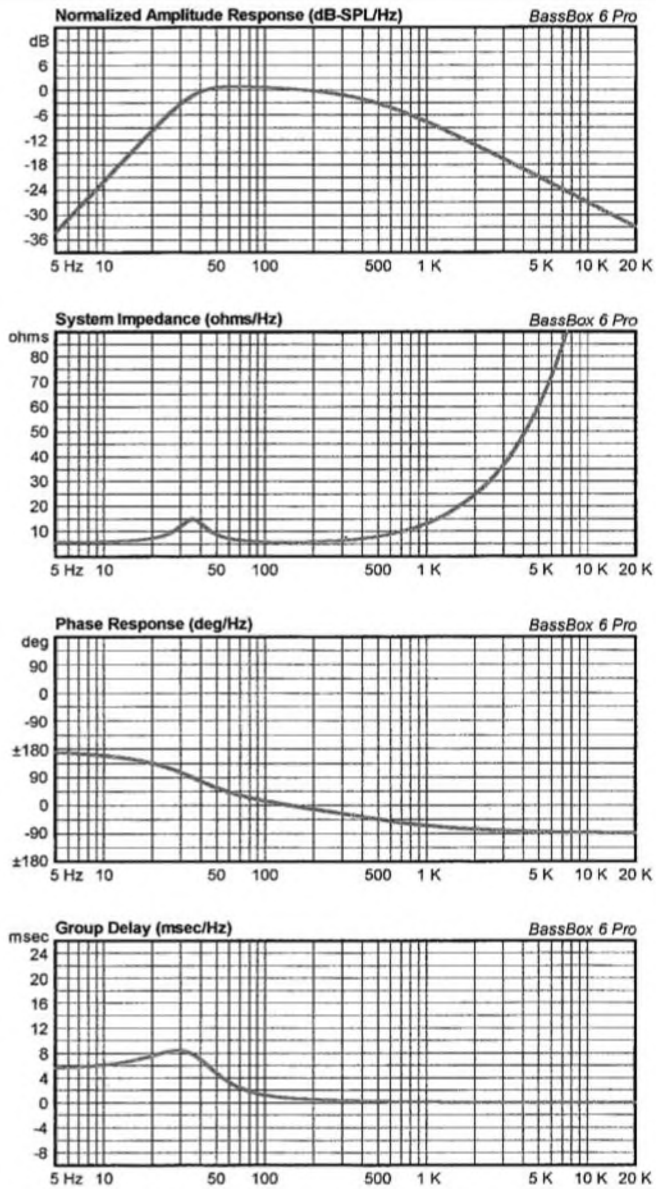


FIGURE 3: BassBox Pro bass cabinet graphs.

From the results shown of the woofers I had on hand, I could obtain an f_3 (–3dB down point) of 31Hz with a Q_{TC} of .86 in a cabinet volume of 85 ltr. This was almost exactly what I was looking for. Time to start making sawdust.

First, I must apologize for not having an exact wood sheet cutting diagram. I had not originally intended for anyone other than me to duplicate this project, so I hadn't attended to such matters. The final result, however, turned out well enough that I believe it is definitely worth sharing with my fellow DIY speaker-building enthusiasts. I chose the major dimensions to make reasonably good use of standard 4 × 8 sheets.

BUILDING THE CABINET

Although it looks as though it might be a little tricky, building the bass cabinet is not at all difficult. You can do it all with minimal standard equipment—a router with edge guide, circle jig, and table saw. Keep the drawing (*Fig. 1*) handy and follow the text instructions.

You'll note that the tops and bottoms are doubled-up pieces of ¾″ MDF. The back and side panels are ¾″ MDF backed with ⅝″ particleboard. The front panel is ¾″ MDF backed with two layers of ⅝″ particleboard for a total thickness of 2″.

Start by cutting the side panels. You will note that the finished depth is

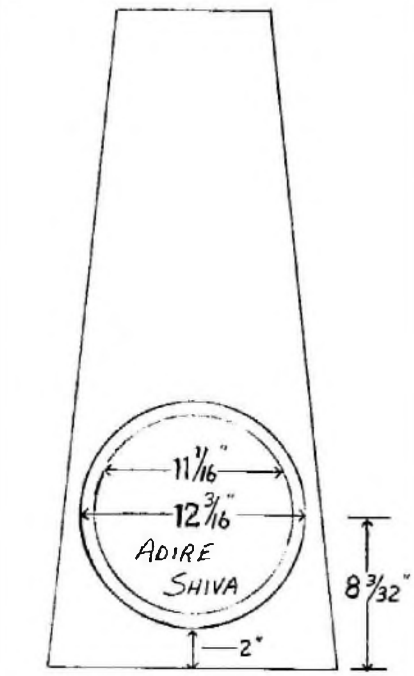


FIGURE 4: Woofer cut-out diagram.

23½″, so you can start by rough-cutting a 4′ × 8′ sheet of MDF lengthwise in half. The final width dimension of the side panels comes out to 22¾″, because the front baffle board will lay on top of the side panels with its doubled-up ⅛″ particleboard liners residing inside.

Cut the height to 34¾″. The extra ¾″ is required to compensate for the triangular shape and the rabbet overlap (the drawing shows the finished height of bass cabinet at 34″). Rout a rabbet into the tops and bottoms of the side panels, ⅛″ deep by 1″ wide. Now make the top and bottom pieces.

The edges that face the side panels will need to be cut at approximately 6°. The bottom should be 14¼″ at its widest (bottom) point by 22¾″ deep. The top should be 7¼″ at its narrowest (top) point by 22¾″ deep.

Once all the outer pieces are cut and the rabbets are in the top and bottom of the sides, you'll need to rout or cut a dado into the inside edges of all the panels for the back panel. Locate the rear edge of the dado ½″ in from the back edges of the cabinet so the back will be recessed in ½″. Make this dado ¾″ wide by ¼″ deep (make sure the depth is exactly ¼″).

When finished, dry-fit the outer shell and hold it together temporarily with screws. Do not tighten the screws yet. Check that the cabinet is symmetrical with a bevel, ensuring that both sides are at the same angle and the top and bottom are parallel. Lay the cabinet shell on a flat surface when you do this to ensure that the dado grooves for the back will stay in line on all the panels. When satisfied, tighten all screws. You will note that the side panel's outsides will overhang the tops and bottoms at the rabbet edges. This is done to compensate for the angle without having to cut the side panel edges at an angle also.

After assembly is complete and all gluing is dry (do not glue anything yet), simply rough down the rabbet overlaps of the side panels flush with the top and bottom. You will also note a slight gap at the rabbet joint where the top and bottom meet the sides. You could eliminate this gap by cutting the rabbets in the side panels with a dado blade set to the same angle as the top and bottom side edges. But this is not necessary. You can fill the gap with glue, and the inside liners will provide plenty of strength at the joints.

Rough-cut a piece of ¾″ MDF about 16″ × 34¾″ for the back. Lay the cabinet shell on top of the back piece and trace the inside lines of the cabinet shell onto the back piece. Add 1/16″ to the traced line on the back to produce a slightly loose fit of the back into the dados of the sides, top, and bottom. Carefully cut the back right to this line with either a band saw or a hand-held jig saw. This cut does not have to be pretty because it will be hidden, but the sizing should be good.

I also cut a shallow (about 1/64″ deep by ¼″ wide) step into the inside surface of the back's edges with a router and edge guide to allow an easy fit into the ¾″ dado groove of the cabinet shell. Remove the bottom and slide the back into the dado grooves. Replace the bottom and ensure that all has remained symmetrical. Screw the sides, top, and bottom to the back through the centerline of the dado groove. Do not overtighten these screws, as the back's fit may not be perfect.

Now, you are ready to make the top and bottom liners from another layer of ¾″ MDF. Note that all the inside liners do not go all the way to the front edge. You need to allow for the two ⅛″ front-panel backers and also a slight sealing gap. So, the liners should be about 20 and 1/16″ deep.

BRACING

When you are fitting the depth of the liners, use two pieces of scrap ⅛″ particleboard, laying them on top of the liner and making sure there is still about 1/16″ or slightly more space to the front edge of the cabinet (*Photo 8*). You'll note that I used horizontal bracing for the prototype. Although I do not detect any sound-related problems with the horizontal bracing, the prototype cabinet does not fare as well as I would like on the old knuckle-wrap test.

The primary resonance modes of MDF and particleboard are in the 300/400Hz range, and as noted earlier, the bass cabinets don't do a whole lot above 100Hz. This could explain why I don't really hear any problems. However, I recommend that you use vertical bracing as shown in the drawing and *Fig. 2*.

The large, flat panels of the bass cabinet are most likely to be better stiffened with vertical bracing, especially considering the almost 2′ depth of the bass cabinet. This is really the only change I would make were I to re-build this system.

After fitting the top and bottom liners, temporarily screw them in. Now, temporarily unscrew and remove the bottom and the back panel. Replace the bottom again (with its liner).

CABINET LINERS

Rough-cut a piece of ⅛″ particleboard for the back liner to the same dimensions you used to cut the back piece. Trace the cabinet inside edges onto it, the same as you did for the back (with the top and bottom liners in place). Cut to the line, ensuring that you leave the line intact for a close fit here. Check the fit of the liner with the dry-fit cabinet.

The fit here must be good. Take your time and do some hand work to get it right. You want a small gap all the way around, just enough to easily slide the liner in and out of the cabinet. You need a good fit with this liner because you will now use it as a pattern for the second cabinet back liner, the four pieces of ⅛″ particleboard for front baffle liners, and the four vertical ¾″ plywood braces. This will ensure that your cabinet stays symmetrical from front to back.

Using the fitted back liner as a pattern, trace its outline on five more pieces of ⅝″ particleboard and repeat on four pieces of Baltic Birch Plywood (BBP). Rough-cut the pieces on a band saw or with a hand-held jig saw. Leave no more material than necessary to preserve the traced line. Screw the back liner/pattern on each of your previously mentioned rough-cut pieces one at a time and follow the pattern with a flush-cutting router bit. Now, all the ⅝″ particleboard liners and the four ¾″ vertical braces will be exactly the same.

Lay the cabinet on its back, remove the bottom, and slide the back panel into place again. Slide the ⅝″ particle-

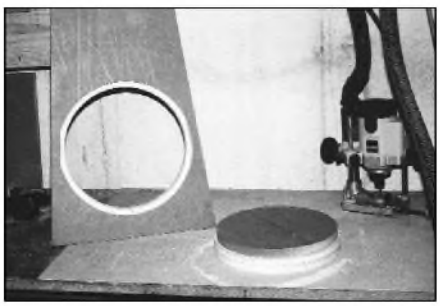


PHOTO 9: Cut-out for Shiva woofer. Note close proximity to edges; centering is critical.

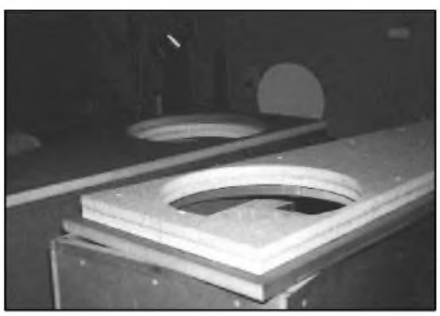


PHOTO 10: View of bass cabinet with woofer cut-out.

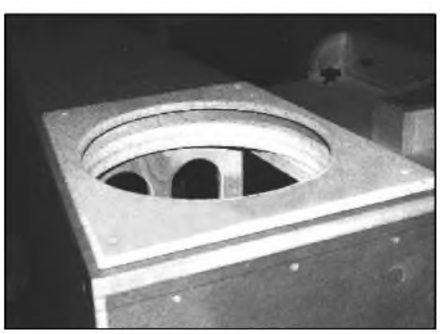


PHOTO 11: View of bass cabinet grille.

board back liner into the cabinet on top of the back panel and screw it in place so that you can relocate it exactly later. Replace the bottom and screw the back panel to the sides, top, and bottom again with a few screws as before, ensuring you do not overtighten the screws.

You are now ready to make the side liners. First, make the side liners in a single piece, then section them apart for the braces to fit in between. Cut the appropriate angle on the top and bottom edges of the side liners to fit against the top and bottom liners. Use some scrap pieces if you need to find the exact angle. Cut the side liners 19 and ⁵⁄₈″ deep for now.

After cutting the liners, slide them into the dry-fit cabinet and make sure they fit OK. Do not screw them in yet. Now you can cut the side liners into three unequal sections and space them as in *Fig. 1*. The first section (starting at the back of the cabinet) can be 5″, the second section 7″, and the third will be approximately 6″.

Before you start cutting the side liners in sections, you should finish the BBP braces. Cut air passages in the braces as shown in *Fig. 2*; or you can rout circular holes in them if that suits your fancy. It's not a bad idea to use a small round-over bit and radius the edges of the cutouts, too. Your hands will thank you later.

The braces should already fit the inside contour of the cabinets, because they have been patterned from the back particleboard liner. Check them now; you should be able to slide them into the cabinet without forcing them. After you finish the braces, you can cut the first (back) section of the side liners at approximately 5″ wide and temporarily screw them in place. Drop the first vertical brace in and make sure it is touching the side liners on both sides top to bottom.

Cut the second section of side liners at approximately 7″ wide and fit and screw them in place on top of the first brace. Now drop the second vertical brace in place, ensuring that it fits tightly to the side liners. Measure from the front surface of the second brace to the front edge of the top/bottom liner. This will give you the exact width you need to cut the last section of side liners. All the liners should be even with each other and somewhere between 1 ⁵⁄₁₆″ to 1⁵⁄₈″ from the front edge of the outer shell. This will allow space for the front baffle's doubled up ⁵⁄₈″ line and a bead of sealing silicone.

FINAL ASSEMBLY

Once all the liners are fitted and temporarily screwed in place, recheck the cabinet again for symmetry while it is still dry-fit. Mark a centerline where the back and bottom liners meet and where the top and back liners meet. If all is OK, you can now disassemble the cabinet and prepare for final assembly of the basic shell.

Start by gluing the back liner to the back and the top and bottom liners in place. Also glue the first (back) set of side liners in place at this time. When I glue laminated panels, I spread the glue evenly with a small, short-napped paint roller. Make sure that all is placed the same as when it was dry-fit.

After you have the back panel assembled, you'll want to fit up and install your binding posts or terminal cups. The back liner will most likely need to be relieved if you use binding posts as I did. I used Madisound's BIG POSTS at standard ¾″ spacing. I installed two sets of posts so I could hook up the second voice coil of the woofer for the resistive damping operation option mentioned earlier.

Relieve the back liner with a router as needed to mount the binding posts. Mount the binding posts and solder on a 3′ piece of the wire of your choice to each set of posts for the woofer. Assembling the rest of the cabinet can become a little tricky; but if you follow the order I give you here, it will go together with no problem.

With the bottom assembly lying flat on your work table, glue the left side to the bottom (the back section of the side liner and bottom liner are already glued together). Before you screw the left side and bottom together, stick a small piece of scrap ¾″ MDF in the dado groove where the panels meet to ensure that the dado groove stays lined up. Screw the bottom and left side together and remove the scrap piece.

Next, apply glue to the back panel assembly and slide it into the dado groove. Make sure that the center marks you made on the back and bottom liners match up so that you'll know the back is all the way into the left side's dado groove. Screw the bottom to the back panel. Screw only the bottom of the left side to the back for now. Do not screw the center or top of the left side to the back at this point.

Now, lay the partially assembled cabinet down on its back to finish the assembly. You will need to glue and assemble the top and right side more or less simultaneously. Apply glue to the top assembly and loosely screw it to the back and left side. Make sure the center lines you marked earlier on the top and back liners match up. Apply glue to the right side and screw it to the bottom first. Complete the assembly now by screwing the tops of the sides to the back and top assemblies.

After the basic cabinet shell dries, seal the back, side, top, and bottom liners with silicone. Apply glue to the front edges of the first section of side liners and the sides, top, and bottom where the first brace will go and drop the brace in place. Glue and screw the second (middle) section of side liners in place. Seal them with silicone to the top and bottom.

Repeat the above procedure and glue the second vertical brace in place. Now glue and screw the front sections of side liners in place and seal them with silicone to the top and bottom. Sand the sides flush to the top and bottom (the rabbet overlaps will protrude slightly). You are now ready to make the front baffle.

