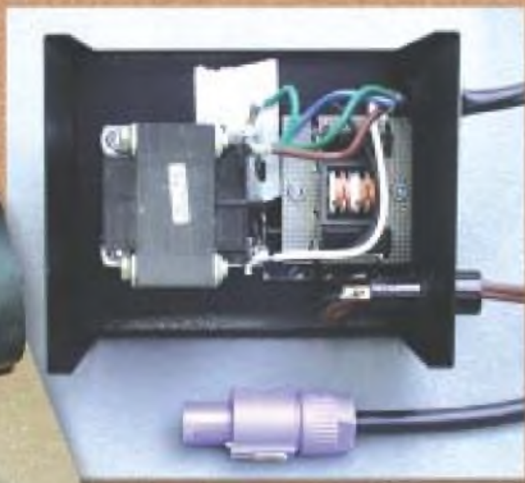
AN INSIDER'S LOOK AT VACUUM TUBES

audioXpress

THE AUDIO TECHNOLOGY AUTHORITY

NOVEMBER 2003
US $7.00 Canada $10.00

Continuing *Audio Electronics, Glass Audio & Speaker Builder*



A Purist Approach To Preamp Re-Design

How to Achieve Clean, Accurate Bass

Burst-Testing Your Amps

Add a Little "Flare" To Your Speaker Ports

Testing a DIY Two-Way Speaker

Rock On! With the Restored Peavey Bass Guitar Amp

www.audioXpress.com

0 74470 84697 0 11


CARDAS
CARDAS
Quad Eutectic Solder


Never Build Another Speaker Cabinet
Sit back and enjoy the music
Premium quality finished cabinets available in three real wood veneer finishes and six different sizes. Baffles are removable and blank providing limitless possibilities for custom loudspeaker designs. Complete with grills, hardware, and veneered backs.
To see our complete line of finished cabinets go to www.partsexpress.com/cabinets
Parts Express 725 Pleasant Valley Dr., Springboro, OH 45066-1158 Phone: 1-800-338-0531 ◆ FAX: 937/743-1677 E-Mail: sales@partsexpress.com
Source Code: AXM

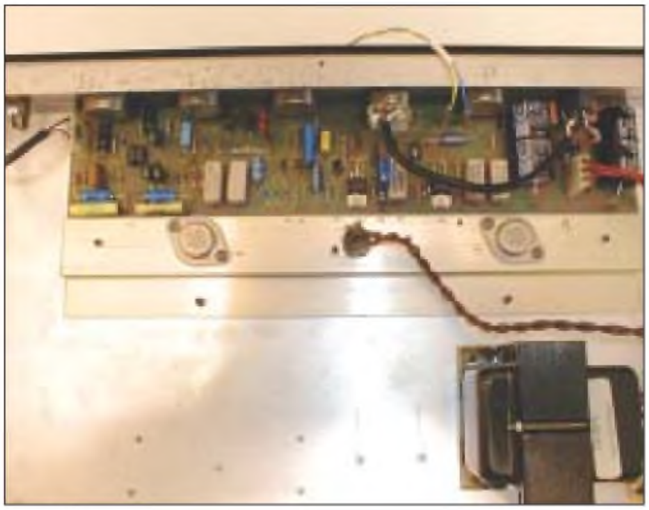
audioXPRESS

CONTENTS

VOLUME 34 NUMBER 11 NOVEMBER 2003

FEATURES

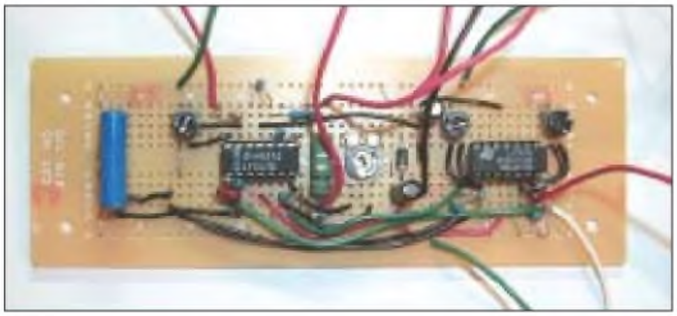
page 50



page 40

audioXpress (US ISSN 0004-7546) is published monthly, at $34.95 per year, $58.95 for two years. Canada add $12 per year; overseas rates $59.95 per year, $110 for two years; by Audio Amateur Inc., Edward T. Dell, Jr., President, at 305 Union St., PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH, and additional mailing offices.

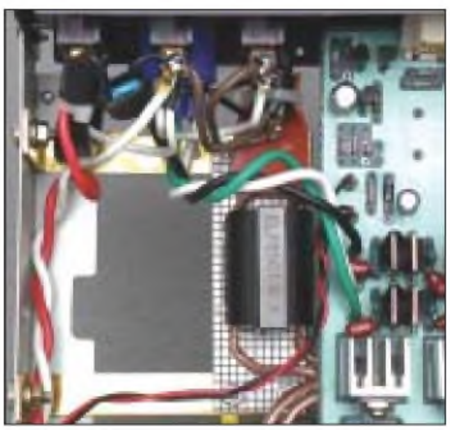
POSTMASTER: Send address changes to: *audioXpress*, PO Box 876, Peterborough, NH 03458-0876.



page 18

DEPARTMENTS

IN EVERY ISSUE



page 6

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

THE STAFF

Editor and Publisher
Edward T. Dell, Jr.

Regular Contributors

Eric Barbour	Erno Borbely
Robert Bullock	Richard Campbell
Joseph D'Appolito	Vance Dickason
Bill Fitzmaurice	Gary Galo
Charles Hansen	G.R. Koonce
James Moriyasu	Nelson Pass
Richard Pierce	Reg Williamson

Vice President
Karen Hebert

Dennis Brisson	Assistant Publisher
Amity Palmer	Editorial Assistant
Tina Hoppock	Graphics Director
Amy Wyman	Production Assistant
Laurel Humphrey	Marketing Director
Kelly Bennett	Customer Service

Advertising Department
Strategic Media Marketing
1187 Washington St.
Gloucester, MA 01930
Peter Wostrel
Phone: 978-281-7708
Fax: 978-283-4372
E-mail: peterw1@gis.net
Nancy Vernazzaro
Advertising/Account Coordinator

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

SUBSCRIPTION/CUSTOMER SERVICE INQUIRIES
A one-year subscription costs $34.95; a two year subscription costs $58.95. Canada please add $12 per year. Overseas rate is $59.95 for one year; $110 for two years. To subscribe, renew or change address write to the Customer Service Department (PO Box 876, Peterborough, NH 03458-0876) or telephone toll-free (US/Canada only) 888-924-9465 or (603) 924-9464 or FAX (603) 924-9467. **E-mail: custserv@audioXpress.com. Or online at www.audioXpress.com**

For gift subscriptions please include gift recipient's name and your own, with remittance. A gift card will be sent.

EDITORIAL INQUIRIES
Send editorial correspondence and manuscripts to *audioXpress*, Editorial Dept., PO Box 876, Peterborough, NH 03458-0876. **E-mail: editorial@audioXpress.com.** No responsibility is assumed for unsolicited manuscripts. Include a self-addressed envelope with return postage. The staff ***will not*** answer technical queries by telephone.

CLASSIFIEDS & WEB LISTINGS
Contact Nancy Vernazzaro, Advertising Department, *audioXpress*, PO Box 876, Peterborough, NH 03458, 603-924-7292, FAX 603-924-6230,
E-mail nancy@audioXpress.com.

SUBWOOFER DRIVER FROM BULLY SUBWOOFERS

Bully Subwoofers announces the production of a new 15″ subwoofer driver, the "Bully-15." It is designed for DIY, OEM, and Custom Theater applications. It features a 3″ diameter voice coil for superior power handling; aluminum shorting ring for lower distortion, lower inductance, and excellent heat dissipation; as well as a finely tuned motor structure and suspension for extended linear excursion. Get the full details at www.BullySubs.com.

BOSTON AUDIO SOCIETY TEST CD

The Boston Audio Society Test CD-1 contains 71 minutes of contributions by members. It begins with a setting of Lewis Carroll's "Jabberwocky" for chorus; then selections by Saint Saens (Organ Symphony), Bach, Bruckner, Soler, Schumann, Mahler, and Verdi (Requiem); an excerpt from Shoptalk (the radio show which led to the BAS); 21 tracks of test signals which test the limits of CD players and playback systems and your hearing; and a selection from the Opus "Frogs." The complete booklet is posted on the BAS website. If you wish to order a copy, contact the Boston Audio Society through David Hadaway, PO Box 460, Rindge, NH 03461.