FRONT BAFFLE ASSEMBLY

Glue (laminate) two of the ⁵⁄₈″ particleboard pieces (that you patterned after the back liner earlier) together and set them aside to dry. Make sure they are perfectly aligned to each other; then hold them with screws and/or clamps to dry. While the front liners are drying, you can rough out the front MDF baffle. Lay a piece of ¾″ MDF on top of the bass cabinet shell (shell lying on its back), and trace the bass cabinet outside profile on to the back of the panel.

Mark several lines around the traced line at a little less than ¾″ in so that you will know where to place the doubled-up front baffle liner. Now, rough-cut the front panel close to the line on a band saw or use a hand-held jig saw. When the front liner assembly is dry, temporarily screw a couple of window pull handles on one side at the top and the bottom. You'll need these to remove the liner assembly as you check the fit in the cabinet shell.

Make sure that you can drop the liner assembly into the cabinet shell and that it goes all the way down and touches the front set of side liners. You'll need a good fit here—not too tight or too loose. When the front liner assembly is down in the shell to the side-liners, it should "sit" about 1/16″ below the front edge of the shell.

When you are satisfied with the fit, remove the liner assembly and glue it to the front panel using the inner marked lines as the placement guides. Be sure to clean up any glue where the front panel and liners join, as this will affect the fit. Once the front panel assembly is dry, drop it into the cabinet shell and clamp the front MDF panel tight to the sides, top, and bottom. If you have any bad gaps here, now is the time to correct them.

If all looks good, temporarily screw the front panel on with a few screws (no glue yet). I actually do this from the sides instead of the front. MDF tends to split fairly easily when screwed into the edges, unless your pilot hole is exactly the right size and depth. Particleboard is a bit more forgiving in this regard.

Drill your pilot holes/countersinks through the sides and into the particleboard front liner at a slight angle toward the front. This will help draw the front panel tight to the shell. Also, keep the pilot holes about ½″ to ⅛″ back from the front edge of the shell.

After screwing the front baffle assembly temporarily in place, use a router with a flush trim bit to finish-size the front to the cabinet shell. The reason for keeping the screw pilot holes back from the front edge is so the guide bearing from the flush trim bit does not drop into them.

When you are finished flush-routing, remove the front panel and mark a centerline on the front of the panel from top to bottom. Make sure that this centerline is accurate because the woofer cutout is a little close to the outside edges and if it's off-center, it will show rather obviously. The dark-colored strip on the front of the bass cabinet in the photos is actually an inlaid piece of East Indian Rosewood. It is ⅛″ thick and ¾″ wide. This is, of course, an option.

If you do decide to use it, you can either rout its recess channel before or after veneering the cabinet. It is less risky to do it before veneering as I did. Then you can flush-rout the veneer to the channel. You can do this with a solid carbide flush trim bit with a short pilot.

It's easier to rout the channel before the woofer hole is cut out; otherwise, you will need to install a scrap piece across the woofer hole to support the router. If you decide to do the inlay trim piece, its color should contrast to your choice of wood veneer. You should surface the trim strip through a thickness planer before you glue it in, as you'll prefer to do minimal sanding.

Drill a pilot hole for your router's circle jig on the center mark at 8 and 1/32″ up from the bottom. Drill this hole all the way through the panel and liner assembly, because you will need to work from the back as well as the front. It is best to drill the pilot hole on a drill press, but if you don't have one, try to keep your hand drill as straight as possible.

I flush-mounted the Shiva, because its cork gasket is ¾″ thick to accommodate the height of the surround. Follow the cut-out dimensions in *Fig. 4*. You will actually have to rout the recess (outer) step all the way through the front MDF panel (which is about where you'll need to stop depth wise).

Use a 1″ diameter bit for the step, then use a ¼″ bit to cut the through hole. Rout as deep as possible from the front, then finish the through hole from the back. Refer to *Photos 9* and *10* to see what the front panel should look like when finished.

When you've finished routing the woofer mount hole, check it with the driver to make sure the woofer fits correctly. If all is OK, then you will soon be ready to glue on the front panel. But first, pack in the stuffing, so you do not have to work through the woofer hole. Stuff the bass cabinet with 2 to 2.5 lbs of "Acousta-Stuf" synthetic fill (available from Parts Express).

Now you can prepare to glue on the front panel. Apply a generous bead of silicone sealant to the front edges of the front side liners. Apply wood glue to the front edges of the MDF cabinet shell. Apply wood glue to the side edges of the doubled-up particleboard front baffle liner.

Now, drop the front panel in place and clamp it tight to the cabinet shell. Screw the front panel on while it is clamped. Reach through the woofer hole and smooth out the silicone sealant as best you can, while it is still wet. Allow the glue to dry overnight before continuing.

COMPLETING THE BASS CABINETS

After the front panel assembly is glued on and dry, you can prepare to add some of the finishing touches, such as a grille to cover the woofer (*Photo 11*). Install ¼-20 threaded inserts on the bottom of the cabinet for the large cone points. I used "Large Tenderfeet" available from The Audio Advisor. I also used three points (two in front, one in the back) instead of four to produce a tripod effect to prevent the cabinet from rocking on an uneven floor.

After that's done, set the cabinet on its bottom and prepare to rout in the recesses for the "T tracks" on the top. When you order the "T tracks," order 24″ long pieces. Using an edge guide with your router, rout the channels for the "T tracks." Space the channels exactly 6″ apart, from center to center, each spaced 3″ from the cabinet center. Stop the channels at ½″ from the front of the cabinet and take them all the way through the back side.

The depth of your final pass is critical. The top of the "T tracks" should be flush with the finished veneered cabinet, so figure the final channel depth accordingly (*Photo 6*).

After routing the channels, square off the front radius left from the cutter with a wood chisel. Cut the sections of "T-tracks" to length so that they are flush with the back of the cabinet, but do not install them yet. Once the routing and fitting of the "T tracks" is done, you can fill all of the screw holes, sand the filler, check your work, and prepare to cover the cabinet with your choice of wood veneer. My cabinets are finished in natural cherry.

After the veneering is complete, rout the veneer out of the woofer hole, the "T track" channels, and the front inlay strip channel if you are installing one. Glue in the inlay strip and sand it flush with the veneer. Finish-sand the veneer and apply your favorite finish. I used four coats of gloss Minwax Wipe-On Poly, sanding lightly with 220-grit dry paper between coats.

I buffed the final coat with a fine grade of synthetic steel wool on a random orbit sander and finished with a coat of finishing wax. This provides a finish that is durable and yet not too thick. It has no brush marks and the fine dirt particles are removed with the synthetic steel wool so it feels perfectly smooth. It has a nice sheen without looking too glass-like. It looks much like the finish you see on better-grade kitchen cabinets.

When the veneer finishing is done and before you apply any wax, mask off the veneer and the binding posts and paint the back of the cabinet flat black. Install the "T tracks" with countersunk, flat-head wood screws. Put the end of the track that you cut toward the front so that the finished end shows at the back of the cabinet.

Connect the leads from each set of binding posts to each voice-coil terminal pair. You will connect only one set of posts to your amplifier. The other set of posts connected to the second voice coil are there in case you decide to use the "resistive damping operation" option I mentioned earlier.

Apply gasket tape to the woofer and install it. I use McFeely's Square Drive washer head wood screws, which are non-slip and have a good coarse wood thread. I've tried the "T-nut" and machine-screw method, which works fine, but the screws constantly need tightening after playing the speakers for a while. If you ever have any trouble with a wood screw stripping, you can always drill out the hole and repair it with a wood dowel.

With the bass cabinets finished and waiting to be mated with the MTM satellite, this concludes the first part of the article. Part 2 describes the construction of the satellite, along with putting the whole system together. Part 2 also includes some final thoughts and listening impressions. ❖

SOURCES

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"T tracks"

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XM-9 active crossover

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Square drive screws

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e-mail: sales@parts-express.com
Acousta-Stuf
Peak Instruments' woofer tester

Testing the PSB Stratus Silverⁱ

Reviewed by Joseph D'Appolito and Dennis Colin

The PSB Stratus Silver^i is a 2½-way floor-standing loudspeaker. The driver complement comprises two 6.5″ midbass drivers, one above and one below a 25mm metal dome tweeter. Although the system appears to be an MTM, the bottom woofer is primarily active at lower frequencies, augmenting the bass response. The system is vented.

I ran a series of impedance, frequency response, and distortion tests on the Silver^i. *Figure 1* is a plot of system impedance magnitude. At low frequencies the plot displays the double-peaked curve of a vented system. The impedance minimum of 3.76Ω at 27.5Hz indicates the vented-box tuning frequency. The low-frequency peaks are much smaller than those of many of the other vented systems I have tested for *audioXpress*. This indicates a possibly overdamped reflex action. More on this later.

There is a second local impedance minimum of 6.6Ω at 170Hz. Impedance phase lies between +28° and −14° over the full audio range. This should be a relatively easy load for most amplifiers. I would rate this a 4Ω loudspeaker. The impedance peak of 9Ω at 1kHz in *Fig. 1* is probably caused by the interaction of the woofer and tweeter crossover networks forming a parallel resonance at that frequency. We have seen this phenomenon several times before.

FREQUENCY RESPONSE

Figure 2 shows the PSB Stratus Silver^i full-range on-axis frequency response. This response is obtained as a combination of the far-field quasi-anechoic response and properly summed near-field woofer and near-field port responses. I placed the microphone along the tweeter axis at a distance of 1.25m to produce the far-field response. And then I spliced the near- and far-field responses together at 200Hz to produce the full-range response[1].

The response shown in *Fig. 2* has been normalized to 1m to obtain system sensitivity. Sensitivity averages 88.1dB between 250Hz and 1.2kHz. There is a broad, shallow response dip in the on-axis of 3dB between 1.2 and 7.5kHz. This might induce a somewhat dark and recessed character to program material, but you should reserve judgment until you see the complete horizontal polar response picture.

Notice the rather slow fall-off in low-frequency response below 150Hz. This is in sharp contrast with more typical vented systems. Typically, response holds up well to about a half-octave above the box frequency and then falls off sharply.

Figure 3 gives a clue as to the source of this behavior. This figure shows plots of woofer and port near-field responses along with their acoustic sum. Typically, in a high Q reflex system you will see a sharp drop in woofer output at the box frequency. With the Silver^i there is only a slight hint of this output reduction. The port is providing little woofer cone control and is increasing total acoustic output by less than 10dB. Numbers in the range of 15–18dB are more typical. In smaller rooms the Stratus's slow rolloff may be a benefit, complementing room gain for a more balanced bass response.

WOOFER/TWEETER TIMING

Figure 4 shows the Silver^i step response. It is obtained by a numerical integration of the system impulse response. The ideal step response should be a single rapid rise followed by a smooth decay through the 0.00 level. *Figure 4* shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival. It is followed by the woofer pair arrival, peaking about 0.4ms later. The drivers are both connected with positive polarity, but the system is not time coherent.

The exact woofer delay is best determined by examining the excess group delay plot (not shown). Excess group delay is a very sensitive measure of inter-driver delay. This plot shows the woofer pair to be 300μs behind the tweeter.

CROSSOVER ACTION

The Silver^i has two pairs of binding posts for bi-wiring. This allowed me to measure the response of the individual drivers (*Fig. 5*). The crossover frequency is 1922Hz. In the octave below crossover, tweeter response falls 22dB. Above crossover, woofer response falls off at 12dB/octave in the first octave. Beyond this point woofer response falls at 24dB/octave. The dual slope behavior is a result of the frequency response contouring of the bottom woofer.

CUMULATIVE SPECTRAL DECAY

The Silver^i cumulative spectral decay (CSD) response is presented in *Fig. 6*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves for-

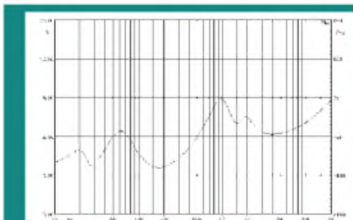


FIGURE 1: PSB Stratus impedance plot.

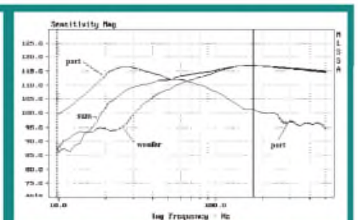


FIGURE 3: Near-field woofer and port responses plus their sum.

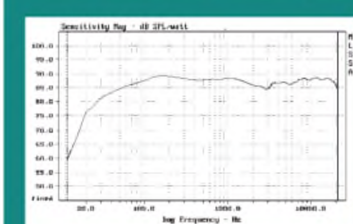


FIGURE 2: PSB Stratus full-range frequency response.

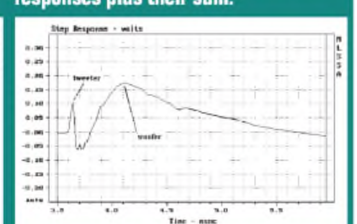


FIGURE 4: PSB Stratus step response.

ward from the rear. Each slice represents a 0.07ms increment of time. The total vertical scale covers a dynamic 35dB range.

Ideally the response should decay to zero instantaneously. Inertia and stored energy that take a finite amount of time to die away, however, characterize real loudspeakers. A prominent ridge parallel to the time axis would indicate the presence of a strong system resonance.

The first time slice in *Fig. 6* (0.00ms) represents the system frequency response. Initial tweeter high-frequency decay is relatively good; the bulk of its response has decayed away in about 1.2ms. However, two distinct response ridges at 2.4 and 3.75kHz appear beyond 1.4ms. It will be interesting to see whether these ridges have any sonic imprint. Low-frequency decay is typical for two-way systems.

HORIZONTAL POLAR RESPONSE

Horizontal polar response is shown in *Figs. 7* and *8*. *Figure 7* is a waterfall plot of horizontal polar response in 10° increments from 60° right (–60°) to 60° left (+60°) when facing the speaker. All off-axis plots are referenced to the on-axis response, which appears as a straight line at 0.00 degrees. Thus, the plotted curves show the *change* in response as you move off-axis.

For good stereo imaging the off-axis curves should be smooth replicas of the on-axis response with the possible exception of some tweeter rolloff at higher frequencies and larger off-axis angles. For home-theater applications a more restricted high-frequency response is desirable.

Figure 7 reveals that the dip in on-axis response fills in as you move off-axis. For example, at ±20° and 3kHz, response is up 1.7dB relative to the on-axis response. The 20° off-axis response is actually smoother than the on-axis response.

The average response over a 60° horizontal window (±30°) in the forward direction is shown in *Fig. 8*. This average response is one of the smoothest I have measured. Between 200Hz and 18kHz response is within ±1.5dB. This is excellent horizontal performance and suggests good direct field coverage in the primary listening area with only small changes in timbre with position. Image stability should be very good.

Due to the size of the Silver^i, it was not possible to safely mount it on my test turntable for vertical polar response measurements.

PSB STRATUS SILVER^i TOWER

FREQUENCY RANGE	
Response	**(Anechoic Chamber)**
On-axis at 0° ±3dB	35–21,000Hz
On-axis at 0° ±1½dB	40–20,000Hz
Off-axis at 30° ±1½dB	40–10,000Hz
LF cutoff –10dB	26Hz
SENSITIVITY	(1W (2.83V) at 1m, IEC-filtered pink noise, C-weighted)
Anechoic chamber	89dB
Typical listening room	91dB
IMPEDANCE	
Nominal	4Ω
Minimum	4Ω
INPUT POWER	(RMS, clipping <10% time)
Recommended	15-250W
Program	200W
Dynamic peak	400W
ACOUSTIC DESIGN	
Tweeter (nominal)	1″ (25mm) Aluminum dome with ferrofluid
Woofer (nominal)	2 × 6½″ (165mm) Polypropylene cone Rubber surround 1½″ (38mm) voice coil 28 oz (794g) magnet
Crossover	2,100Hz, 24dB/octave Linkwitz-Riley 500Hz, 18dB/octave Butterworth
Internal volume	2.00ft^3 (56.3 ltr)
Design type	Bass reflex 2½″ (60mm) front port
SIZE (W × H × D)	9½ × 39 × 13¾″ (241 × 990 × 349mm) with 1″ (25mm) base
WEIGHT	
Net	56 lb (25.2kg)/each
Shipping	70 lb (31.8kg)/each
FINISH	Black ash or dark cherry wood veneer or high-gloss black
FEATURES	Dual, 5-way gold-plated solid metal binding posts, adjustable spikes and rubber levelers
MSRP US$	$1,899/pair $2,049/pair high gloss black

PSB Speakers, 136 Myrtle St., Boston, MA 02114

HARMONIC DISTORTION

I performed harmonic distortion tests at an average level of 90dB SPL. Ideally, harmonic distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can produce false readings by reducing the level of the fundamental while boosting the amplitude of a harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Second and third harmonic distortions at 50Hz and 90dB SPL were 3% and 2%, respectively—a rather good result. All HD distortion was below 1% above 110Hz, which is also a good result.

INTERMODULATION DISTORTION

I next measured intermodulation distortion. In this test two frequencies are input to the speaker. Intermodulation distortion produces output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion.

Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I examined woofer intermods first by inputting 900Hz and 1000Hz signals at equal levels. These frequencies should appear predominantly in the woofer output. I adjusted total SPL with the two signals to 87dB at 1m. Because steady tones are used in the IM test, I thought it safer to use a lower power level to prevent possible tweeter damage. Principal woofer IM products occurred at 800, 1100, 2800, and 2900Hz. However, the overall level was only 0.4%, which is an excellent result.

I measured tweeter intermods with a 10kHz and 11kHz input pair also adjusted to produce 87dB SPL at 1m. I observed only one IM product at 12kHz. Total distortion was 0.06%. Again, this is a very low figure.

The last IM test examines cross-intermodulation distortion between the woofer and tweeter using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic distortion lines, confusing the results.) Ideally, the crossover should prevent high-frequency energy from entering the woofer and low-frequency energy from entering the tweeter. A single IM product appeared at 10.9kHz at a level of 0.03%. Along with the North Creek Borealis Loudspeaker (*SB* 7/00), this is the lowest figure I have measured for this type of distortion.

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 9* shows the Silver^i response with the grille on, but referenced to the response with the grille off. That is, it plots the *change* in response under the two conditions.

Below 1.5kHz the grille has little effect. Above this frequency, however, the grille causes ragged response deviations of +1 to –3dB. As usual, the grille is there only for cosmetic effect and to protect the drivers from prying fingers. The perfect grille is still the Holy Grail of speaker design!

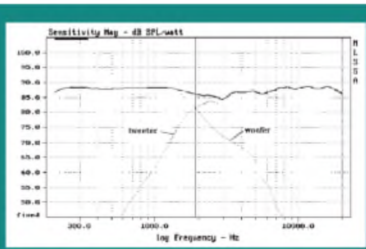


FIGURE 5: PSB Stratus system and individual driver responses.

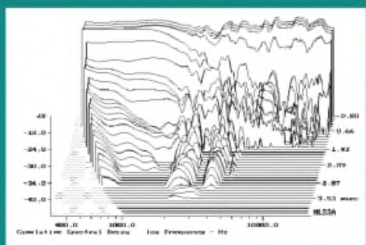


FIGURE 6: PSB Stratus cumulative spectral delay.

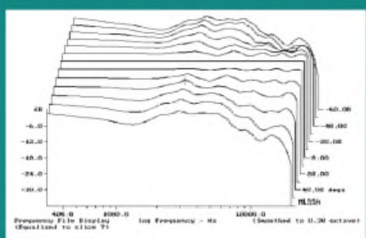


FIGURE 7: PSB Stratus horizontal polar response.

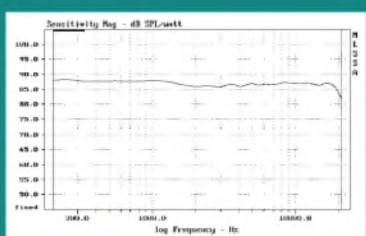


FIGURE 8: PSB Stratus average horizontal response (±30°).

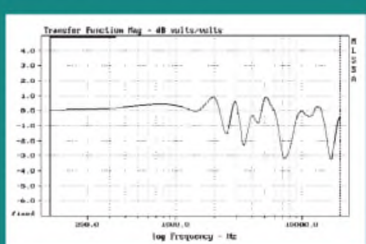


FIGURE 9: Effect of grille on PSB Stratus frequency response.

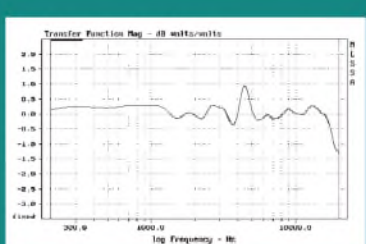


FIGURE 10: Frequency response matching of two Stratus samples.

Two samples of the Silver^i system were available for testing. All of the tests I have described so far were conducted on one sample. Frequency response matching between the pair is shown in *Fig. 10*. This is a plot of the response difference between the first and second samples. Pair matching is within ±0.5dB over most of the audio range. There is a brief mismatch of 1dB at about 4.5kHz, but overall this is excellent pair match performance.

CLOSING COMMENT

After finishing these tests I had a chance to talk with Paul Barton, the "P" and "B" in PSB Stratus—the "S" is for his wife, Sue. Paul informed me that the design axis for the Silver^i is set at about 36″ off the floor, which would place it along the upper woofer's axis. At this position the dip in on-axis response between 1.2 and 7kHz will be much smaller. ❖

A note on testing: The PSB Stratus Silver^i was tested in the laboratories of Audio and Acoustics, Ltd. using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems. Acoustic data was measured with an ACO 7012 ½″ laboratory grade condenser microphone and a custom designed wide-band, low-noise preamp. Polar response tests were performed with a computer controlled OUTLINE turntable on loan from the Old Colony Division of the Audio Amateur Corporation.

REFERENCE

1. J. D'Appolito, *Testing Loudspeakers*, Audio Amateur Corporation, Peterborough, NH, 1998.

■ CRITIQUE

■ **Reviewed by Dennis Colin**

EQUIPMENT AND SETUP

In a departure from the usual venue of my room and equipment, I reviewed the PSB Stratus Silver^i at the studio room of Audio and Acoustics, Ltd., designed by Joe D'Appolito. I had pulled a back nerve, and Joe was kind enough to set up the speakers, including the heavy lifting. In addition to being the planetary authority on loudspeaker testing, Joe is also a good guy.

I'll leave any description of the room, equipment, and number of oxygen-free molecular cable pathways to Joe. I can't judge the last category because my discriminatory hearing isn't allergic to oxygen atoms! I would say the room and components are first rate.

PERCEIVED ACCURACY IN PERSPECTIVE

Human hearing sensitivity is capable of discriminating well over 4000 frequencies (the number of resonating hairs, called cilia), over a 120dB range (a trillion-to-one in power, roughly the weight ratio of a large horse to a small hair), thanks to a "computer" of trillions of cells (not one of which is completely understood by science).

But you must place this in the perspective of sonic details that matter to you. For example, you might hear the 0.2dB change in a steady tone caused by moving your head a few millimeters. But no two notes played on any acoustic instrument ever repeat to that degree, so you don't care. In fact, this very variability is part of the "life" (or "soul" or "feeling") of music.

On the other hand, large enough variations in, say, a loudspeaker's response impose a consistent aberration on all the sounds. How much is enough? That depends on your tastes and sensitivity, of course. But below some level of imperfection, as you become absorbed in the music you won't be distracted; with still more accuracy, you won't be able to detect any coloration even when looking for it (as reviewers must).

OVERALL SONIC IMPRESSION

I simply could not find any coloration or other flaws not attributable to the recordings. This doesn't mean the speakers are "perfect"; if I had quickly A/B'd them with live music, there certainly would have been audible differences. But those may have been only the limitations of two-channel stereo, bass rolloff below (estimated) 40Hz, and, of course, maximum SPL.

When reviewing, you're always A/B'ing the reproduced sound with your memory of live music. Since the latter is itself imperfect, there's an acceptable level of coloration that's low enough not to notice, even when trying to. Such is the case with the PSB Stratus Silver^i speakers in this review. Very good, indeed!

DETAILED LISTENING IMPRESSIONS

Hi-Fi News Disc

- Track 2—voices natural, good space, but recording sounded somewhat restricted after hearing Turtle Creek Chorale; see following.
- Track 4—Trumpets slightly bright (as heard on many other speakers), no coloration attributable to the Stratus Silver^i. Strings very natural, space very realistic.
- Tracks 5, 6, 7, 10—all instruments very natural. Percussion was extremely solid; at loud levels drums sounded real.

Then Joe volunteered the disc "Postcards" by the Turtle Creek Chorale.* Well, holy wow! If you want goosebumps, listen to this on good speakers. The voice tone fidelity and 3-D spatial presence on this disc are the best I've heard. And I think the Stratus Silver^i speakers reproduced those voices near-perfect as is possible with today's components. Sorry, *Hi-Fi News*—Turtle Creek is my new reference for voice ensemble (and voice tone period).

The spatial reproduction was also excellent. Dimensionality was second to none (of the speakers I've heard), and coherence was possibly as good as the Tannoy S8LR Dual Concentric (Sept. '01 *aX*). The latter is no mean feat for a conventional (non-coincident) driver array. The Stratus Silver^i shows just how well the MTM, or D'Appolito, configuration can behave as a coherent acoustic point source.

OTHER SOURCE MATERIAL

- **Stereophile Test CD 2:** St. James Infirmary (The Brassworks)—The horns were reproduced extremely well, as was the dramatic sense of 3-D space and dynamics captured on this disc.
- **Go-Go and Gumbo, Satchmo 'N' Soul:** A la carte brass and percussion (Wildchild 04752)—Very realistic in all aspects: tonal naturalness, spaciousness, impact, and imaging were reproduced superbly.

GENERAL COMMENTS

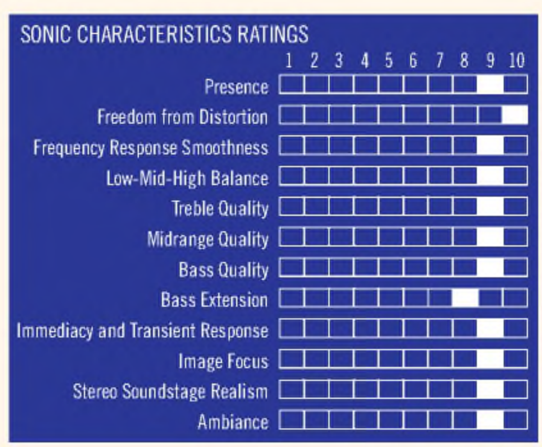
1. I estimate bass extension to be around 40Hz. But even with the lowest octave missing, pipe organ sounded remarkably deep and full.
2. On a variety of rock and blues, the speakers sounded very natural, powerful, and lifelike.
3. Violin tone (one of my most stringent personal criteria) was just superb. I rate the Stratus Silver^i as possibly equal to my reference for violin tone, the Swans MI (with an excellent ribbon tweeter); see *SB* 3/99.

COMMENTS ON MEASUREMENTS

- *Figure 2*—While the smoothness was expected, I didn't hear the mild dip around 3kHz. Normally I would, having complained often about recessed midrange. I think *Fig. 8* explains this:
- *Figure 8*—Joe said that "average horizontal response over a 60° angle is a good indicator of tonal quality." Well, one look at *Fig. 8* certainly agrees with the superb, uncolored tonal realism I heard.
- *Figure 6*—The excellent decay above 4kHz, coupled with the outstanding smoothness in *Fig. 8*, explains the pristine violin tone and ultra-clear transient reproduction I heard. I was not aware of coloration due to the "ridges" Joe mentioned at 2.4 and 3.75kHz. I think that's because these ridges don't appear to start their slow decay until about 30dB below the initial response (which decays that first 30dB very quickly, in about 1 ms).
- Distortion Plots—Joe commented on the good performance here; I thought the speakers were very clean-sounding even at levels loud enough (>100dB) to feel bass and percussion physically.

Overall, I highly recommend this speaker for its unusually excellent naturalness of tone and spatiality, plus its maintenance of clarity over a wide dynamic range.

*Reference Recording; RR-61 CD



Onkyo Home Theater

Reviewed by Charles Hansen and Ken & Julie Ketler

PHOTO 1: TX-DS787 front view.

PHOTO 2: TX-DS787 rear view.

Onkyo USA Corp., 200 Williams Dr., Ramsey, NJ 07446, 201-825-7950, www.onkyousa.com. Suggested retail price: $1,050 US. Dimensions: 435mm W × 453mm D × 175mm H, (17 1/8″ × 17 13/16″ × 6 7/8″); net weight: 16.9kg, 36.6 lbs. Warranty: two years parts and labor.

Onkyo's press release states that its TX-DS787 home-theater audio/video (A/V) model is the only 6.1-channel surround EX receiver in the $1,000 price class and is the first Onkyo home-theater receiver to earn THX Select certification from LucasFilm.

The TX-DS787 is compatible with Dolby Digital and DTS 5.1 program materials. The system is also compatible with program materials encoded for 6.1-channel DTS-ES Matrix with a rear center surround channel. The TX-DS787 has six channels of power amplification rated at 100W per channel into 8Ω with all channels driven (120W per channel into 6Ω).

INTRODUCTION

The receiver has composite and S-video inputs for all four video sources, plus two assignable component video inputs and one component video output. The video circuitry includes component video switching for use with HDTV systems.

In addition to one 7.1 multichannel input for future DVD-Audio players and other components, the TX-DS787 includes analog source inputs for a turntable, a CD player, and a tape deck. You can assign two optical and two coaxial digital inputs to any audio or audio/video source. The unit is equipped with line-level analog preamplifier outputs (after the tone and volume-control alterations) for all eight channels, including the two back surrounds and a subwoofer. A single digital optical output connector provides an MD, DAT, or CD digital rec out signal from the selected digital input.

With these preamp outputs, the TX-DS787 is designed for multi-room, multi-source two-zone operation, with dual-zone infrared remote inputs, an assignable 12V trigger, and stereo speaker outputs for a second zone. The AM/FM tuner provides you with 40 presets. Given the rapidly changing nature of home-theater A/V technology, Onkyo has provided an RS-232C serial port for future software updates. There is also a good troubleshooting guide in the instruction manual.

The programmable Theater-Dimensional (T-D) setup mode allows you to select speaker listening angle, center speaker level calibration, front expander, virtual surround level, dialog enhancement, and room size for each surround mode. With all those surround modes and other options to play with, you are sure to muck up the sound sooner or later. I know I did any number of times in setting up the various menus during my measurements.

In that event, you simply hold down the video-1 button and then press the rec out button. This resets all settings to the factory defaults and puts the TX-DS787 into standby mode. In my opinion, this important instruction deserves a prominent place in the beginning of the manual, not merely as a note back on page 72.

CONTROLS

The receiver has a black, anodized aluminum front panel and heavy-duty steel chassis. *Photo 1* shows the front panel. The left side of the unit features a standby/on switch, standby and zone 2 indicators, power switch, rec out and zone 2 buttons, a standard stereo headphone jack, and audio selector button. The latter control cycles the audio input selection between auto (automatic detection), multichannel, and analog.

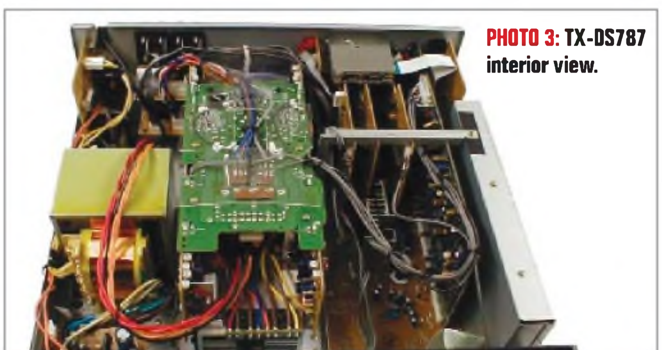


PHOTO 3: TX-DS787 interior view.

PHOTO 4: RC-390M remote control.