EUROPEAN TRIODE FESTIVAL 2003

The fourth European Triode Festival (ETF.03) will take place in Germany from Thursday, Dec. 4–Sunday, Dec. 7. The festival, a gathering of tube audio hobbyists and professionals, will host participants from all over the world. The official language at the show is English, and the lectures and workshops will cover topics of interest to the triode lover and amp builder. The event will take place in the holiday village of Langenargen, near the border of Germany and Switzerland, on the shore of Lake Constance. For further information, go to www.triodefestival.net, or, for reservations, e-mail Wolfgang Braun at wb@braunbaustoffe.de.

PROFESSIONAL MOVING MAGNET PREAMP

Esoteric Sound introduces the Professional Stereo Moving Magnet Preamp, which is designed to convert any AUX input to phono use in home, broadcast, and studio applications. This device will make it possible to use a line level input on any amplifier or mixing console for playback with a turntable that is equipped with a conventional, moving magnet phono cartridge. Typical input sensitivities of such inputs are roughly 200mV, and the preamp boosts the level of the magnetic cartridge and also applies "record compensation" in the form of the RIAA curve—this adjusts the relative levels of bass and treble to match that of modern records. For more, contact Esoteric Sound, 4813 Wallbank Ave., Downers Grove, IL 60515, (630) 960-9137, www.esotericsound.com, e-mail: EsotericTT@aol.com.

Adcom's GFP-565 Preamplifier, Part 1

This author's purist approach improves upon the design of a classic preamp. **By Gary Galo, Regular Contributor**

In the Feb. 1990 issue of *Stereophile*, I reviewed Adcom's then new GFP-565 preamplifier, one of the products issued during the tenure of C. Victor Campos as Adcom's Director of Product Development (clearly a "golden-age" for the company; in 1995 he became Adcom's Vice President for Product Development).[1] Under Victor's direction, Walt Jung was brought into the design team for this IC op-amp-based preamp.

In my extremely positive review, I made some rather bold statements regarding modifications to the preamp. In a nutshell, I noted that the GFP-565 was a thoroughly engineered product that probably wouldn't benefit from the usual parts substitutions. I actually warned potential tweakers of the possibility of making the preamp worse, unless modifiers knew exactly what they were doing.

It is no secret to readers of *aX*—as well as its predecessors, *Audio Electronics*, *Audio Amateur*, and *Speaker Builder*—that I began tweaking my 565 preamp within a year of the publication of my review. Indeed, over the years I have made numerous references to my "extensively modified" Adcom GFP-565 in these pages. Many readers of these magazines have asked for details on these modifications. Some have even asked, "What gives? I thought you said it couldn't be modified!" Have I contradicted myself? Not really. I still believe that, at the time I wrote the review, substituting your favorite capacitor or op amp would probably have degraded the preamp.

The GFP-565 was an excellent product, especially for the price, when it was introduced. But, my goal has been to raise the level of performance to something well beyond that of a "best under-$1000 commercial preamp." The GFP-565 first appeared in the fall of 1989—14 years have now passed since its introduction, and much has happened in the interim. An article describing possible upgrades is probably long overdue.

MODIFICATIONS

The modifications I'll describe amount to far more than parts substitutions. They involve wholesale replacement of most of the original circuitry, to the point where the modified preamp is really an entirely new design in the Adcom GFA-565 case. The original Adcom case and PC board are quite versatile.

With some ingenuity you can change most of the circuitry, yet still retain the attractive and functional case design. The new preamp is a "purist" design, eliminating the tone controls, high filter, loudness contour, processor loop, and headphone amplifier. If you find these functions necessary, this may not be the right preamp for you.

I have divided the project into three parts. Parts 1 and 2 will involve complete replacement of the power supply, adapting Walt Jung's "Improved Positive/Negative Regulator" specifically for this project.[2] This regulator is a substantially improved version of the one described in the 1995 series "Regulators for High-Performance Audio," Parts 1 through 4.[3] Part 3 of the 565 mod will cover the new line stage, and Part 4 will describe a new phono preamplifier with DC-servo control.

I spent some time wrestling with the

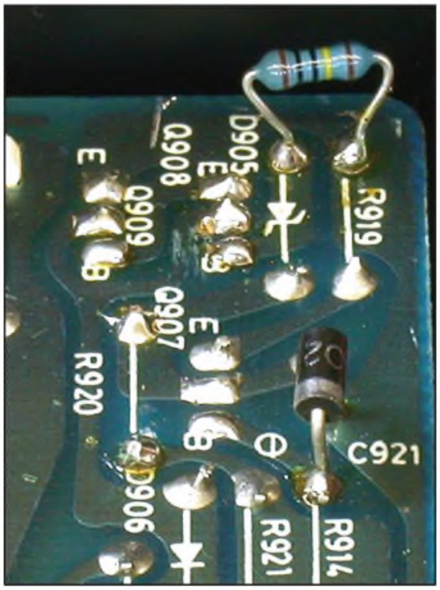


PHOTO 1: Delay-circuit mod components D916 and R922 are soldered to the bottom of the Adcom PC board.

order of presentation. Much of the new circuitry in the line stage and phono preamplifier was developed after my initial replacement of the original Adcom power supply. I had no idea how the new circuitry would perform—or if it would even be stable—with the original supply. For this reason alone, the power-supply modifications *must* come first.

The power-supply mods are also the most difficult and tedious part of the entire project, perhaps sufficiently so to discourage some builders. I believe it is better to put the hardest part out of the way first, rather than lead readers through three parts of the project only to have them find that the final installment is more than they had bargained for.

In order to begin these modifications, you'll need to purchase a GFP-565 service manual, which is mandatory; I don't recommend attempting to modify this preamp without consulting it. You'll need to refer to the original schematic and PC board layout many times during the course of these mods. You can purchase the GFP-565 service manual from Adcom's parts department (see Sources for their e-mail address). If you have difficulty obtaining a service manual, e-mail me at galoga@potsdam.edu.

Of course, you'll also need a GFP-565 preamp to modify. The 565 was an ex-

Behind the Scene

Dr. Joseph D'Appolito has been working as consultant for Usher Audio since early 2000. A world renown authority in audio and acoustics, Dr. D'Appolito holds BEE, SMEE, EE and Ph.D. degrees from RPI, MIT and the University of Massachusetts, and has published over 30 journal and conference papers. His most popular and influential brain child, however, has to be the MTM loudspeaker geometry, commonly known as the "D'Appolito Configuration," which is now used by dozens of manufacturers throughout Europe and North America.

Dr. D'Appolito designs crossover, specifies cabinet design, and tests prototype drivers for Usher Audio, all from his private lab in Boulder, Colorado. Although consulting to a couple of other companies, Dr. D'Appolito especially enjoys working with Usher Audio and always finds the tremendous value Usher Audio products represent a delightful surprise in today's High End audio world.

With an abundance of original concepts in loudspeaker design, backed by thirty years experience in manufacturing and matched with an eye for fashion and unparalleled attention to detail, is USHER the ideal original design manufacturer you've always been looking for? Find out the answer today by talking to an USHER representative.

USHER

USHER AUDIO TECHNOLOGY
67, Kai-Fong Street, Sec.1, Taipei 100, Taiwan
Tel: 886 2 23816299 Fax: 886 2 23711053
Web site: www.usheraudio.com E-mail: usher@ms11.hinet.net

tremely popular product, and used units do turn up. I suggest trying Ebay (www.ebay.com) for starters, as well as Adcom dealers (you'll find a list of Adcom dealers on their website). I recently saw two 565 preamps on Ebay for "Buy it Now" prices of less than $300.

A thorough familiarity with the Jung regulator design is necessary for the successful completion of Part 1. Walt's "Improved Positive/Negative Regulator" article is required reading. The background described in the four-part series, "Regulators for High-Performance Audio," is also extremely important, particularly Parts 3 and 4 by Jan Didden and myself, respectively. Back issues of *Audio Electronics* and *Audio Amateur* are available on the *aX* website, www.audioXpress.com. If you intend to pursue this project, you'll need to order the back issues now.