The central display screen is surrounded by a number of controls. Across the top are the FM mute/stereo/mono button, tuning up/down selectors (50kHz increments for FM and 10kHz increments for AM), a character/memory control (to add or delete names for radio stations and input sources, and to assign tuner station presets), bass/treble mode, channel-level selector, and zone 2 volume.

The Smart Scan Controller (SCC) knob is to the right of the display and the remote-control IR sensor. This dial controls the setting for the parameter displayed in the central display. Pressing this knob cycles among the controllable parameters (speakers, inputs, listening modes, preferences, and OSD on-screen display setup).

Below the screen are the display input information button, listening mode buttons (direct, stereo, Dolby or DTS, and THX), the OSD menu, on-screen cursor, and OSD exit buttons. The OSD menu will also appear on a TV monitor connected to the TX-DS787.

To the right of the display is a large master volume dial for the main zone. Below this dial are video 4 and video cam input connectors. Finally, there are ten input source buttons below the display area for DVD, video 1–4, tape, FM, AM, phono, and CD. You can assign any of these inputs to zone 2 or record out by first selecting the rec out or zone 2 button.

CONNECTIONS

The left side of the rear panel (*Photo 2*) is festooned with the input/output connectors I described in the introduction. The digital optical jacks are Toslink, and the S/PDIF digital jacks accept RCA-type coax connectors. All analog input/outputs are tin-plated RCA jacks.

There is a ground terminal for connecting a turntable ground wire next to the digital coax jacks. An Onkyo "RI" remote-control connector allows the RC-390M remote control to operate other Onkyo equipment such as cassette decks and CD players also equipped with "RI" connectors. You can control Onkyo DVD players directly with the RC-390M remote.

TABLE 1
MEASURED PERFORMANCE

PARAMETER	MANUFACTURER'S RATING	MEASURED RESULTS
Power output (FTC)	100W per channel, 8Ω 120W per channel, 6Ω	142W per channel, 8Ω
Dynamic power, stereo	2 × 210W, 4Ω 2 × 130W, 8Ω	
Frequency response	20Hz–20kHz 100W, 8Ω	4Hz–54kHz ±1dB, 1W, 8Ω
Output impedance	N/S	0.11Ω 20Hz–1kHz, 0.25Ω 20kHz
Damping factor	60 at 8Ω	
Total harmonic distortion	Max 0.08%, 8Ω; 0.1% 6Ω 1W to rated power	0.031% max, 8Ω; 1W–100W
IMD—CCIF (19 + 20kHz)	0.08%, 1W to 100W	0.008% CCIF 12V p-p
MIM (9 + 10.05 + 20kHz)	N/S	0.01% MIM
INPUT LEVEL AND IMPEDANCE		
Phono	2.5mV, 50k	48k
Line (CD, tape, DVD, video)	200mV, 50k	51k
Multi-channel	200mV, 50k	51k
Subwoofer	36mV, 50k	51k
Coax digital	0.5V p-p, 75Ω	75Ω
DVD, video 1–4	1V p-p, 75Ω (Y) 0.28V p-p, 75Ω (C)	75Ω
Component video 1, 2	1V p-p, 75Ω (Y) 0.7V p-p, 75Ω (P_B, P_R)	75Ω
OUTPUT LEVEL AND IMPEDANCE		
Rec out (tape, video 1, 2)	200mV, 2k2	3k2
Pre out	1V, 470Ω	465Ω
Monitor, video 1, 2	1V p-p, 75Ω (Y) 0.28V p-p, 75Ω (C)	75Ω
Component video	1V p-p, 75Ω (Y) 0.7V p-p, 75Ω (P_B, P_R)	75Ω
Phono overload	110mV, 1kHz, 0.5% THD	123mV, 1kHz, 0.5%
RIAA deviation	20Hz–20kHz: ±0.8dB	±0.35dB
Frequency response (CD in direct mode)	20Hz–30kHz, ±1dB 10Hz–100kHz, +1/–3dB	4Hz–64kHz, ±1dB 4Hz–108kHz, +1/–3dB
Bass control	±10dB at 100Hz	±10.4dB
Treble control	±10dB at 10kHz	±10.5dB
SIGNAL TO NOISE RATIO (STEREO)		
Phono	80dB, (IHF A, 5mV in)	
CD, tape	100dB, (IHF A, 0.5V in)	
Muting	–50dB	–52dB
FM TUNER SECTION		
Tunable range	87.5–108.0MHz, 50k step	
Sensitivity (mono)	11.2dBf, 1.0μV, 75Ω IHF	11dBf
Sensitivity (stereo)	17.2dBf, 2.0μV, 75Ω IHF	
50dB quieting (mono)	17.2dBf, 2.0μV, 75Ω	17dBf
50dB quieting (stereo)	37.2dBf, 20μV, 75Ω	36dBf
Capture ratio	2.0dB	
Image rejection	40dB	
IF rejection	90dB	
Signal-to-noise (mono)	76dB	
Signal-to-noise (stereo)	70dB	
Alternate channel rejection	55dB	
Selectivity	50dB (DIN)	
AM suppression	50dB	
THD+N (mono)	0.2%	
THD+N (stereo)	0.3%	
Frequency response	30Hz–15kHz, ±1dB	20Hz–15kHz, +0.1/–1dB
Stereo separation	45dB at 1kHz 30dB, 100Hz–10kHz	48dB at 1kHz

AM TUNER SECTION	
Tunable range	530–1710kHz, 10k steps
Usable sensitivity	30μV
Image rejection	40dB
IF rejection	40dB
Signal-to-noise	40dB
THD+N	0.7%
Power requirements	5.5A, 440W maximum

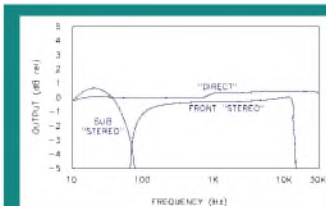


FIGURE 1: Phono stage RIAA response.

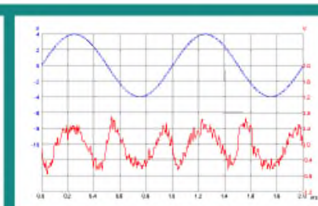


FIGURE 5: Spectrum of residual distortion, 1kHz.

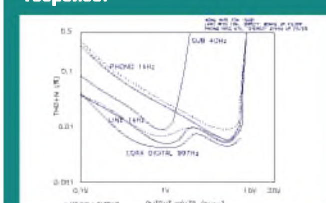


FIGURE 2: Preamp/phono distortion versus output voltage.

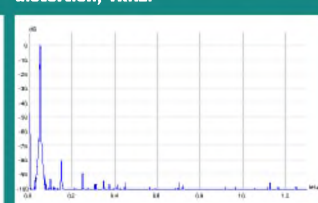


FIGURE 6: Spectrum of 50Hz sine wave.


FIGURE 3: Preamp DSP square-wave response, 1kHz 2.5V p-p 8Ω.

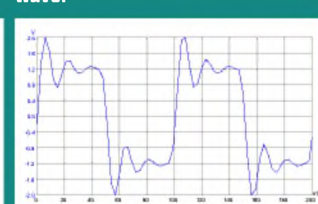


FIGURE 7: Square-wave response, 10kHz 2.5V p-p 8Ω in parallel with 2μF.

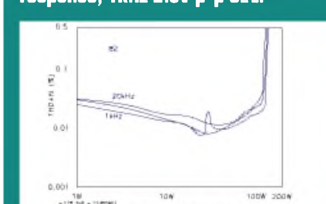


FIGURE 4: THD+N versus output power.

■ Reviewed by Ken and Julie Ketler

Do you find yourself waiting in line at the same coffee shop every morning? Although you're given choices of 32 different blends of coffee or tea—hot or cold, in large, extra large, or jumbo sizes—most probably get the same thing every time you open your car windows and yell into the little circle in the drive-up menu. It's just a sign of our times, isn't it? Any company that knows beans about running a business these days understands that it must cover the plethora of peoples' desires or swiftly declare Chapter 11. Apparently, less is not more!

Another example of this is the audio/video world (you knew we were going there, didn't you?). If you are the owner of a home theater system, you may have a 4.1, 5.1, or 6.1 audio configuration with Dolby Digital, DTS, and/or Dolby Pro Logic decoding. Perhaps you have CD, minidisc, cassette, or DVD players. Maybe you're still using a laser disc player, VHS, or even a Betamax!

Do you use surround sound for movies or music or both? Harrumph! Before you know it, filling out your tax return seems like a real pleasure compared to figuring out which media configuration you need to buy. Enter the Onkyo TX-DS787 A/V receiver.

THE INS AND OUTS

The TX-DS787 steps right up to the plate as a preamplifier/control center, offering you:

- CD in (1)
- Phono in (1)
- Tape in/out loop (1)
- Coaxial digital in (2)
- Optical digital in (2)
- Optical digital out (1)
- Audio/video throughput with both RCA and S-Video connectors (5)
- Line level in/out for center, fronts, subwoofer, surrounds, and back surrounds

The Onkyo TX-DS787 offers six output channels of 100W/8Ω simultaneous power. Additionally, this receiver has an independent two-channel second zone output, which allows you to, for example, run a movie in one room and listen to another input (or radio station) in another room. This is a very attractive feature that saves you the trouble of setting up receivers and music sources all over your house.

As a surround decoder, the Onkyo TX-DS787 receiver can handle Dolby Digital, Dolby Pro Logic, and DTS (Digital Theater System) multi-channel schemes as well as straight stereo and mono. Like many surround decoders on the market today, the TX-DS787 includes some additional DSP artificial "ambience" schemes, which are supposed to enlarge your apparent listening space by adding various delay and reverb effects to the output channels.

As far as we are concerned, these are fun toys, but don't typically sound very good. The only possible exception is a mode called "all channel stereo," which can be useful for filling your room with background music. Great for parties!

At first, we were somewhat overwhelmed by the number of features offered by the Onkyo TX-DS787. Each feature is accessible via Onkyo's On Screen Programming, but this requires a fairly steep learning curve. We highly recommend spending time with the 75-page manual, which accompanies the receiver. Oh yes, we know that operator's manuals are often the first things to be thrown away when people purchase new products, but believe us, you just may want to keep this one. It's well written, easy to understand, and impossible to live without!

After becoming used to programming the TX-DS787, we find many of its features invaluable. For instance, the TX-DS787 includes a very nice backlit programmable remote control. It does a great job learning the commands of our other remotes and, therefore, putting them out of business!

By using the built-in pink noise generator and our Radio Shack SPL meter, we can set the relative level of each speaker to match one another according to our viewing/listening position. Once we set the relative levels, the TX-DS787 is kind enough to remember them for the next time we return to our living room-turned-cinema.

Have you ever turned on your receiver only to jump out of your skin when the program starts at full volume? Oh, the war stories we could tell! This receiver offers the Onkyo IntelliVolume feature, which allows you to set the level at which the TX-DS787 will play when you power it up.

If someone in your household cranked the volume for their umpteenth viewing of *Jurassic Park* during the previous night, your nerves can be spared the next morning when you turn on some Strauss with a relaxing cup of tea. Conveniently, the IntelliVolume function is individually programmable for each input, allowing you to match input levels from different sound sources. You can even limit the maximum volume at which the TX-DS787 will play; an elegant solution to loud "volume-creep," which happens to all of us from time to time.

Besides comprehending different surround systems and allowing many customized setup attributes, the Onkyo TX-DS787 receiver also offers you many different ways to configure your speakers. For surround sound playback, you can use a minimum of four loudspeakers (left/right front and left/right rear) all the way up to a 7.1 configuration! The latter setup requires three front speakers (left/center/right), four surround speakers (left/right side and left/right rear, which require an additional stereo amplifier), and a powered subwoofer! Whew, where are those tax returns?

OUR SETUP

For this review, we used the Onkyo TX-DS787 in both stereo and 6.1 surround mode, which provides the standard left/center/right front channels along with left/center/right surround channels, plus a subwoofer. The center surround channel from the TX-DS787 is not a "discrete" signal, but is derived from the other two surround channels. We used a Boston Acoustics Lynnfield VR 6.1-channel speaker system, which we will review in an upcoming issue of *audioXpress*.

The back panel of the TX-DS787, although a veritable cornfield of connectors, is very easy to follow, but we do have one complaint. The speaker output connectors are quite close to one another, making them somewhat difficult to twist and secure. We highly recommend using banana plugs; save your knuckles!

TEST TRACKS

The Sound of Music—Twentieth Century Fox presents this release in digitally remixed 4.1 surround sound on DVD, which means that the audio comes from the left/center/front channels, a subwoofer channel, and the rear channels in mono. This movie tends not to have wiz-bang effects flying around our heads, but we decided to mention it for those of you who may enjoy classic musicals.

JK: As you probably can guess, this movie pick was mine! The story and music have inspired me ever since I was a young girl. As we watch the DVD release, our living room comes alive as Maria sings "The Hills Are Alive" at the opening of the movie. I hear it mostly through the front speakers, but it is very full and rich sounding. This movie looks and sounds better than I ever remember it.

During "My Favorite Things," Maria's gentle voice calms the children even through the harsh thunderstorm. The opposing sounds of her voice and the thunder work very succinctly together and make the scene extremely dynamic.

KK: There was very little information in the rear channels for the most part. Apparently, the re-mix engineers didn't have enough available signal to produce a "real" surround channel and decided to make a spacious "ambience" channel. Let's face it, when *The Sound of Music* was released in 1965, surround sound wasn't even an idea yet. I am, however, particularly impressed with the left-to-right stereo panning of the characters according to their on-screen locations during close-up shots. The aural dynamics of this classic flick are surprisingly super with the soft sound of the rain surrounding Liesl and Rolf as they sing "You Are Sixteen."

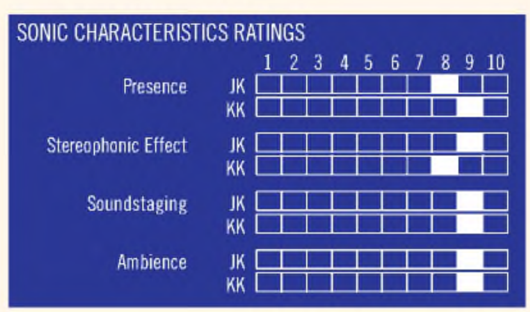
The music, itself, sounds absolutely wonderful, as though it was recorded yesterday! The strings and woodwinds are perfectly crisp, without being strident. The bass violins are full and deep—brilliantly encompassing!

Get Shorty—We're using this DVD simply because we haven't seen it before! Since it was highly recommended to us and we're reviewing a surround receiver, this seems like a good review piece.

JK: When Ken suggested we watch this movie, I thought he was making fun of my vertically challenged stature of barely 5′. Then he handed me the DVD! One part of the movie stands out to me because it makes me glad to be me! Mobster Chili Palmer (played by John Travolta) decides he wants to retrieve his stolen jacket from another mobster. So Chili marches straight to this guy's place, bangs on his apartment door and whacks him straight in the nose as the door opens. The sound of Chili's fist smashing the mobster's nose is so life-like, I can actually hear it crack!

If that isn't enough, Chili turns without a word and walks straight out the door, slamming it behind him. Bam! If I closed my eyes and listened, I would be able to "see" through my ears! This entire scene seems very real. Aren't you glad you're you, too?

KK: The quiet and loud transitions are impressive and at times stunning. Although this movie has its occasional obligatory surround effects, there is a scene at the airport where we are "surrounded" by the whine of jet planes. Although this is a bit jarring, it's diffi-



cult to miss the change of scenes. I suppose that's the goal of the sound engineers...to use the audio portion of the movie to transport the viewer to different places. It worked.

The Perfect Storm—This is a great DVD to use for testing a home theater. The majority of the movie has great music, stormy weather, and crashing sea vessels. The storyline and drama are very moving and are greatly enhanced by a properly set up surround system with subwoofer.

JK: This was a great movie to hear through the Onkyo TX-DS787 sound system. Before the storm hits, the receiver nicely depicts the gentleness of the ocean. The splashes come from all around us, but it is very calming. When the fishing crew lifts a large swordfish out of the water, its fin flaps, hitting the wooden floor of the ship. Each tap from the fish's fin is hard, yet slimy sounding!

During the height of the storm, the aggressive sea envelops us from every angle (every speaker) alternating from the left to the right and center depending on which side the waves are crashing. It is awesome being part of the storm! The 6.1 surround system does a superior job of making us feel as though we're with the crew of the Andrea Gail right in our own living room.

KK: Highlighting the tragic end of the Andrea Gail fishing boat, this movie has many scenes with devastatingly stormy weather and panic. I must say, the subwoofer rumbling as the boat flies over enormous waves only to abruptly hit the waters below, along with the gushing and splashing of water randomly from all channels is staggering!

Days after the sinking of the Andrea Gail, there is a memorial service in a rather large church. The church acoustics are beautifully represented with a choir in the back of the sanctuary (front channels) echoing in the surround channels. Although I can see only a small piece of the ceremony on our video screen, the music is all-encompassing.

Chick Corea "Remembering Bud Powell"—With the rapid death of quadraphonic systems still lingering in the minds of many audiophiles, even after all these years, we were very curious to hear this 5.1 DTS surround mix on CD. Whether multichannel audio will or should become the next new permanent standard for audio is still up for debate, but for now, it's certainly an option for home theater audiophiles.

JK: We listen to a lot of music in our house, but this is the first time we've played a music CD in surround sound. It is wonderful how the instruments seemed three-dimensional. As I listen, I imagine I am in a club watching Chick Corea, rather than having him and his band in our living room (not that they aren't welcome).

KK: I was, admittedly, somewhat afraid of a music-only program in surround. Although I like to stay open-minded, the thought of different instruments coming from all corners of the room is really quite cheesy! However, "Remembering Bud Powell" is tastefully mixed, very pleasant, and enveloping. I can practically hear Christian McBride's fingers rub the strings of his upright bass as he plays. Roy Haynes's meticulous cymbal work is also highlighted by the subtle surround-to-front 3D placement.

This makes me realize, though, that speaker versus listener placement is even more critical for "proper" music listening than it is for conventional two-channel audio. Another lesson I learned is that CDs encoded in DTS surround are unfortunately incompatible with standard CD players and will play a continuous stream of pink noise-like signal. Do I smell another Betamax?

SUMMARY

We auditioned many different pieces of stereo music through the TX-DS787. Overall, we believe it performs best in "straight stereo" mode and isn't bad in "all channel stereo" mode. It tends to have a nice, bright top end, sounding very much like the solid-state device that it is, meaning that tube fans may not want to ditch their valves any time soon.

However, Onkyo seems to have rated the TX-DS787's output power quite conservatively, resulting in very clean-sounding output even at higher SPLs than our household can handle. The good news is that you can customize its configuration in an infinite number of ways to fit your A/V needs. The bad news is that folks who are new to surround sound may have a helluva time figuring out the TX-DS787's menus.

This is a wonderful piece of equipment, especially in the $1,000 range, and we give it two thumbs up for quality and features. With a little patience, the menu-driven nature of this receiver will likely become second nature to you.

Antenna jacks are provided for the AM loop antenna and the 75Ω FM "F" connector. If the receiver is in a location that does not allow the remote beam to reach the IR sensor or exceeds its 16′ range, you can plug an optional remote sensor into the rear of the TX-DS787.

Component video jacks are RCA types and the S video jacks are DIN. All jacks on the rear panel are color-coded to reflect their intended input/output signal. Preamp output jacks are available for connecting external power amplifiers to the front, surround, surround back, subwoofer, and center channels.

Eight pairs of tin-plated multi-way binding posts are provided for the speakers (six main speakers and two zone 2 speakers). These binding posts are on US 0.75″ spacings, so you can use dual banana plugs.

The non-polarized power cord is attached to the unit, and does not have a chassis-grounding third pin. The power cord is looped through a toroidal ferrite EMI suppression core (three turns) inside the unit. There are also two switched two-prong AC outlets rated 1A each.

INSIDE THE AMP

Photo 3 shows the amplifier without the top cover, which engages two tabs on the front panel and is secured with seven screws. It has cooling slots above the central and right-side heatsink areas, while the bottom cover has cooling slots below the power transformer and along the front edge. Vent slots are located in the center of the rear panel, through which the cooling fan exhausts heated air.

While there is adequate finger space under the unit to lift it easily, several screw heads are located in this area, which can make lifting uncomfortable should your hands not be in the proper location.

There are two relay protection boards just inside the speaker binding posts. The large E-I power transformer occupies most of the left side of the unit. Another PC board interfaces with the power and standby switches. The front video input board lies just below the master volume dial.

The center of the receiver has two large heatsinks, each carrying three pairs of Toshiba bipolar output transistors. Only one pair of output transistors is used for each channel for 100W per channel into 8Ω load with all channels driven, so I imagine the cooling fan is necessary for high-volume listening. A caution label notes that speaker impedance cannot be less than 6Ω. The remainder of the power amplifier circuitry for each channel is located on PC boards between the heatsinks.

Most of the circuitry consists of surface-mount discrete components and ICs. The right side of the chassis resembles a personal computer. A large phenolic motherboard supports six vertical PC boards, whose rear-facing input/output jacks protrude through the rear panel. A bracket ties four of the boards together for support.

The shielded AM/FM tuner front end sits just inside the antenna jacks, while a larger shielded box is located on the right sidewall of the receiver. Three TO-220 power devices are mounted on heatsinks located between the vertical PC boards.

The Onkyo TX-S787 is supplied with the RC-390M remote control (*Photo 4*), with LCD back-lighting that makes it easy to read in the dark. This remote has a learning capability and macro functions. The TX-DS787 uses the Smart Scan Setup feature that allows simple, intuitive system setup and control.

A full description of the remote-control capabilities is beyond the scope of this section of the review. Suffice it to say that it duplicates all the controls on the receiver and adds remote-control capability for compatible CD/Tape/DVD and MD operation. A four-page basic operating guide is useful for day-to-day operation.

TOPOLOGY

A schematic was not furnished with the unit. Presumably you could obtain it by ordering the service manual, as is common with many consumer audio components. Each power amplifier channel consists of a low-feedback Class-B 100W per channel complementary-symmetrical amplifier using one pair of 15A 150W Toshiba bipolar output devices.

Analog circuitry (prior to digital conversion) is implemented with 4570 and 5532 dual op amps. The TX-DS787 has 96kHz/24 bit digital-to-analog converters (DACs) for Dolby Digital and DTS soundtracks and music. The dual 24-bit DSP processors use a non-scaling configuration.

An "Optimum Gain Volume Control" feature adjusts low-level sig-

TABLE 2
CROSSTALK PERFORMANCE

FREQUENCY	PHONO (DB)	PREAMP (DB)	POWER AMP (DB)
100Hz	−67	−72	−67
1kHz	−58	−63	−57
10kHz	−43	−53	−53
20kHz	−41	−51	−51

nals to keep them well above the noise floor, even on the quietest passages. The master volume control is an encoder rather than a set of analog volume controls. It communicates directly with the display PC board by means of a three-wire cable.

MEASUREMENTS

I preconditioned the TX-DS787 using white noise at 10W output for one hour. There is no sound other than the protection relays cycling at power-on and shutdown.

The receiver has four listening modes. Direct mode bypasses the digital signal processing (DSP) "sound alteration" sections and other filters, for all-analog performance. I made most of my measurements in this mode, with the tone controls off. For THD measurements I used the test-set 80kHz LP filter to limit out-of-band noise.

The stereo mode is similar in performance to the direct mode when the subwoofer is selected off. With the subwoofer on, the audio is processed by the DSP, sending the low frequencies to the subwoofer output and the higher frequencies to the left/right front outputs. I made some comparative preamp measurements in direct mode, using the test-set 27kHz LP filter to limit noise from the DSP, since the frequency response was consistent with 44.1 to 48kHz digital conversion.

The DTS Surround and THX modes are designed to decode movie soundtrack information on DVD disks, so I did not measure those 24/96kHz modes for their audio performance. I did some listening and, as with most A/V receivers, I found the audio surround effects somewhat exaggerated.

PHONO STAGE

The TX-DS787 has a moving-magnet phono stage, whose performance I measured using the Old Colony Inverse RIAA Network. Input impedance was 48k, although using an ohmmeter, it was initially 50k then rose to 57k. This suggests the cartridge loading is a combination of pure resistive and capacitor-coupled resistive loads. The phono stage preserves input signal polarity.

A fixed 1.63V RMS signal from the distortion test set produces an output from the Inverse RIAA that tracks the equalization standard within ±0.15dB, based on 10mV at 1kHz. The direct-mode left-front preamp output in *Fig. 1* shows 1/3dB peaking at higher frequencies, but still remains within the RIAA limits. I also ran the phono preamp signal through the DSP in stereo mode, with the subwoofer on. The results for the subwoofer and left-front preamp outputs are shown in *Fig. 1* as well. This frequency-band division also occurs when you use the line inputs.

THD+N versus preamp output voltage into 10k is shown in *Fig. 2*. I again used the Inverse RIAA network to generate the phono input signal. The solid phono trace is the direct mode, and the dashed line just above it is stereo mode with the subwoofer off.

The direct-mode phono stage distortion at 2V RMS preamp jack output was 0.012% from 20Hz–4kHz, gradually rising to 0.024% at 20kHz. Using the stereo mode, the distortion at the subwoofer output was noticeably higher—0.47% at 20Hz—while the higher frequencies were only slightly more distorted than the direct mode.

Phono input overload (1kHz, 0.5% THD) was 123mV for 9.5V RMS output, probably representing the ±13V maximum output swing of op amps with ±15V DC supplies. Since the phono stage signal is only available through the following preamp stage, I cannot tell which of the op amps reached its limit first.

The phono stage square-wave response was very good. The 40Hz response showed only a slight tilt, and the 1kHz response was just about perfect. The 10kHz response barely rounded the leading edge, with no sign of ringing or oscillation.

PREAMP LINE INPUTS AND OUTPUTS

Line-level analog inputs are available for CD, tape, DVD, and four video inputs. I used the CD input for most measurements. Input impedance at 1kHz was 51k, although DC resistance was over 1M, indicating capacitively coupled line inputs.

The phono jack video input impedances all measured 10k. The line output impedance was 465Ω at 1kHz, so the output levels varied only 0.2dB for output loads from 10k–100k. Video outputs all measured 75Ω.

The preamp section does not invert polarity in direct mode. However, when an input signal is processed through the DSP, the output polarity constantly varies with frequency. For instance, polarity was normal at 460Hz, lagged 90° at 480Hz, was inverted at 500Hz, led 90° at 525Hz, and was back to normal at 550Hz.

This phase modulation was not evident from the CD input to tape out, however. Tape out impedance measured a high 3k2, and both output level and frequency response were affected somewhat by the tape out load. With a 100k load, the tape out level was the same as that presented to any of the analog inputs, dropping 1dB with a 10k output load. The distortion level was about equal to the distortion test-set oscillator, so the tape output does not appear to be buffered. There was also some drop in frequency response above 50kHz, so I assume the tape out is simply RC-coupled from the selected line input.

The master volume control is a position encoder that requires approximately three turns from minimum "0" to maximum "80." "Ref" appears in the display where "62" would be. Tracking is excellent, with no measurable difference between the left and right front-channel levels from 50mV–5V output. At the minimum setting, I could hear a very faint pop in the speaker. I assume this is a type of muting circuit.

At maximum volume there is noticeable hiss and a very low hum in the speaker. This is not apparent at "Ref" level. Even at relatively low output levels I could occasionally hear the cooling fan quietly cycling on and off.

True unity gain is not available, being somewhere between "64" (−0.63dB) and "65" (+0.42dB). The "Ref" level produced a gain of −2.6dB. Maximum preamp gain ("80" in both direct and stereo modes) was +14.3dB.

For most of the preamp tests I used a volume setting of "77," which produced (as closely as possible) an output of 2V RMS for a line input of 0.5V RMS (+12dB). The right-channel preamp output had slightly higher distortion than the left, so I present its measurements here.

The direct-mode line-input frequency response was very flat. It was down only 1dB at 4Hz and 64kHz, and down 3dB at 108kHz.

When I selected stereo mode, with the subwoofer on, the divided response was similar to that in *Fig. 1*. The receiver response was also the same for a digital CD coax input signal. The subwoofer output was down 1dB at 4Hz and 47Hz. The −3dB point was 63Hz, and −5dB occurred at 75Hz.

I repeated the test in the default DD/DTS Surround mode using the DVD input (not shown in *Fig. 1*). Frequency response in the front channels was 3.7dB lower than direct mode and rolled off above 12.5kHz. The rear surround channels were 8dB lower than the fronts, and their response rolled off above 8kHz. Distortion was about 0.2% at the respective response peaks.

Refer to *Fig. 2* for the line-level THD versus output voltage. The CD input performance in direct mode is the fifth trace down at 1V output. Just above (dashed line) is the distortion after processing the signal through the DSP section. The maximum preamp output voltage (1% THD+N) was 9.5V RMS, the same as the phono stage. The analog inputs were successfully converted to digital without overloading the A-D converters.

(continued to p. 85)

The Ultra Fidelity Computer Sound System, Part 3

Computer subwoofers present their own unique concerns that need to be addressed. This series continues with a look at the design, placement, and construction of these units. **By R.K. Stonjek**

Low-frequency sound is difficult to locate. This simple fact makes it easy to place the subwoofer almost anywhere in a listening room. Creative, maker of the famous "Sound Blaster" sound card and the well-reviewed "Platinum" series of high-quality computer audio hardware, recommends placing the computer's subwoofer in a corner or against a wall to "reinforce the bass" (read it on their website at http://csw.creative.com/ resources).

DESIGN CONSIDERATIONS

While you might not detect the exact location of deep bass, you will notice the subbass delay if it is placed too far away. This is not usually a problem in the hi-fi listening environment. The main speakers are usually placed close to the rear wall of the listening room, making it impossible to place the sub any farther away.

The situation is quite different in the computer environment. Placed behind the monitor, the subbass is at double the distance of the main speakers. With the sub at your feet the distance is triple.

Remote subbass confuses and muddies the words of spoken voice, blurs the location of objects in computer games, and makes following fast-moving sequences in DVD movies confusing and unclear. With three times the delay of the main speakers, voices sound as though they are spoken into an empty barrel.

Most people turn the sub down or off when they need accuracy. The modern computer is a fast and accurate information server. The speakers that work with the computer should be equally as accurate. When it comes to computer speakers, you should think "information."

The location of the subbass for very low frequencies is not critical. Below 80Hz you can place the subbass just about anywhere. At 100Hz you really should place the sub close to the main speakers. At 150Hz you should align the sub as closely as possible with the main speakers.

The problem with the 5.1 (AC-3) system is the separation of subbass at 80Hz. Looking through a few speaker catalogs with recommended cabinets, you will find that a 5″ woofer in a 7-10 ltr ported cabinet will make it down to 80Hz. For a sealed box you need at least a 6″ speaker in around 15 to 30 ltr. This is far too big in the computer environment. To give you some idea of just how big these boxes must be, my 17″ monitor encloses a volume of around 30 ltr!

There is no real reason why the subbass channel crossover point on 5.1 hardware couldn't be at a different frequency or even adjustable. No doubt they will become available, if they are not already by the time this article goes to print.

Ported bass/midrange boxes allow too much high-frequency sound through the port, especially noticeable when listening up close. A 5″ woofer needs a cabinet of at least 15cm width—too much for the critical computer environment. The speakers described in part two have a width of just 11.5cm and a cabinet of only 1.1 ltr.

The Ultra Fidelity Computer Sound System has a crossover point at 135Hz. The sub doubles as a monitor stand in a way similar to those computers that have their tower lying horizontally beneath the monitor. LCD monitors sit happily on top of the sub box as well. The overall subwoofer height is just 15cm.

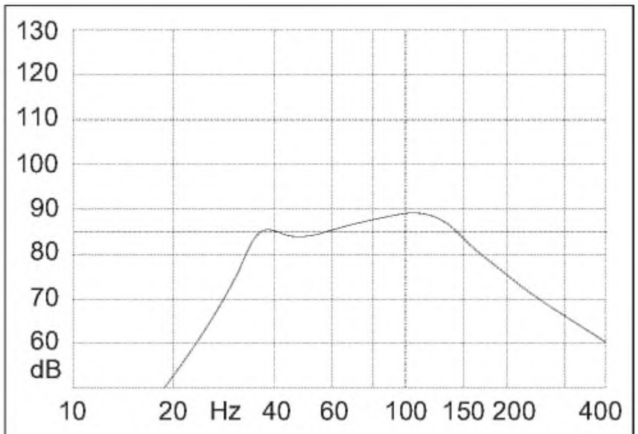


FIGURE 17: SPL at 1W/1m and driver sensitivity.