Adcom does not endorse these modifications, and any changes to the preamp will void your warranty. Since the GFP-565 preamp was discontinued several years ago, this is probably not an issue, but don't expect any help from Adcom or your dealer once you've begun this project. The modifications described in these articles are for advanced builders with sufficient technical background and test equipment to troubleshoot and fix problems if they occur.

PRELIMINARY STEPS

Before beginning the power-supply replacement, some preliminary steps are necessary. Here and throughout this project, you'll need to remove components on the Adcom PC board. I suggest a good desoldering tool, such as Radio Shack's cost-effective 64-2060.

- Check the main supply rails to make sure the regulators are at ±17.5V DC. Some early samples of the GFP-565 had the regulators set at ±19V, higher than the absolute maximum ratings of the IC op amps used in the preamp. In these preamps R908 is 15k. The rail voltages can be lowered to ±17.5V by changing R908 to 12.7k, the value used in most production samples of the GFP-565. I suggest doing this even though the main regulators won't be used for too long.

Early on in my work on the GFP-565, Walt Jung suggested a couple of improvements to the circuit that controls the output muting relays. I also opted to increase the turn-on delay to 20 seconds. Refer to *Fig. 1* for the following steps:

- Change R920 to 332k. This increases the turn-on delay.
- Add a 1M resistor from the base of Q908 to the −15V supply. This is to drain leakage from the Darlington pair Q908/Q909. You can solder this resistor to the bottom side of the PC board (*Photo 1*).
- Add a 1N4002 diode from the base of Q907 to the −15V supply. The cathode of the diode connects to Q907 and the anode to −15V. This protects the emitter/base junction of Q907 from reverse voltage during power-down. You can also solder this diode to the bottom side of the PC board (*Photo 1*).

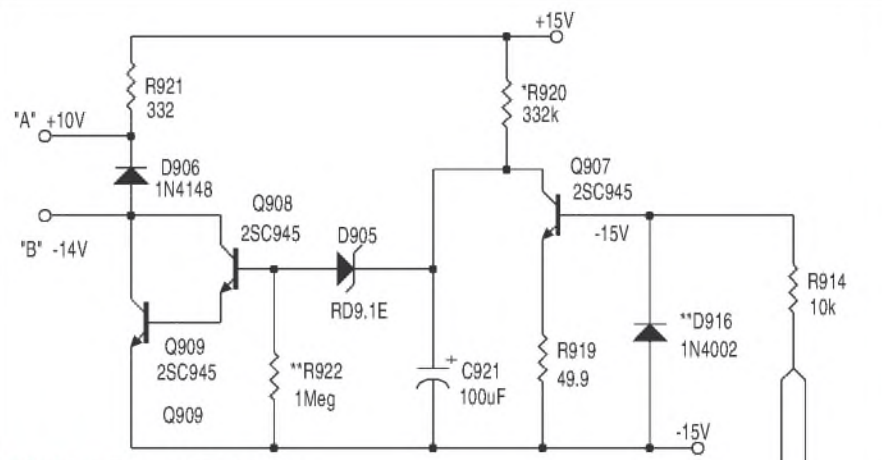


FIGURE 1: The modified delay circuit for the muting relays, showing the addition of R922 and D916.

Turn on the preamp and check the voltage at the base of Q907: it should be −15V. Verify correct operation of the muting relays.

Adcom lowered the impedance of the power-supply buses by paralleling them with brass rails on the component side of the PC board. But, they left two weak links in the supply distribution chain: jumpers J51 and J52, which are 22AWG wire. This is where the supply rails branch off the main buses to power the line stage. Here's an easy fix (*Photo 2*):

- Remove J51 and J52 on the main PC board. Replace these with two short lengths of 16AWG insulated solid bus wire soldered to the bottom side of the board. Bend the bus wire to form a right-angled "U." You could leave the old jumpers in place, but it will be more difficult to solder the new jumpers to the PC board with the old ones protruding through the board.

NEW RCA CONNECTORS

Adcom used cost-effective gold-plated, PC-mount RCA connectors on the GFP-565. The modified preamp is refined enough that differences between input and output connectors should be readily audible. Adcom followed normal industry practice by putting left-channel jacks on the top row and right-channel jacks on the bottom. Unfortunately, it is difficult to fit insulated, chassis-mount jacks in the bottom holes—the PC board obstructs the insulators, and the mounting nuts and/or ground washers may make accidental contact with the PC board ground plane.

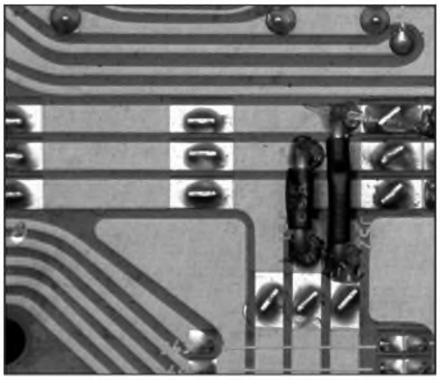


PHOTO 2: New 16AWG jumpers for the main power-supply buses. These jumpers replace a weak link in Adcom's power-supply distribution.

NEW !!

DH LABS

SILVER SONIC™

HIGH PERFORMANCE AUDIO CABLES

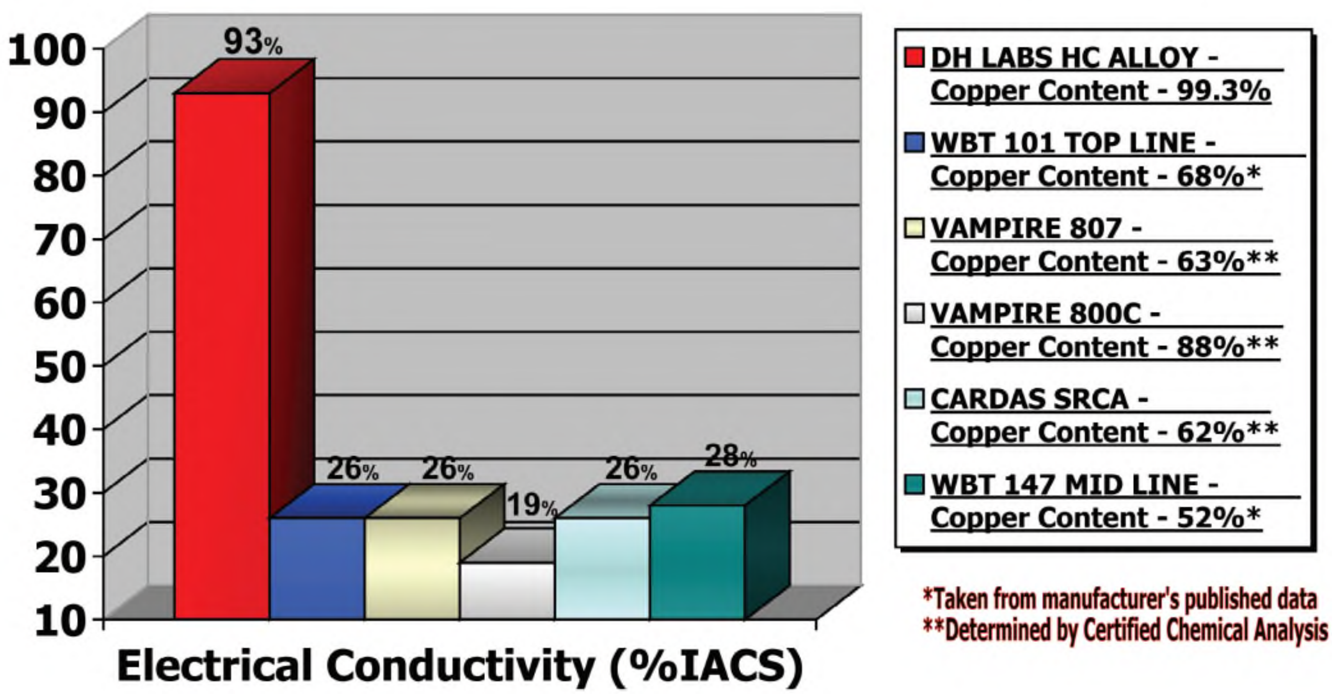
The Ultimate RCA Connector

DH Labs "High Copper Alloy"

Virtually all of the RCA Connectors used by today's top cable manufacturers are made from brass. There are a number of different brass alloys, generally featuring a combination of copper (around 60%) and varying amounts of zinc, lead, and even tin. The presence of these other metals greatly reduces the manufacturing cost and conductivity of the resulting alloy.

The conductivity of brass alloys is only about 26% that of pure copper. This loss (almost 75%), directly affects the performance of a terminated cable.

Our exclusive *"High Copper Alloy"*™ material is sourced from the U.S.A. for purity, and composed of more than 99% pure copper. Simply stated, this is the finest connector anyone has ever made....



*Taken from manufacturer's published data
**Determined by Certified Chemical Analysis

DH LABS
612 N. Orange Ave. #A-2
Jupiter, FL 33458 **(561) 745-6406**

www.silversonic.com

Fortunately, my purist approach to this project allows the removal of some unnecessary connectors, specifically the processor loop inputs and outputs, along with the normal and lab outputs. This will allow space for replacement of the critical RCA connectors: the CD and phono inputs, plus the bypass outputs.

There are many excellent chassis-mount RCA jacks available with gold plating and Teflon insulation. I've listed a few in the Parts List. One of the best-kept secrets in high-end audio construction is the jacks sold by MCM Electronics for only $1.69 each. These are outside-mount, so the nut goes on the inside, which is a bit inconvenient.

It's also a little tricky to grip the outside of the connector with pliers to prevent turning while you tighten the nut. But, if installed properly, their performance will belie the cost, and the insulating washers fit the 1/8″ Adcom holes. I suggest applying Caig Pro-Gold GP5S-6 contact conditioner to the ground washer and nut before installation to ensure a good long-term connection (do this with any connector that has a separate ground washer).

I use the Connex 53451 house-brand connectors sold by the Parts Connexion in my own preamps. These are inside-mount, outside-nut, so they are easy to install. The insulating washers require a 1/2″ hole, so you must enlarge the Adcom holes. The Vampire M2F is similar to the Connex, and also requires a 1/2″ hole. I suggest a Greenlee 1/2″ type 730 round knockout chassis punch (Mouser #586-3801) for this purpose. Let's start by replacing the output jacks.

- Remove the rear panel, which allows access to the rear panel jacks. You'll need to remove all of the jack assembly mounting screws in order to loosen the rear panel.
- Remove the processor loop input and output jack assembly. Unsolder the four signal and two ground leads from the PC board and remove the block.
- Remove the output jack assembly, which consists of six RCA connectors in one block. Unsolder the six signal and three ground leads and remove the block.
- Remove the phono and CD input jack assembly. Unsolder the four signal and two ground leads from the PC board and remove the block.

The rear corner of the PC board is now hanging loose, so retain one pair of jacks from the output assembly to provide proper support (*Photo 3*).

- With a band saw or coping saw, separate the left-most pair of connectors (looking at the back of the jack assembly) from the rest, carefully cutting around the mounting screw hole so it is retained. This is the "norm" out jack pair. If you do this right, you'll still have plenty of plastic around the mounting screw hole, while providing adequate clearance for the new RCA jack insulators.

You might want to do a "dry fit" before soldering everything in place. Re-solder the jack-assembly connections to the bottom of the PC board. Then, reinstall the rear panel, including all of the mounting screws for the remaining chassis-mount jacks. The "norm" out jack assembly has no electrical function in the new preamp—its only purpose is to provide proper support for the corner of the PC board.

- Mount the new RCA output connectors in the holes previously occupied by the left lab and bypass connectors, enlarging the holes if necessary. The ground washer tabs should face the PC board. The connector in the lab hole will be the right channel, so put a red-banded connector here if yours are color-coded.
- Solder bare hook-up wire from the ground washers to the ground holes in the PC board. I suggest using stripped D.H. Labs OFH-20 hookup wire, or a stripped conductor from their BL-1 interconnect cable.
- Solder insulated hook-up wire from the center pins of the new jacks to the left and right bypass out holes in the PC board. Again, I suggest D.H. OFH-20 Labs hookup wire, or the center conductors from their BL-1 interconnect cable. The old lab out holes in the PC board are no longer used.

In Part 3, covering the new line-stage circuitry, I'll offer the option of adding a pair of auxiliary output jacks, electrically identical to the main outputs, for passive biamping such as I described in *Speaker Builder* 2/92.[4] Active biamping may also benefit from the auxiliary outputs. If you are building your own active crossover, you can probably eliminate the input buffer and drive the high- and low-pass sections directly from the preamplifier's main and aux outputs. If you need the aux outputs:

- Mount another pair of RCA connectors in the holes previously occupied by the left processor out and in jacks. These will be the right and left outputs, respectively.
- Solder bare hookup wire from the ground washers to the ground holes on the PC board.
- Solder insulated hookup wire from the center pins of the new jacks to the left and right processor out holes on the PC board. These jacks will not be functional until Part 3 is completed.

Photo 3 shows the new output connectors installed in the 565 preamp chassis and wired to the PC board.

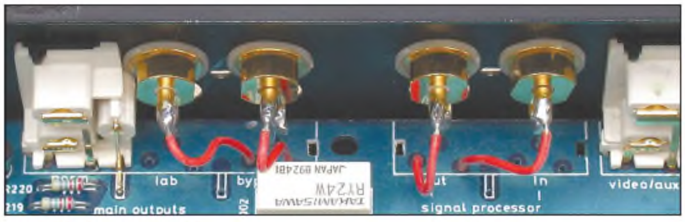


PHOTO 3: New RCA output connectors. The pair on the right are an optional set of aux outputs that may be useful for biamping. The "norm" output jack block, cut away from the original three-pair assembly, has no electrical function, but it supports the corner of the PC board.

PHOTO 4: New CD input jacks are shown on the left, along with new phono jacks on the right. The twisted-pair leads to the left phono input jack are soldered to the bottom of the PC board.

Now install the new CD and phono input jacks (*Photo 4*).

- Install the new CD input jacks in the holes previously occupied by the left CD and left phono jacks. Enlarge the holes with a punch, if needed.
- Solder bare hookup wire between the ground washers, and solder a single piece of insulated hookup wire from the bare wire to the CD ground hole on the PC board.
- Solder insulated hookup wires between the center pins of the new CD jacks to the CD input holes on the PC board.
- Punch two new holes for the new phono input jacks. If you are using the MCM jacks, Greenlee doesn't make a 3/8″ punch. I suggest Sescom's PUNCH-1. You can locate these holes in the same plane as the new CD input jacks, between the ground post and the edge of the chassis.
- Mount the new connectors in the newly prepared mounting holes.
- Use a short length of twisted-pair hookup wire to connect the right phono input jack to the PC board through the top side of the board. Twist together D.H. Labs OFC-20 hookup wire, conductors from their BL-1 interconnect, or use their T-20X unshielded twisted-pair hookup wire. Use another short length of twisted-pair hookup wire to connect the left phono input jack to bottom side of the PC board. Note that each new phono input jack has its own ground wire; both ground wires connect to the same PC board ground hole.

Photo 4 shows the new phono and CD input jacks mounted on the 565 preamp chassis and wired to the PC board.

I also suggest replacing the original ground post with hardware that's electrically more secure and a bit easier to get your fingers around, such as a 6-32 machine screw with a knurled-head thumb nut. These are normally hardware-store items, but I've also included Digi-Key part numbers in the Parts List. Minimum order quantities are 100—unless you need to stock up, you may choose to obtain them locally.

- Remove the original grounding screw. Adcom put a dab of solder on the end of the screw so it wouldn't accidentally come out. Hold your soldering iron against the end of the screw while removing the screw *with a screwdriver.*
- A 6-32 machine screw should fit the existing hole without enlargement. If necessary, enlarge the hole slightly; the protrusion on the inside of the chassis will remain.
- Scrape the paint off the end of the protrusion to ensure a good connection, and mount a 6-32 × 5/8″ machine screw from the inside of the chassis using a hex nut and lock washers. Use two lock washers—one for the screw head inside the chassis and another for the nut on the outside of the chassis. You can then tighten the nut with a nut driver without the screw turning (there's no room to get a screwdriver inside the chassis to hold it). Put the knurled nut on the machine screw.

I painted over the unneeded labeling on the rear panel with black semi-gloss modeling enamel, and made computer-printed labels for the new input and output jacks using Avery 8160 labels cut to size (avoid Avery's new "Smudge-Free" labels—they have poor print quality on ink-jet printers). I also covered the labels with Scotch Magic Tape. *Photo 5* shows the rear panel with the new connectors and ground post.