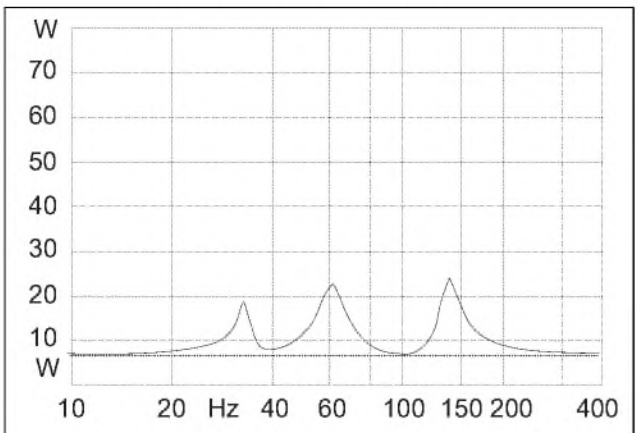


FIGURE 18: Impedance and DCR.

For computer subwoofers, the main design consideration is integration. The subwoofer should be on the same plane as the main speakers and the higher tuned port should be between the two main speakers if possible. The woofer should be as accurate as the main speakers, that is, not muddy, boomy, or delayed. Naturally, the output level must be high enough to keep up with the satellites.

PERFORMANCE

Figures 17–19 show the design tool predictions. *Figure 20* shows the measured output at 1W/1m. The –3dB point is at 35Hz if you take the sloping response into consideration. *Figure 19* shows the main problem you encounter when using small-diameter woofers (4″, Audax AP100G0). While the box tuning and not the speaker size determines the frequency response of the bandpass box, below the frequency range the smaller-diameter woofer will move more freely and soon "bottom out."

Figure 21 shows how this problem is overcome electronically—a high Q notch filter is placed just below the passband. This notch filter acts as a very steep high-pass filter dropping more than 60dB in less than an octave. Sound cards generally roll off at lower frequencies and produce little if any in-

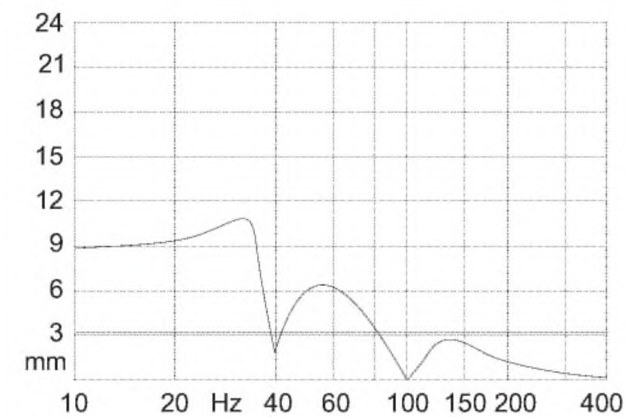


FIGURE 19: Driver excursion and X_{MAX} at 30W.

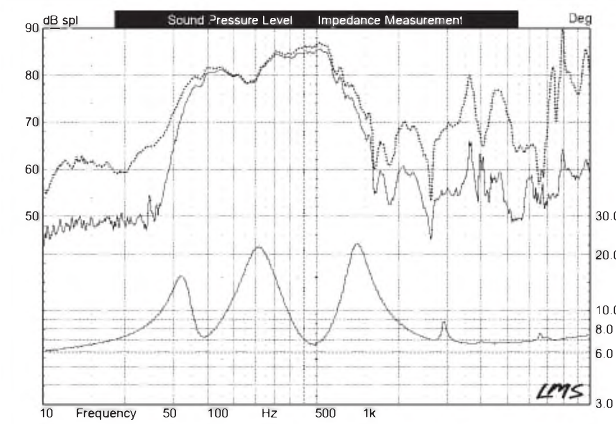


FIGURE 20: Sub with and without active crossover, and impedance.

frasonic noise. The amplifiers used also roll off at the lower frequency.

With a little basket damping and 20W rather than the 30W shown in *Fig. 19*, the speaker does not bottom out except with a few very deep pipe organ pieces played flat out. Rock music rarely contains any really deep bass, the lowest note on the bass guitar being 70Hz; drums are over 100Hz.

By contrast, the tuba, piano, pipe organ, and double bassoon all have fundamental notes below 50Hz. Thus, you can play rock music on this system quite loudly (I measured Rammstein's *Sehnsucht* album at 110dB at the listening distance), while classical music is full and all-embracing, yet misses none of the subtleties.

CONSTRUCTION OF THE SQUARE SUB

There are two versions of the subwoofer which I'll refer to as the "square" and "round" subwoofer. The square subwoofer is much easier to build. The round subwoofer is a better looking and sounding speaker. The high port on the square subwoofer is toward one of the ends, whereas on the round sub it's central. This gives much better bass/mid to subbass integration—imagine how the cube shaped subbass in the corner compares!

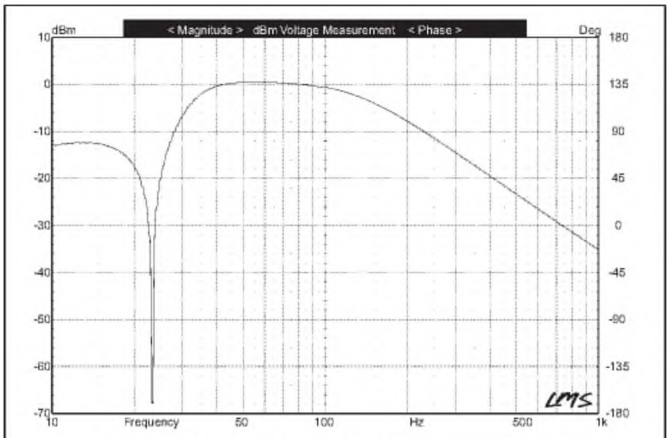


FIGURE 21: Subbass active crossover.

To build the square sub, start by cutting out pieces A to F (*Fig. 22*). (Always wear a protective mask when working with MDF (medium density fiberboard). MDF dust can cause cancer and respiratory problems.) Piece F is a brace; its exact width is not critical. If you have a router with an edge-trimming bit, you can make pieces G and H a little bigger than specified, then trim them back after assembly. G and H are made from 9mm, and all the other pieces are from 18mm MDF (*Photo 10*).

Cut the port holes. Use a hole cutter if possible. Cut the speaker hole. The speaker will butt up against this hole (see the "Assembly" drawing). You might prefer to round over the inside edge of the hole.

Use 10 × 10mm square pine molding to attach the front to the sides. If no molding is handy, use 18 × 18mm MDF. This saves having screws showing through the front of the finished speaker.

Screw all of the parts to the base as shown in *Fig. 22*/Assembly. You should seal the speaker into place using a suitable silicone sealant—black window sealant is my favorite. Make sure there are no leaks. Pack some fiberglass damping material around the rear of the speaker as shown in *Photo 11*. Run the speaker cable out through a hole in the back panel. Seal the hole—I used hot glue.

Now you can glue the cover and screw it into place. This means that the speaker is sealed inside. To retrieve it, you will need to cut open the box. You might choose to include a removable hatch just above the woofer. Cut out a square about 110 × 150mm from the cover before assembly. However, I think this step is unnecessary.

Once you have trimmed the box and rounded the edges, you can sand it. Then use an MDF sealant or suitable undercoat. Make sure you have a well-ventilated area for painting, and always wear a protective mask. Use the same painting procedure that you used for the satellite speakers. My current passion is gloss black with a gold webbing.

You should cut the ports from 50mm PVC pipe and paint it at the same time as the speaker. When dry, wind about four layers of black electrical tape around the end. This should give a flush fit into the port hole. Add or remove tape as needed. When it is about right, place some black silicone in the hole and push the port in by hand. I

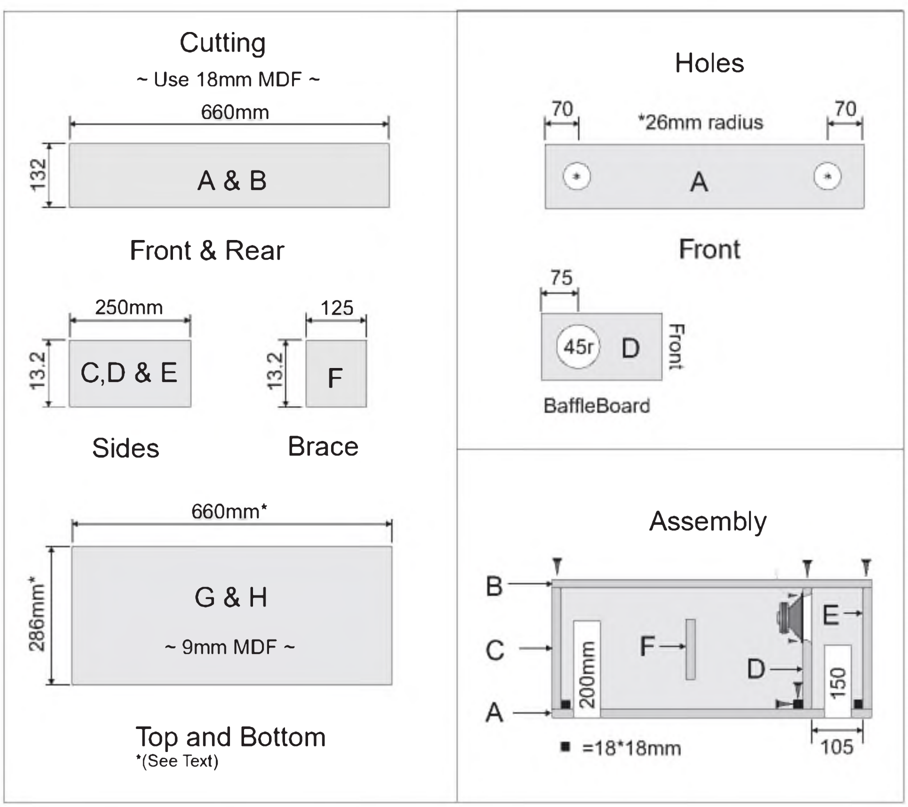


FIGURE 22: Construction of square subwoofer.

spray the gold webbing on with the ports in place.

CONSTRUCTION OF THE ROUND SUB

Cut parts as shown in the cutting list except for part B (*Fig. 23*). If you have a router with a trimming bit—preferably a dual-bearing type—then cut B a little bigger than specified, around 156 × 666mm. You can cut parts F and G together as a single wheel then cut them in half. Most routers have a parallel cutting guide, which you can usually insert upside down and, with a hole drilled through the end plate, turn the router into an accurate circle cutter.

Clamp A and B together and make port holes center and left. The other hole is for the attenuator. Cut the 20mm (radius—40mm diameter) hole from the "A" side. When you're through with the "A" board, remove the clamps and make the 26mm radius hole through the "B" board. The central guide hole made by the hole saw should have passed through board "B" and can be used as a guide for the larger hole.

Keep the best-looking disc left over from the hole drilling. You will use this to mount the attenuator.

Cut the baffle board hole in the center of piece C. The speaker will butt up against this hole (see the "Assembly" drawing). You might prefer to round over the inside edge of the hole.

Make attachment points on the front ends of pieces F and G using a couple of pieces of scrap MDF. Note: the photos show the original experimental version that had the speaker in a different position than the model described here.

Assemble the small chamber as shown. You should seal the speaker in place using a suitable silicone sealant. Make sure there are no leaks. Carefully solder the speaker cable into place. Leave plenty of cable as shown in *Photo 11*. Pack some fiberglass damping material around the rear of the speaker.

Assembly drawings 3 and 4 (in *Fig. 23*) show the assembly of the top and front. Drawing 5 (also in *Fig. 23*) and *Photo 12* show the next step—attaching the curved surface, made of two layers of 3mm MDF cut to around 170 × 1,300mm. If you don't have any assistants to help hold the curved piece, then clamp weights on the ends. Use a pipe clamp to secure the top, then clamp the ends near the front (*Photo 12*). Some "pre-curling" of the 3mm MDF will help.

Have some large tacks handy. If worse comes to worst, you can always try hammering in a few tacks. When dry, trim the ends a bit, then repeat the whole procedure with the second curved piece (after removing any tacks).

PHOTO 10: Subwoofer framework.

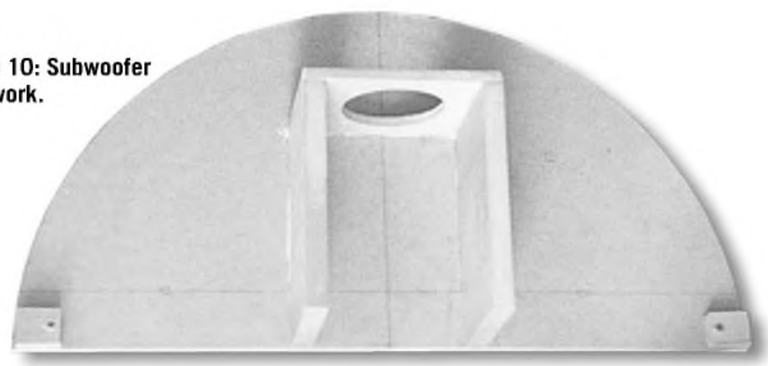
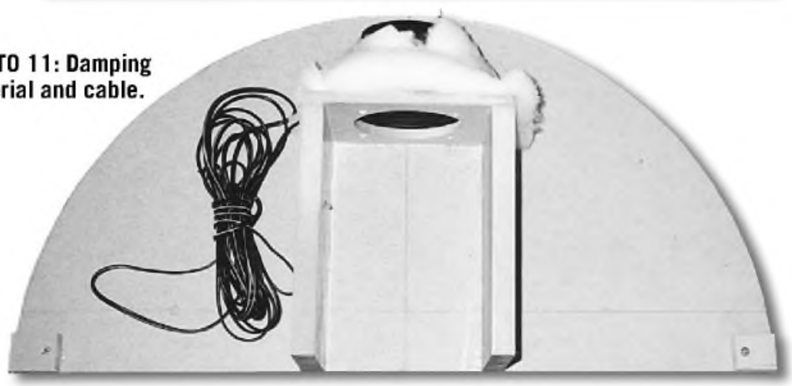


PHOTO 11: Damping material and cable.



This curved section, though only 6mm thick, is actually stronger than a flat board of 24mm (1″)! Trim the 3mm pieces so they are flush with the front. You can now attach the front part "B." You must place clamps through the open port holes. You should be able to get two in each of the large holes, and one in the smaller hole. Revert to screws only as a last desperate means of holding the board flat. I had no troubles.

Photo 13 shows the trimmed box, with edges rounded over and sanded. Seal the box with an MDF sealant or use an appropriate undercoat. Match the finish to whatever you used for the satellite speakers.

Give it a coat of matte black (*Photo 14*) and a light sanding. Once done, give it a coat of gloss (*Photo 15*). Using only gloss paint gives a shallow, shiny surface rather than a deep black. I chose gold webbing to finish the job. *Photo 16* shows the completed cabinet with ports and attenuator installed.

You can insert the ports using the same technique I detailed for the square sub.

You can prepare and paint the saved disc at the same time as the cabinet. The drawing "Attenuator Disc" (*Fig. 24*) shows the detail. Drill and countersink the two holes as shown. I

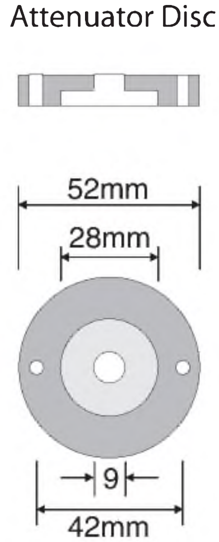


FIGURE 24: Attenuator disc.