I suggest then applying Caig ProGold GP5S-6 contact conditioner to the newly installed jacks. I find that it is less messy to apply the ProGold to the RCA plugs on your interconnect cables, shake off the excess, and then insert the plugs into the jacks once or twice. D.H. Labs' new Silver Sonic Ultimate RCA Connector is an excellent mate to the RCA jacks I've recommended.

SIGNAL-PATH PURIFICATION

Although I don't describe the new line-stage circuitry until Part 3, there are some preliminary steps that you should take at this point to simplify the signal path and remove active devices that will no longer be used. First, disconnect the Loudness Contour circuit (*Photo 6*).

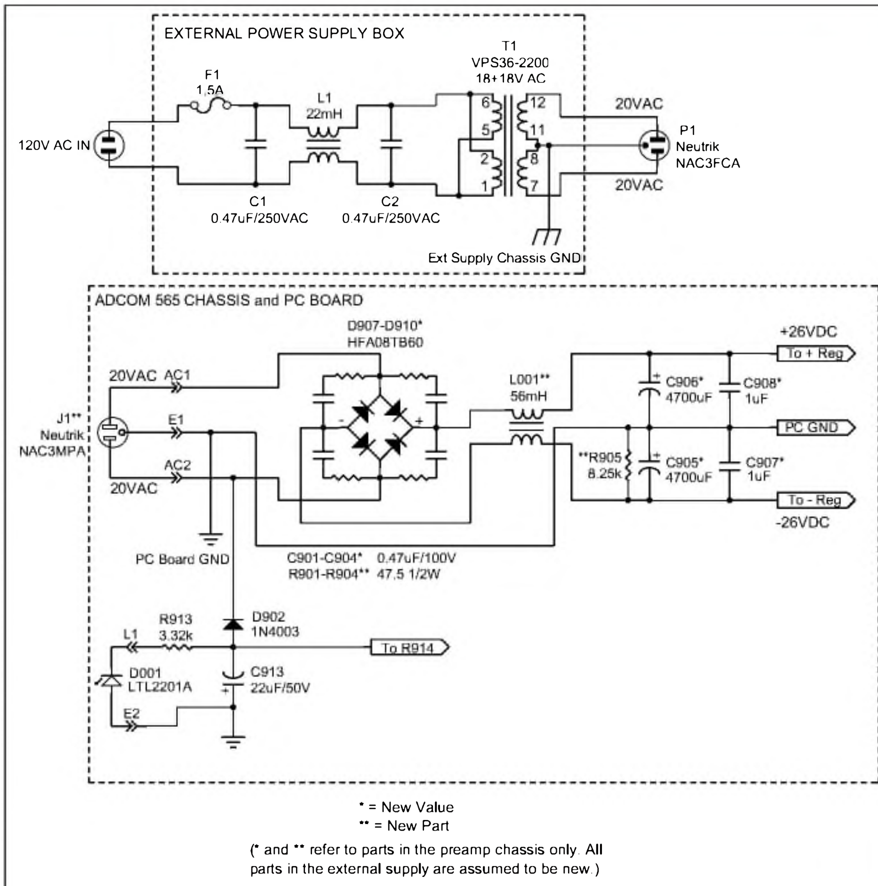


FIGURE 2: Top schematic is the external power supply. Bottom schematic shows internal changes to the GFP-565 preamplifier that will be described in Part 2, including the new rectifier bridge, R/C snubbers, a DC common-mode choke, and new raw-DC filter capacitors.

A-2289-2

- Cut a ¼″ gap in the PC trace closest to the front edge of the PC board, directly in front of—and perpendicular to—J49, next to the volume control VR204. Everyone has his or her own method for cutting PC traces. I use a combination of single-edge razor blades and X-Acto knives (be careful!). You can also use a Dremel tool with a small grinder attachment, but I suggest holding a vacuum cleaner nozzle close to the area you're grinding so that the fine debris is removed without making a mess. Either way, be sure that the PC board is completely clean when you are finished.

- Remove jumper J49 near the volume control VR204.

Next, remove the tone-control circuits and headphone amplifier from the signal path:

- Remove jumpers J124 and J27. These are the signal connections to the tone-control circuits, high filter, and the headphone amplifier.
- Remove IC205 and IC206. These are the tone-control op amps.
- Remove jumpers J7 and J9. These connect the main-supply rails to the tone-control circuits. On some samples of the GFP-565 I observed a very small local supply oscillation at IC205/206 due to the lack of any local supply bypassing. Removing these

PHOTO 5: The rear panel with new input and output connectors and a new ground post.

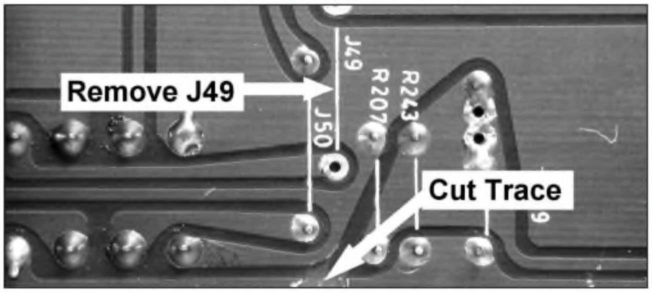


PHOTO 6: The loudness-contour circuit is disconnected by cutting one trace and removing jumper J49.

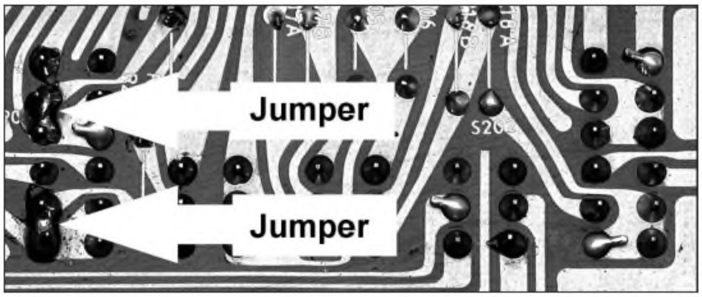


PHOTO 7: Two jumpers are soldered to the PC board to bypass the processor loop switch contacts.

jumpers eliminates the distribution of the supply to these op-amp footprints.
- Remove IC301. This is the headphone amplifier dual op amp.

Finally, bypass the processor loop switch contacts:

- Solder two jumpers to the processor loop switch S205, on the bottom side of the PC board (*Photo 7*).

Always leave the processor loop switch "out" when operating the preamp, so unused PC traces for the send and return lines aren't hanging off the signal path.

This completes the preliminary steps. Check the preamp to make sure it is operating correctly via the bypass outputs. Remember that the turn-on delay is now 20 seconds.

EXTERNAL POWER TRANSFORMER

The new power-supply scheme houses an AC line filter and a new power transformer in a separate metal box. There are two reasons for doing this. First, the 80VA transformer I use is too large to fit inside the Adcom chassis. Second, the hum field radiated by the unshielded transformer would compromise the low-noise performance of the preamp, especially the phono section.

I have not used exotic wiring for the power-supply construction in this preamplifier. You are free to do so, if you wish. I have also developed an alternate supply that houses dual rectifier bridges and raw DC filtering in the external box. The alternate supply is a bit more difficult to implement, but if there is sufficient interest, I may describe it in a future installment. Nearly all of the parts used in the external supply described in this article could also be reused in the alternate supply.

See *Fig. 2* and *Photo 8*. The Parallax transformer I recommend is a dual-bobbin type, which is much more effective in attenuating power-line noise than toroidal types, even those with electrostatic shielding. Rick Miller supplied some measurements documenting this for Part 4 of the 1995 power-supply regulator series.[5] The Parallax transformer is also very tightly wound, with virtually inaudible mechanical buzzing, and is readily available from Mouser and Newark.

MagneTek divested itself of its power transformer line some time back; Parallax is the successor. There are equivalent transformers from other manufacturers, including Hammond Manufacturing and Signal Transformer. The Signal A-41-series I recommended in the 4/1995 article is still available, but they now have a $15 handling charge on orders less than $100, which I consider unreasonable. I have also observed a higher level of mechanical buzzing from the Signal A-41 transformers, compared to the Parallax.

Some builders may wish to use an even heftier transformer. The next size up in the Parallax series is 130VA, but the higher secondary voltages under the load encountered in this preamp will increase the heat dissipation of the regulators, which may necessitate additional heatsinking. I have not tried a 130VA transformer with this preamp.

Neutrik PowerCon connectors are used to connect the transformer secondary to the preamplifier. An AC linecord that plugs into a switched outlet on

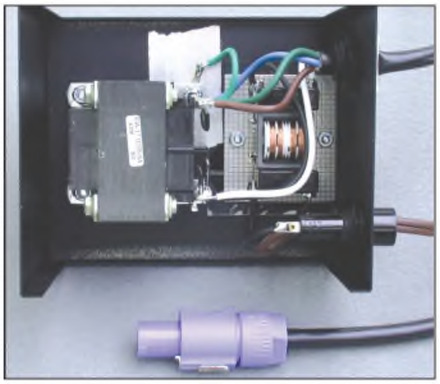


PHOTO 8: The external power supply. This supply houses an 80VA dual-bobbin power transformer and AC power line filter. A Neutrik PowerCon connector is used to connect the transformer secondary to the preamp.

PARTS LIST

PRELIMINARIES
(1) Adcom GFP-565 Service Manual (Adcom)
Adcom Regulator Voltage Change
(1) 12.7k, ¼W resistor (R908), Digi-Key 12.7KXBK-ND
Muting Circuit
(1) 1N4002 diode (D916), Digi Key 1N4002DICT-ND
(1) 1M, ¼W 1% metal-film resistor (R922), Digi-Key 1.00MXBK-ND
(1) 332k, ¼W 1% metal-film resistor (R920), Digi-Key 332KXBK-ND
New Input and Output Jacks
(6) or (8) Gold-Plated, Teflon-Insulated, Chassis-Mount RCA Jacks: MCM 50-2110 (Black Marking) and 50-2105 (Red Marking) (MCM Electronics) or CONNEX-53451 Inside-Mount (sold in L/R pairs; Parts Connexion) or Vampire M2F (sold in L/R pairs; Welborne Labs)
New Ground Terminal
(1) 6–32 × 5⁄8″ machine screw, Digi-Key H364-ND
(2) #6 Lock washers, Digi-Key H240-ND
(1) #6 Hex nut, Digi-Key H220-ND
(1) #6 Knurled head thumb nut, Local Hardware Store
Miscellaneous
Hookup wire: D.H. Labs OFH-20, T-20X twisted pair, or center conductors from BL-1 (D.H. Labs & Welborne Labs or Welborne Labs TWR, TWB or TWW (Welborne Labs);
16 AWG insulated solid bus wire, Digi-Key A3057B-100-ND
Caig ProGold GP5S-6 contact cleaner, MCM Electronics 200-190 or Caig Laboratories GP5S-6
Black semi-gloss modeling enamel
Avery 8164 or equivalent computer labels (NOT Avery "Smudge-Free")
EXTERNAL POWER SUPPLY
External Supply Parts:
(1) Bud Aluminum Minibox, 6 × 5 × 4″, Mouser 563-CU-3007A (or Sescom MC-8A)
(2) Panasonic ECQ-UV, 0.47µF, 250V AC, Digi-Key P4614-ND (C1, C2)
(1) Panasonic H-series Line Filter, 22mH, 1A, Digi-Key PLK1034-ND or Panasonic V-series Line Filter, 56mH, 1.1A, Digi-Key PLK1017-ND (L1)
(1) Parallax (Formerly MagneTek) VPS36-2200, 18+18 VAC/80VA, Mouser 553-VPS362200 (T1)
(1) Neutrik Powercon NAC-3FCA, Mouser 568-NAC3FCA (P1)
(1) 1.5A Fast-Blow Fuse, Radio Shack 270-1006 (F1)
(1) Panel-mount fuse-holder, Digi-Key 283-2344-ND, Radio Shack 270-367
Miscellaneous
16AWG, 2-cond. AC power cord and plug
Strain relief for above
3-cond. 14AWG SJT wire
Strain relief for above
3-cond. 16AWG SJT wire
16 AWG insulated solid bus wire, Digi-Key A3057B-100-ND
Prototyping board, Circuit Specialists IF-RFB
Plexiglass
Preamp Chassis/PCB Parts:
(1) Neutrik Powercon NAC3MPA, Mouser 568-NAC3MPA (J1)
(1) 2.0A Slow-Blow Fuse, Radio Shack 270-1024 (F001)

the preamp provides 120V AC to the line filter and transformer. *Fig. 2* and *Photo 8* show the outboard supply schematic and give an idea of how to assemble it in the new external box. I won't give step-by-step assembly procedure for the external box, because it should be self-explanatory. However, here are a few suggestions for assembly:

- I used a Bud 6 × 5 × 4″ aluminum minibox for the outboard chassis. After drilling and punching the necessary holes in the box, spray-paint it with black, semi-gloss, fast-dry enamel. Sescom's MC-series of metal cabinets is available in a black-anodized finish, and offers an alternative to the Bud box. The bottom plate of the modular Sescom boxes may need mounting reinforcement in the center, to support the weight of the power transformer.
- Mount the AC line-filter components—C1, C2 and L1—on a small piece of RF prototyping board from Circuit Specialists (*Photo 9*). The sturdy solder pads on this board facilitate mechanically secure mounting of the components. One 10 × 10″ piece is enough for this entire project. Although the cost is rather high—about $55—the durability and ease of use is well worth the expense. As of this writing, Circuit Specialists is still giving away a $29 DMM free with any purchase over $50—an added incentive.
- Put a piece of insulating Plexiglas between the prototyping board and the metal box. You must make the Plexiglas larger than the prototyping board, so it extends beyond the board by ½″ on the ends where AC connections are made. This ensures that 120V AC won't accidentally make contact with the chassis. Use two 4–40 × ⅜″ machine screws, lock washers, and nuts to mount the board and Plexiglass to the metal chassis.
- Use 16AWG insulated solid bus-wire to parallel the transformer primaries (*Fig. 2*). You can use the same type of wire to connect the secondary windings in series.
- Use a length of 3-cond. 14AWG SJT wire for the secondary connection, with an appropriate strain relief. The wire should be available from any electrical-supply house. I used a hardware-store strain relief designed for AC electrical-box use.
- The line cord should be 2-cond., 16AWG. Again, use an appropriate strain relief.
- The line cord and transformer secondary cable can be up to 6′ long, allowing you to place the external supply away from the preamp. If you can place the outboard supply close to the preamp, shortening these cables may provide a minor sonic benefit.
- Note the labeling of the Neutrik NAC3FCA PowerCon connector, particularly the ground pin. Connect the PowerCon to the end of the SJT cable. You can use the screw-terminal connections provided by Neutrik, or remove the screws and hardware, and solder the wires to the connector (I prefer soldering). The plastic used by Neutrik seems to have a very high melting point—you can apply considerable heat when soldering without deforming the plastic. If you opt not to solder, treat the wiring with Caig ProGold before connecting it to the PowerCon.
- After assembling the supply, check the secondary voltages at the Neutrik PowerCon connector. It's easier to do this before putting the connector together. With no load you should measure ~21.5V AC either side of ground.

CONNECTION TO CHASSIS

Now that the outboard supply is built, it's time to prepare the preamp chassis for the connection of the external supply.

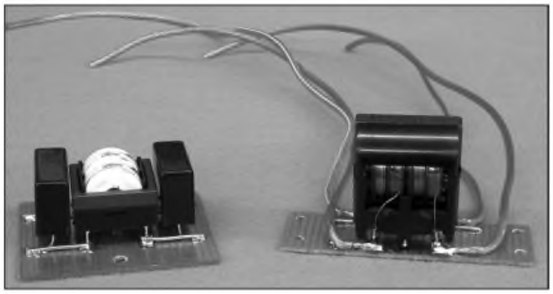


PHOTO 9: The AC line-filter assembly for the external supply is shown on the left. The raw- DC common-mode choke (assembled and installed in Part 2) is shown on the right. Components are soldered to Circuit Specialists RF Prototyping Board.