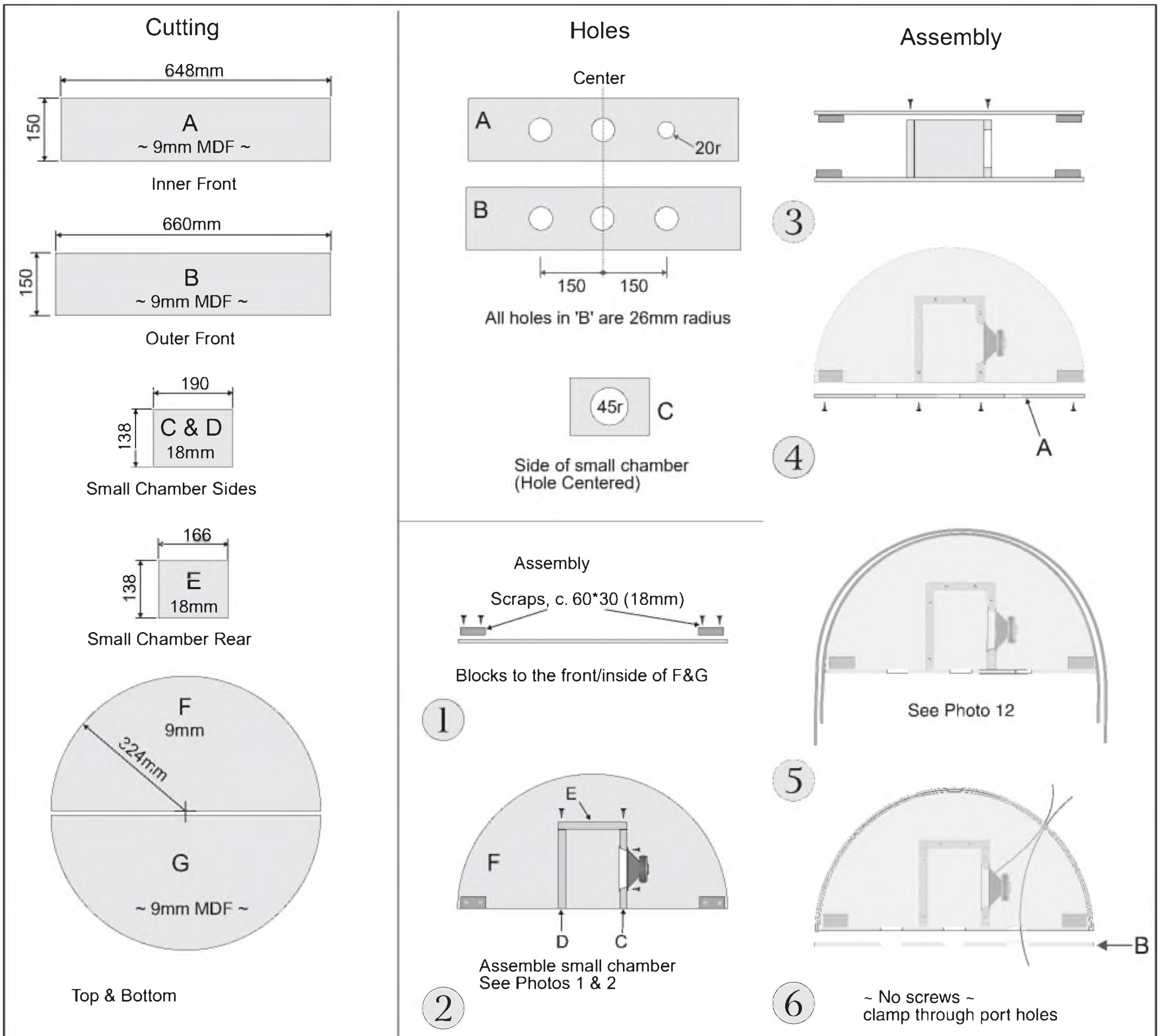


FIGURE 23: Cutting list.

used bronze screws—the same as used on the satellite speakers.

You will need a two-pole five-position "make-before-break" (shorting) rotary switch for the attenuator as described in part 5. The threaded shaft is usually too short to fit through the 9mm MDF thickness. To overcome this problem, cut a trench into the back of the disc as illustrated.

The final job is to run the cables. Drill a hole in the curved part on the same side as the speaker—it is probably best to drill the holes before painting.

Run a scrap of wire in and fish it out through the attenuator hole. Use a piece of stiff wire bent to make a hook. Do the same for the speaker's wire. Expose about 30mm of copper on both the scrap and the speaker wire, then twist them together. Pull the scrap so the speaker wire is through the rear of the cabinet.

You will complete the curved speaker with the rest of the electronics as described in part 5. You will run a twin shielded cable through the cabinet to the assembled attenuator.

In part 4 I will describe the design of the electronics, including the three-way active crossover with notch filter, pull down "T" attenuator, and five power amplifiers. ❖

PHOTO 12: Curving the speaker.

PHOTO 13: Trimmed and sanded.

PHOTO 14: Undercoating.

PHOTO 15: Gloss coat.

PHOTO 16: Finished speaker.

CABLES

I am a lifetime subscriber and submit this information for the possible interest of fellow readers. After reading an article in the British publication *Electronics World* on measurements of speaker cables, I added ferrites to all the cables in my hi-fi system, noticing some improvement in the sound.

Later I happened to be using a microwave meter, measuring possible leakage in my oven. Out of pure curiosity, I moved the meter to my amplifier, and to my enormous surprise, the meter pin registered at near maximum. This seemed very illogical to me.

I reread the article and then another about RF measurement, in addition to a book I have on electromagnetic capability (EMC). As a result, I began to suspect my cables. I then bought some low-noise microphone cables of oxygen-free copper at about one dollar per meter, plus four gold-plated RCA connectors. From these I made two unusual cables, which I used to connect my CD player to my amplifier.

The result was a general improvement with sweeter highs, better mids and lows, improved sense of information, and larger, more coherent image. When I again measured the amplifier with the microwave meter, the radiation had dropped almost to the minimum.

My old cables were 60 centimeters in length, which is also the wavelength of 500MHz, the half-wavelength being 250MHz and a quarter of 125MHz. The cable is connected in a hi-fi system but does not realize that "he" is acting as an antenna for these and many other frequencies, saturating the system with ugly noise. At least this is my theory of what was happening.

In my *Fig. 1* design, I use the two inner wires to transport the signal, and connect the shield to only one end through a 1k resistor. This "stopper" shunts any high frequencies arriving at the cable to ground.

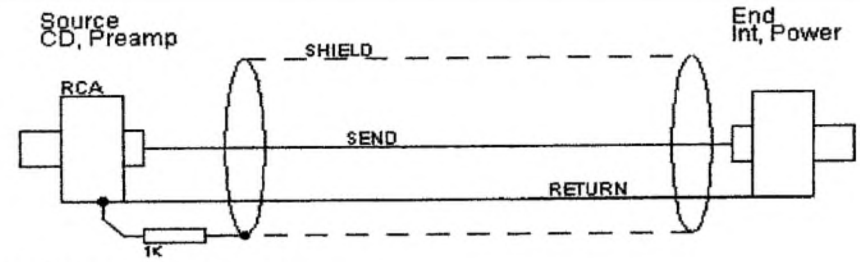


FIGURE 1: One-resistor cable.

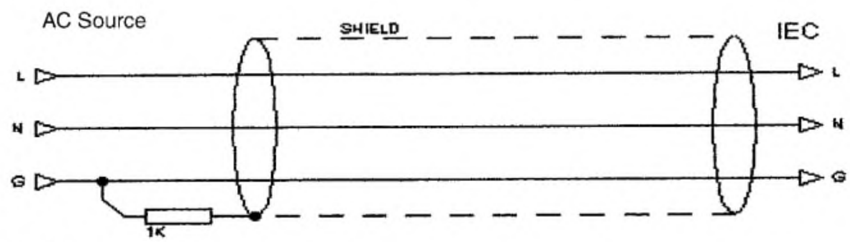


FIGURE 2: Power cable.

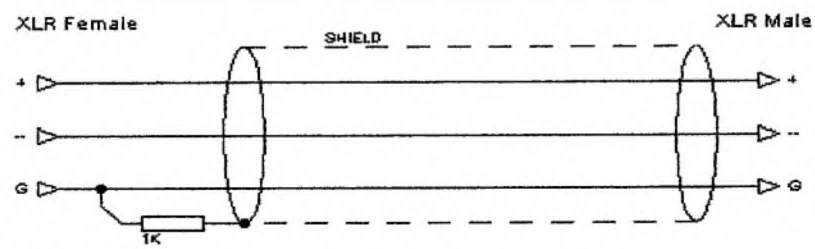


FIGURE 3: Balanced cable.

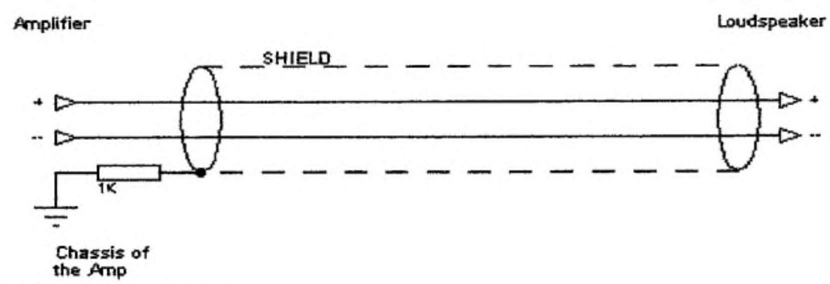


FIGURE 4: Proposed cable.

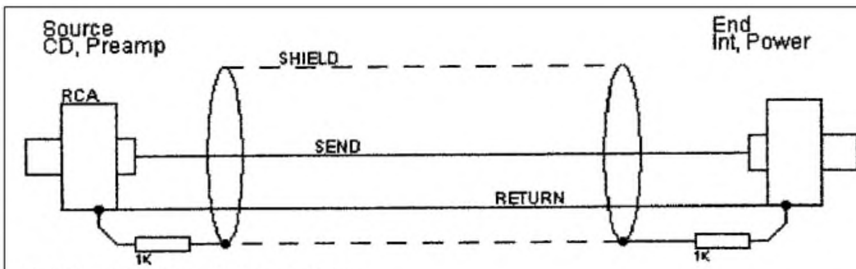


FIGURE 5: Cable with two resistors.

My theory about how this works may be crude, or even inaccurate. But it does provide a clue for me as to how the problem happens. It could also explain how some cables may be causing problems for many amplifiers. Connecting a dummy load equivalent to the cable to the amplifier may not cause any problem for it, but connecting the cable itself can destroy the amp because the cable acts as a giant antenna, causing the amp to dissipate large quantities of high frequency power, overheating the transistors.

Unfortunately, I have neither the instrumentation, the knowledge, nor the talent (not necessarily in that order) to develop this idea, but I suppose it could be refined further. My choice of a 1k resistor in *Figs. 2*, *3*, and *4* is totally empirical. I have experimented with a power cable as shown in *Fig. 2* and a balanced cable in *Fig. 3*.

I have constructed several cables for friends who have replaced their more expensive types with these. Although I have not experimentally proved this idea's usefulness in speaker cables, I believe they would look like *Fig. 4*. I have used a floating arrangement using two resistors to isolate the shield because the *Electronics World* article suggested that such a setup would have better frequency response. I retain, however, a healthy skepticism until I have tested the *Fig. 5* arrangement over a longer time.

I encourage other readers to test these proposals for themselves and share their findings with the staff of *audioXpress*. In all cases I used inexpensive cable with shields of 50 to 60% coverage. Perhaps cables of better quality and more complete shielding would produce even better results. I continue using ferrites on all cables, with good results.

Juan Raúl Couto Dominguez
La Coruña, Spain

I²S OUTPUTS

Kudos to *audioXpress*. At various times I've subscribed to all three of your magazines, but not at the same time. Now I won't miss anything.

Twice in past years, you've printed letters of mine in *Audio Amateur*, first in 4/86 about triode linearity and then in 2/91 about a constant-current Class-A bridge amplifier.

I have a C.E.C. belt-driven CD transport feeding an Audio Alchemy DTI Pro 32, which can interpolate missing bits when expanding the word length. To my ears, it works. Without it my system sounds mechanical.

It's my belief these devices didn't become more popular because the average user was unaware of the importance of cabling. I've experimented with Van den Hull, Apogee, C.E.C. (came with my transport), Illuminations D-75, and finally Illuminations D-60, all with BNC connectors. The difference between the best and worst cable is simply enormous! Unfortunately, the D-60 costs $260 ($330 list) for one meter, but if you want to extract music from the S/PDIF interface, that's what my ears tell me you need.

The DTI Pro 32 has an I²S output that I use with a Camelot Arthur DAC. The I²S connection sounds as good or a bit better than the S/PDIF connection with an Illuminations D-60, and the cable is almost free! Camelot threw in an extra I²S cable with the Arthur for nothing. Not only is I²S a more elegant way to transmit digital, it's more cost effective as well.

In the 6/00 issue of *Audio Electronics*, Gary Galo reviewed the excellent Assemblage 2.6 and 3.0 DACs. He said the Burr-Brown DF 1704 will accept an I²S signal directly (16 or 24 bits), unlike PMD-100.

What I want is an "I²S Mate." This would be only four ICs, not counting voltage regulators (*Fig. 6*). An LS device is used as a receiver. CMOS are too vulnerable to static electricity to use with a cable. My Arthur DAC uses a 74LS244 in this position.

I believe this device has a typical rise time of ten nanoseconds. This is good for an 11MHz masterclock and adequate for 96kHz. If the rise time is too fast, it will excite ringing in the filter (the last thing you want at this point in the circuitry). If it's too slow the data becomes corrupted.

The Burr-Brown ISO 150s provide easy isolation between the digital and analog ground planes. All of these chips are available from Digi-Key, including DF 1704.

I bought one only to discover the 28-

pin SSOP package is too small for me to make a circuit board. The pins are only .025″ apart! With care, I could probably solder it to a circuit board. This might sound like a project that few people could take advantage of, but that's not the way I see it.

The well-reviewed Perpetual Technologies PA-1 has the same 5-pin I²S output as the DTI Pro 32. Perpetual (as you probably know) is run by Mark Shifter of Audio Alchemy fame. The word length algorithm has been improved since the DTI Pro 32. Peter Madnic and Kieth Allsop are the brains behind it.

The new Assemblage D2D-1 sample rate converter also has a 5-pin I²S output (and a 2 picosecond jitter spec!). Last, but not least, the Camelot Dragon also has a 5-pin I²S output. I've seen DTI Pro 32s on the Net for $400. They have a 20 picosecond jitter spec and, in my opinion, with the right cabling, go a long way towards getting digital to sound like good analog. Sixteen bits just isn't enough.

I've been studying digital for a couple of years but am not qualified to design such a thing. Obviously the circuit board must be perfect and that's beyond me.

With a separate wire for the DEM connection, I believe this circuit can be done on a two-sided board (traces and ground plane). I want to use PCM 63 20 bit (bipolar DAC—not CMOS and you can disconnect the servo on the analog output) but others would probably opt for a PCM 1704 (24 bit).

Rick Bergman
Missoula, MT

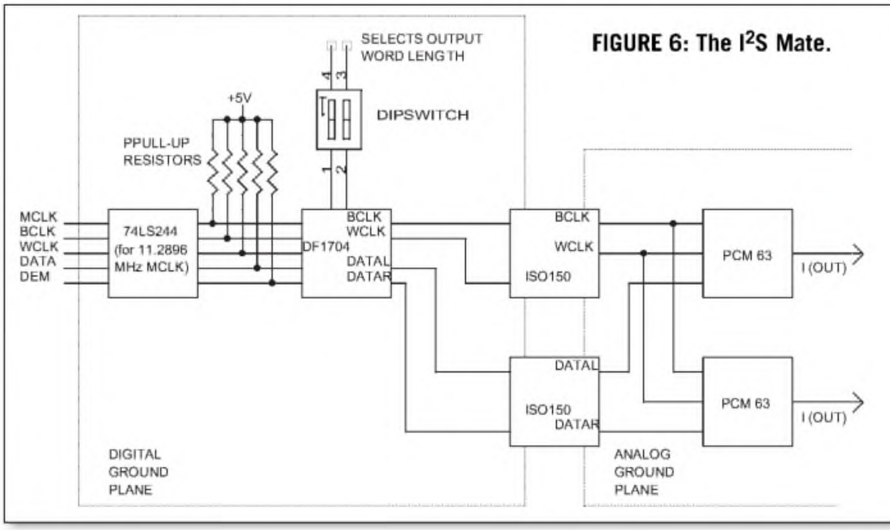


FIGURE 6: The I²S Mate.