- Clip the power leads at AC5 and AC6 on the Adcom TX 919115 voltage-select PC board, as close as possible to the board. Remove the white lead—it goes to one of the switched outlet's neutral contacts. Strip and tin the hanging end of the brown wire from the fuse holder. You will use it shortly.
- Remove the three wire-wrap posts at E1, AC1, and AC2 on the main PC board. These are the power-transformer secondary leads.
- Cut two cable ties to free the Adcom TT-271-GF power transformer leads. Take care to cut only the ties, and not the leads.
- Remove the power transformer, along with the voltage-select PC board and its mounting bracket. Save the power transformer mounting hardware. Also remove and save the metal-oxide varistor MOV001 from the voltage-select PC board. You can discard the transformer and voltage-select board, or save them for another project.

Now you can re-configure the innermost switched outlet so that it is both switched *and* fused (*Fig. 3*). This is the outlet directly above the fuse holder. The AC power cord from the outboard supply will be plugged into this outlet so that the preamp's front-panel power switch will still function normally. Fusing the outlet is an added safety precaution—if there is a fault in the external supply's line cord, this fuse will blow.

The main power-supply fuse is F1 in the external chassis. F1 is now 1.5A, rather than the 1A used in the original Adcom design, which allows for the additional surge currents required to charge higher-value power-supply bypass capacitors when you power up the preamp.

Fig. 3A shows the original Adcom wiring configuration for the switched outlets.

- Re-wire the switched outlet closest to the main PCB (as shown in *Fig. 3B*). You can connect the tinned, hanging wire from the fuse holder to the hot terminal (the top pin) of the switched and fused outlet. Replace F001 with a 2.0A slow-blow fuse.
- Re-connect MOV001 between the fuse-holder and the bottom pin of the re-wired switched and fused outlet. You will need to lengthen the MOV leads and cover them with sleeving. The fuse holder output connects to the brown wire that previously went to the voltage selector board. The bottom pin of the re-wired outlet is the wider of the two pins, and should have a white wire attached.

Photo 10 shows the re-wired switched outlet and fuse holder. *Photo 11* shows new labeling for the re-configured outlet and the fuse holder, and also how to mount the Neutrik NAC3MPA PowerCon chassis connector on the rear panel between the line cord strain relief and the fuse holder.

- Using a single-edge razor blade, carefully remove the adhesive serial number, located between the fuse holder and the line-cord strain relief. Using Scotch Magic Tape, re-fasten the serial number to the left of the fuse holder (*Photo 11*).
- Punch a $\frac{15}{16}$″ hole for the PowerCon chassis connector in the location shown in *Photo 11*. A Greenlee $\frac{15}{16}$″ punch is available from Mouser (#586-3808).
- Center the PowerCon chassis connector in the hole, orienting it so the "Neutrik" label is on the bottom. Mark and drill a pair of holes to accommodate #4 machine screws. Mount the connector with $\frac{3}{8}$″, 4-40, flat-head machine screws, lock washers, and nuts.
- Connect the three pins of the PowerCon chassis connector to holes AC1, AC2, and E1 on the PC board (*Photo*

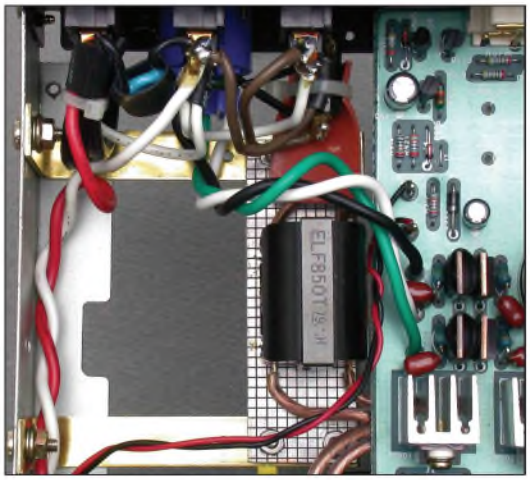


PHOTO 10: The re-wired switched—and fused—AC outlet is shown top center. Also shown is the wiring from the Neutrik PowerCon connector to the main PC board, and the DC common-mode choke assembly (installed in Part 2) mounted with a pair of brass brackets. Note the rubber insulator fastened to the bottom of the fuse holder with a nylon cable tie (also installed in Part 2).

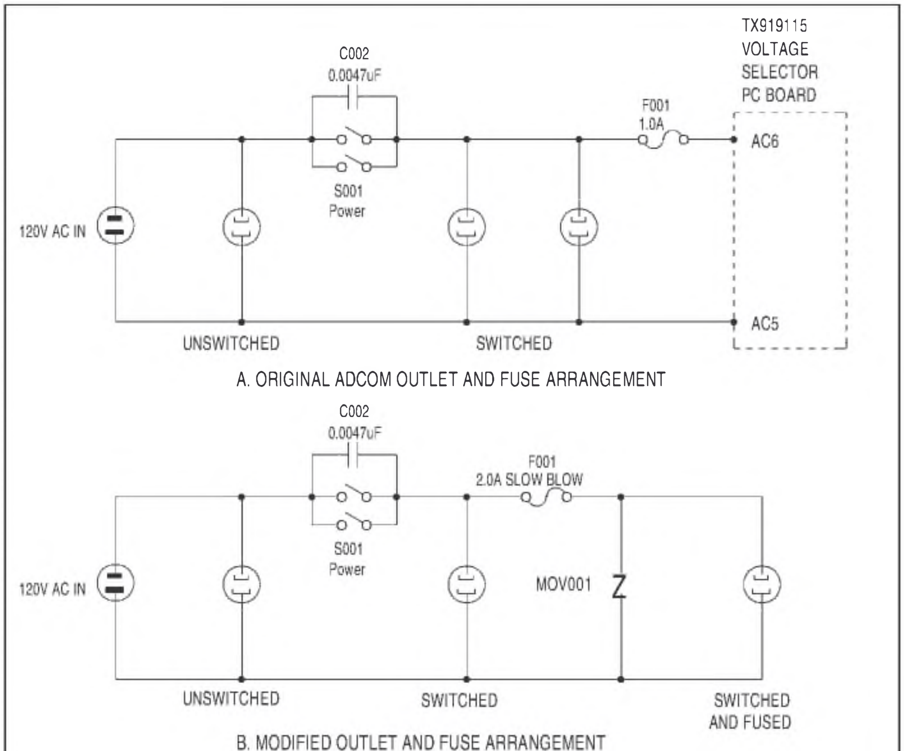


FIGURE 3: The original Adcom convenience outlet wiring is shown in schematic "A." Schematic "B" shows the re-wired switched and fused outlet. A-2289-3

10). Note that E1 is ground; the polarity of the two hot leads—connected to AC1 and AC2—isn't critical. I used 16AWG SJT wire taken from a power cord, twisted together as shown in the photo. 14AWG wire will be difficult to fit in the PC board holes—the length is so short that I find 16AWG to be fine. Leave enough slack in the wiring to allow room for the DC filter choke shown in the photo (the choke will be installed in Part 2).

CHECKPOINTS

At this point, the preamp should operate properly with the new outboard transformer. Treat the PowerCon contacts with Caig Pro-Gold and connect the PowerCon plug on the outboard supply cable into the PowerCon connector on the preamp chassis (Pro Gold is suitable for most precious metals, including the nickel plating on the PowerCon). Align the connector, insert and twist until it clicks. Plug the line cord from the outboard transformer into the fused and switched outlet on the preamp, and plug the preamp's line cord into an AC outlet.

Turn on the preamp power switch and check the transformer secondary voltages at AC1 and AC2 on the main PC board, referenced to E1. There should be ~20VAC at AC1 and AC2. Remember that this is an 80VA transformer, so the secondary voltages will be a bit higher than the specified 18V AC under the load encountered in this preamp. Check the regulated supply voltages, which should read ±17.5V DC.

It is normally a good idea to orient the AC line cord for minimum leakage to the chassis. The optimum position is that which yields the lowest AC potential between the preamp chassis and the AC power line ground. You can check your preamp by *temporarily* adding a non-polarized ground lifter to the line cord from the external supply. This will allow you to reverse the cord in a polarized outlet.

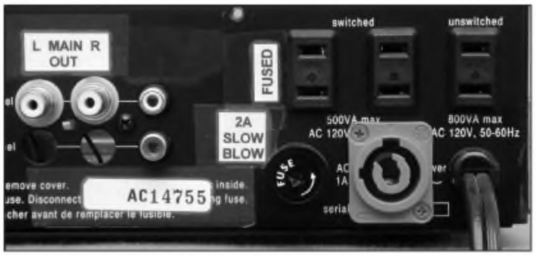


PHOTO 11: Rear-panel view showing the Neutrik PowerCon connector. The serial number has been moved to the left of the fuse holder, and new labeling has been added for the fuse holder and the fused AC outlet.