SCOPE TESTING

I have a technical question about the testing procedures Charles Hansen uses in some of his equipment reviews. I also own a PICO ADC-216 virtual oscilloscope, but can't get it to reproduce the kinds of waveforms published in his articles at higher frequencies (e.g., the

10kHz square wave shown as Fig. 7 on p. 75 of the July 2001 issue). My attempts to capture these waveforms have been thwarted by problems with time base "jitter" and poor resolution.

The manufacturer pointed out that the relatively low sampling rate of the 216 would introduce the kinds of performance limits I was encountering. Yet Charles apparently has found a way to overcome them! I wonder how he does it?

Thanks in advance for passing this message along.

Joe Berry
Charleston, W.Va.

Charles Hansen responds:

I certainly don't do anything out of the ordinary with the ADC-216. It has a single-channel bandwidth of 166kHz at 333ksps. For square wave testing I use a time base of 20uS/div and usually use either Auto mode or 5V vertical setting. This gives about 2 square-wave cycles at 10kHz.

I also view everything on a 100MHz analog scope to avoid aliasing and to note any noise or spikes that may not be displayed on the ADC-216. There are about eight data points in the peak on the leading edge of each rising edge, so high-frequency ringing would not show up.

Are you sure you have Channel B turned off? Two-channel operation cuts the resolution in half. I set the triggering about 50mV positive to avoid zero-crossing jitters. Make sure you stop the sampling, or use single event triggering, before you try to save the waveform.

If you look at Fig. 5 of the Manley Stingray review (aX 1/01, p. 70), you can see even more detail on the 10kHz square wave. I haven't really had a problem with time base jitter or poor resolution, as long as the ADC-216 is used within its specified limits. The 8- and 12-bit DSOs may be faster, but they don't have the resolution of this 16-bit model.

There is always a trade-off between sample rate and bit depth.

TUBE TRAITS

Regarding the 5687 vs 6SN7 letter (July '01 *aX*, p. 87), I must say from experience that using the 5687 as a driver in my latest power amp designs has made an improvement in the sound, most notably in the bass frequencies. My circuit is a version of the classic Williamson design with the 5687 driving a 6SN7 direct-coupled cathode follower before the output tubes (6550/KT88 or 6BG6GA). My front-end stage is a 6CG7, which is electrically identical to a 6SN7. The 5687 gives a little more "drive" to the overall circuit.

Of course, the above circuit could be re-designed with many twin triodes, such as 12AU7s, all 6CG7s, and so on. A 6BL7 could be used in place of the 5687, or 6SN7, except for more heater current. A 7119/E182CC in place of the 5687 provides a little more voltage gain; a type 7044 gives about the same result.

I am pleased with your magazine; an excellent publication, to say the least.

Joseph K. Risher
Sounds Great! Enterprises
Stone Mountain, Ga.

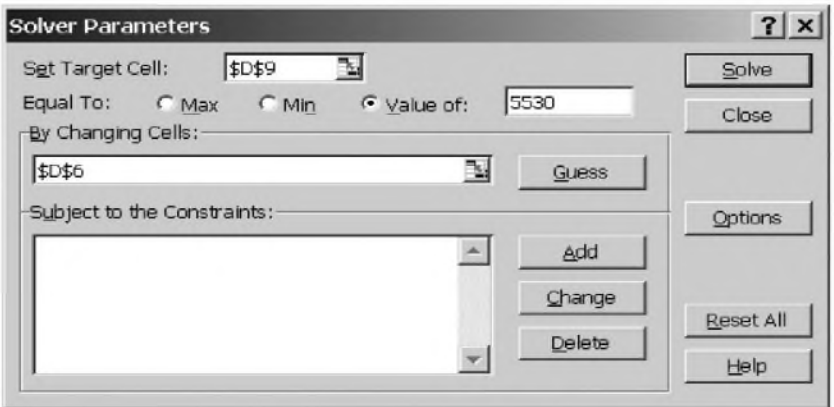
SOURCE

Readers who are interested in the "MOSFET Mod for the Dyna ST-120"

Determining Optimum Box Dimensions by Louis C. McClure
Speaker Builder, Two:2000, P. 42
Using Solver in Microsoft Excel 97/2000 to solve for box dimensions

Side	Dimension(in)
Depth	10.75563
Width	17.92605
Height	28.68168
Box Volume	5530.00000

1) Run Solver (From menu, Tools, Solver)
2) Set parameters as shown below:



Set Target Cell: D9, the total box volume that you are solving for.
Equal To: Value of: 5530, the value for the box volume in the example.
By Changing Cells: D6, the depth of the box.
Click the Solve button and accept the value

FIGURE 7

(*AE* 2/99) project should note that the obsolete transistors listed are available from Cricklewood Electronics, 40-42 Cricklewood Broadway, London NW2 3ET, UK, 020-84520161, FAX 020-82081441.

For example, the 2N-3821 is £3.00 each. They also stock many MPSU-XXX types; their specialty seems to be obsolescent international semiconductors. They even carry "veroboard" prototyping strip board (which even Digi-Key dropped from their catalog), now next to impossible to find.

They provide good service and fast shipping and phone orders. They advertise in *Electronics and Wireless World*, but would be welcome in *audioXpress*.

Andre Thivierge
Hull, Quebec

CALCULATIONS

To clarify some issues regarding my letter ("Xpress Mail," May '01 *aX*, p. 76):

1) The reason that D × W × H displayed in my spreadsheet doesn't equal 5530.000 on a calculator is that the displayed values are rounded off to three places, but the actual calculations behind the scenes are not rounded off.

2) The dimensions are different because I used a ratio of .618:

Width = Depth/.618
Height = Width/.618

Mr. McClure used:

Width = Depth/.600
Height = Width/.625

The attached Excel spreadsheet (*Fig. 7*) shows Mr. McClure's calculations. Sorry for the confusion. I'll consider this a lesson in effective communication.

Gary Manigian
Design Concepts
Martinsville, N.J.

A MOVING EXPERIENCE

I have two questions, but first, the reason for the questions! In Montreal July 1st is Moving Day (it is Canada Day, too, but when close to a quarter-million people are moving, you can safely say it's Moving Day). And when you move, there are a lot of things you don't want to bring along, so you leave them in the back of the house in the small streets—in French, we call them "ruelles." So on Moving Day many people will find treasures there they can use!

And this is where I found my treasure. In a large cardboard box there was a Leak Stereo 30, and below the amplifier, there was a Thorens turntable TD 124-II in fairly good shape, complete with the Thorens tonearm and a Shure M95ED cartridge without stylus.

So I need your help in obtaining the Leak stereo amplifier schematics. It uses AD 140, big germanium transistors. And maybe someone has a copy of the TD 124 service manual, which I would need, because I plan to take apart the turntable, clean it vigorously, and remount it, using proper oil. Bearings seem in good shape, but the oil is fairly old! This turntable was clearly made to last, so any help would be greatly appreciated.

Philippe Trolliet
Montréal
ph_trol@CAM.ORG

See Firsts in High Fidelity, *Stephen Spicer, p. 252 (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458-0876, (603) 924-6371, Toll-free: (888) 924-9465 (US/Canada), FAX (603) 924-9467, custserv@audioXpress.com, www.audioXpress.com).—Ed.*

HELP WANTED

I am looking for the following model of speaker, which is (apparently) very difficult to find.
Frequency response: 25Hz–>20.000Hz
SPL: 90dB (or more)
Diameter: 10″ (25.4cm)

Can you please help me find this? I've already been searching half the world. ❖

Sabine Dubois
Priminfo@skypro.be

The other half of the world is also looking for this same miracle speaker. I think they might have one on Mars.—Ed.

Onkyo Review
from page 73

The THD at higher input frequencies showed closely spaced excursions between 0.01% and 0.2% after processing through the DSP. I guess that this is due to interaction of the analog audio signal with the digital conversion sampling frequency.

The distortion curve for the subwoofer output at 40Hz shows fairly high distortion above 1.5V RMS. This curve continues to increase gradually with no hard clipping, and the LF sine waves became fatter in perspective as the subwoofer output level increased (somewhat tube-like in appearance, although I did not find a hidden 12AX7 among the PC boards).

I also applied a digital test signal from my CD player's coax output using the CBS-1 Test CD 997Hz linearity tracks. Its distortion curve is at the bottom of *Fig. 2*. The digital input THD versus frequency is also better than that of the analog inputs, especially above 1kHz. In the lower three curves you can see a curious bump in the distortion at about 2.5V RMS output. Since this appears in the direct-mode curve that theoretically bypasses all digital processing, it probably is not a DSP-related phenomenon.

The bass and treble processing is engaged by a pushbutton on the front panel. This control has a display-indicated range of ±12dB for each function. The bass control has an actual range of ±10.4dB at 100Hz. Similarly, the treble control range at 10kHz is ±10.5dB.

At 1kHz each control still has some effect: ±3.4dB for the bass and ±1.3dB for the treble. In a DSP-related phenomenon, the bass control still exerted ±3dB of control at 10kHz, while below 800Hz the treble actually increased its gain slightly when the display indicated −8dB to −12dB. The mute function caused a decrease in the output of −52dB.

Preamp square-wave performance in direct mode has the same excellent response as the phono stage. In stereo mode, with the subwoofer on, the 1kHz response (*Fig. 3*) exhibited the Gibbs Phenomenon ringing associated with the steep digital filters used in the DSP section of the receiver. For the same reason, the 10kHz square wave was rounded over into nearly a sinewave shape.

The Onkyo's reproduction of a CCIF-combined 19kHz + 20kHz intermodulation distortion (IMD) signal at 12V p-p into 10kΩ produced a low 1kHz IMD product of 0.004%. Repeating the test with a multi-tone IMD signal (9kHz + 10.05kHz + 20kHz, not shown) resulted in a 1kHz product of 0.003%. The CCIF performance in stereo mode was 0.006%.

MEASUREMENTS—POWER AMPLIFIER

There are no straight connections to the power amplifier inputs, so their performance is a function of the preamp as well as the listening mode. Since I explored the various modes in the preamp tests, I used only the direct mode during the power amplifier testing. The TX-DS787 is not designed for speaker loads less than 6Ω, so I performed all these tests using my 8Ω loads.

I ran the two front channels at 10W for one hour into an 8Ω resistive load. The left-channel preamp/power amp combination had slightly higher distortion than the right (just the opposite of the preamps alone), so I present its measurements here and summarize them in *Table 1*.

The TX-DS787 power amplifiers do not invert polarity. The maximum ("80") overall gain at 2.83V RMS output into 8Ω was 43.5dB. Using the "Ref" volume-control level, the gain was a more normal 26.6dB. The output impedance at 1kHz measured 0.11Ω, rising to 0.25Ω at 20kHz. The amplifier proved insensitive to frequency-related changes in speaker impedance using an IHF speaker load.

The frequency response for the TX-DS787 was within ±1dB from 4Hz–54kHz, at an output of 2.83V RMS at 1kHz into 8Ω. It was down −3dB at 3Hz and 94kHz. The curve was smooth and flat, with no evidence of high-frequency peaking.

Crosstalk performance for the phono stage, preamp, and power amp are shown in *Table 2*. Power amplifier crosstalk is a function of the preamp section. The same is true for the phono stage, whose output signal is not directly available.

THD+N versus frequency with 1W into an 8Ω load did not exceed 0.031% from 20Hz–20kHz. With a complex load of 8Ω paralleled with 2μF, the THD+N increased from 0.035% at 8kHz to 0.083% at 20kHz. The CCIF IMD performance was very good at only 0.008%.

Increasing the output power to 10W per channel dropped the distortion even lower, never exceeding 0.017% over the audio band. This fine performance was maintained out to 2/3 rated load (67W per channel) into 8Ω. The cooling fan began cycling at about 7.5W per channel, and was operating continuously above 20W per channel. This fan is very quiet, so you would not hear it in your listening room.

Figure 4 shows THD+N versus output power into 8Ω at 20kHz (top), 20Hz (middle), and 1kHz (bottom). I engaged the test-set 80kHz low-pass filter to limit the out-of-band noise. There was absolutely no strain right up to the point of maximum power, and the amplifier seemed quite happy to continue at this load all day long.

There is a curious bump in the distortion at 30W in the 1kHz curve. This may correspond to the bump in the preamp section distortion at 2.5V output (*Fig. 2*). I checked this response anomaly a number of times to be sure it was real, and it was.

Using a 1kHz signal, I achieved 142W per channel into 8Ω at 1% THD+N. 128W per channel was available at 20kHz, and 138W at 20Hz. I repeated the test driving only a single front channel. Here I saw 157W, 153W, and 149W, respectively. This is pretty respectable performance for amplifiers rated 100W into 8Ω.

The distortion waveform for 10W into 8Ω at 1kHz is shown in *Fig. 5*. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the THD test-set notch filter), not to scale. This 0.013% distortion residual signal shows mainly the third harmonic, along with some high-frequency noise.

The spectrum of a 50Hz sine wave at 10W into 8Ω is shown in *Fig. 6*, from zero to 1.3kHz. The THD+N measured 0.015%, and the highest harmonic is the third at −81dB. The next highest second harmonic lies just below −93dB.

The 2.5V p-p square wave into 8Ω at 40Hz showed only a slight tilt. The 1kHz square wave was perfect, while the leading edge of the 10kHz square wave showed a very slight rounding. When I connected 2μF in parallel with the 8Ω load, there was some ringing evident, as you can see in the 10kHz response in *Fig. 7*.

MEASUREMENTS—FM TUNER

I did not run any tests on the AM section of the receiver, except to make sure it was functional. The FM stereo section was quite sensitive. I could find a station near or far, across almost the entire 88–108MHz FM band. My FM stereo signal generator is set to 98.1MHz, since there is no station at that frequency in my area. However, in FM/mute mode (mono), the TX-DS787 managed to find one about 85 miles away, though barely listenable.

Frequency response in direct mode measured +0.1/−1dB from 20Hz–15kHz, and maintained normal polarity (audio signal to the FM modulator compared to the line output of the tuner).

With a 75Ω RF signal at the antenna F-input jack, stereo sensitivity (full quieting) was 36dBf, below which the tuner went into muting. Stereo separation at 1kHz was 48dB. I needed to switch to FM/mute mode to measure mono sensitivity at 17dBf, and usable sensitivity was about 11dBf, which is fairly good. ❖

Classifieds

VENDORS

AD INDEX

Yard Sale

FOR SALE

Audio Research M-100s, boxes and papers like new, $3,400 or BO; Classe DR3-VHC, $2,000 or BO; California Audio Labs Aria CD player, $800 or BO; Rega Planet CD player, $550 or BO; Musical Fidelity x-10, $50. Thomas Cox, 304-843-0359.

Seven Scanspeak 18W8545s carbonfiber, three Kevlar 18W8546s, $90 each; four Dynaudio D52af, $75 each; five D54af, $85 each; D54 with horn lens, $85. All drivers are new, tested for purity. Insured shipping included. Write Daryl Jensen, PO Box 373, Nevada City, CA 95959.

Pair Dynaco MK III 60W monoblock tube amplifiers, fully operational. Upgraded I/O connectors, capacitors, resistors, wiring. 6550 output tubes, good cosmetics. Dan, 317-299-6024, fastcat95@juno.com.

PSB Speakers—Stratus Bronze: 2½ way towers with black lacquer finish, 150W @ 4Ω, bi-wire capable, spiked feet, two years old, and in great condition—original boxes and instructions, $500 + shipping. Davis Berglund, 281-347-6157.

WANTED

Hobbyist seeking pre-recorded 2-track reel-to-reel tapes—7.5 or 15 ips. Can be 1950s/1960s commercial product or old studio dubs. Andy Pennella, 203-329-7498 or andrew.pennella@pfizer.com.

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