Using an AC voltmeter, measure the potential between the preamp chassis and the AC line ground with the AC cord in its normal orientation; reverse the AC cord orientation and measure again. If the reverse position gives a substantially lower reading (more than a few volts), you should reverse the AC line cord polarity inside the external supply. The easiest place to do this is at the input to the line filter. In the preamps I've modified, I've found little difference (only a volt or so) changing AC polarity. It's worth checking, though.

Next time, we'll replace the rectifiers, snubbers and filter capacitors, and add common-mode filtering to the raw DC supply. We'll also remove the Adcom supply regulators and install the Jung Improved Regulators (note that *Fig. 2* contains parts that will be installed in Part 2). In the meantime, listen to the preamp and enjoy the improvement a high-current transformer can make. If you use a power-line filter with sequenced switching, such as those made by Adcom or Panamax, you can plug the outboard supply's line cord directly into the power-line filter. In this case, the preamp's power cord is not used. You can also do some critical listening to see whether you can hear a difference when the preamp power cord is eliminated (this may make a greater difference when the rest of this project is completed). ❖

SOURCES

Adcom, division of Klein Technology Group, LLC
8551 E. Anderson Drive
Suite 105
Scottsdale, AZ 85255
480-607-2277
FAX 480-348-9876
www.adcom.com
parts@adcom.com

Accutek Microcircuit Corp.
2 New Pasture Rd., Ste. 1
Newburyport, MA 01950
978-465-6200
FAX 978-462-3396
www.accutekmicro.com
sales@accutekmicro.com

Analog Devices
(factory-direct on-line ordering)
www.analog.com

Caig Laboratories, Inc.
12200 Thatcher Court
Poway, CA 92064
858-486-8388
FAX 858-486-8398
caig123@caig.com
www.caig.com

Circuit Specialists Inc.
220 South Country Club Drive #2
Mesa, AZ 85210
1-800-528-1417
480-464-2485
www.web-tronics.com

D.H. Labs, Inc.
612 N. Orange Ave.,
Suite A-2
Jupiter, FL 33458
561-745-6406
www.silversonic.com

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

Mouser Electronics
1000 N. Main St.
Mansfield, TX 76063-1511
1-800-346-6873
www.mouser.com

Newark Electronics
1-800-463-9275 (Call for the nearest sales office)
www.newark.com

Parts Connexion
2885 Sherwood Heights Drive
Unit# 72
Oakville, Ontario
CANADA L6J 7H1
1-866-681-9602 (Toll-free order line)
www.partsconnexion.com

Radio Shack
www.radioshack.com

Sescom, Inc.
707 N. Lindenwood
Olathe, KS 66062 USA
1-800-634-3457
www.sescom.com

Welborne Labs
P.O. Box 260198
971 E. Garden Drive
Littleton, CO 80126 USA
303-470-6585
www.welbornelabs.com

All photos were taken by the author using a Nikon CoolPix 950 digital camera. Photo editing was done using Microsoft Photo Editor V. 3.0 and Paint Shop Pro V. 7.04. Schematic drawing was done using CircuitMaker 2000 Professional Edition with Service Pack 1.

REFERENCES

1. Galo, Gary A., "Adcom GFP-565 Preamplifier" in *Stereophile*, Feb. 1990, pp. 150-158.
2. Jung, Walt, "Improved Positive/Negative Regulators" in *Audio Electronics*, 4/2000, pp. 8-19.
3. Jung, Walt, Jan Didden, and Gary Galo, "Regulators for High-Performance Audio," Parts 1–4 in *Audio Amateur*, 1, 2, 3, and 4/1995.
4. Galo, Gary, "Bi-Amping the Sapphire II/Sub I System" in *Speaker Builder*, 3/1992, pp. 24-32.
5. Galo, Gary, "Regulators for High-Performance Audio, Part 4: Real-World Implementations and Sonic Evaluations" in *Audio Amateur* 4/1995 (including power-transformer measurements by Rick Miller on p. 36).

ACKNOWLEDGMENTS

Special thanks go to Walt Jung, not only for the design of the Improved Regulators, but for the many suggestions and insights he has shared since I began modifying the preamp over a decade ago. Thanks also to my wife Ellen for her proofreading, to Rick Miller for his input on rectifier diodes and snubbers, and to Rich Markell of Linear Technology for supplying IC regulator samples at a time when dealer's stocks were in short supply. Last, but not least, I would like to thank Victor Campos for his interest and encouragement over the many years I have been working on this preamp, for the meticulous proofreading on which I have relied countless times, and for badgering me to finally get this article finished. Victor's loan of one of the three preamps I have modified was extremely valuable in helping me develop a coherent (I hope!) assembly procedure for these articles.

A Servo Dual Voice Coil Subwoofer

Here's an application of the closed-loop concept to the subwoofer to produce accurate, low-cost bass.

By Daniel L. Ferguson

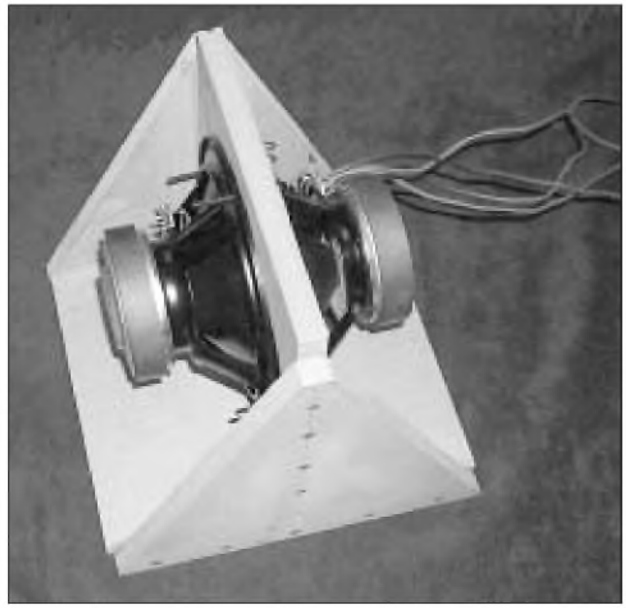


PHOTO 1: Setup to test velocity signal.

A great deal of effort has been put forth over the past half century to find the best way to reproduce clean, accurate bass. Many different solutions to the problem have been presented in publications such as the *Journal* of the Audio Engineering Society and *Speaker Builder*. With few exceptions, all of them are "open-loop" systems. They all depend on the electromechanical ingenuity of the loudspeaker designer and the physics of the enclosure design to, in effect, "predict" how the voice coil and diaphragm will react to an electrical input.

While some of these designs work better than others, all open-loop solutions have limitations in accuracy. Not many subwoofers are capable of reproducing the lowest octave at audible levels, and when they do, distortion levels of 5 to 10% are common.

The "opposite" of an open-loop system is a closed-loop one, in which the movement of the driver's diaphragm is measured and compared to the input signal, and the difference between the two becomes the error signal, that is used to "force" the diaphragm motion to conform more closely to the input. This is the concept of negative feedback that is used in a servomechanism, which Webster defines as "an automatic device for controlling large amounts of power by means of very small amounts of power and automatically correcting the performance of a mechanism." The date of this definition is 1926, so the concept has been around for a while. Applied to the loudspeaker, the implications are that a closed-loop subwoofer should have deeper bass extension, lower distortion, and flatter frequency response.

While the closed-loop woofer concept is not new, it's been slow to migrate to the commercial loudspeaker industry—undoubtedly due to its significantly higher cost and complexity. Examples of commercial servo subwoofers that come to mind are the hugely expensive Infinity Servostatic system of the 1970s and the modern Velodynes. Both of these use accelerometers attached to the driver cones to measure the cone motion. Other methods of directly or indirectly measuring cone motion have been tried, but none that I am aware of have been accepted on a wide basis.

BACKGROUND

The purpose of this article is to present some work I have done on developing an accurate and relatively low-cost method to close the loop on a subwoofer. This method uses a dual voice-coil driver in a closed box configuration, with one coil driven and the other coil used as a velocity sensor to provide the feedback signal to the servo.

Figure 1 illustrates how the concept works. The signal is introduced into a summing junction, where it is compared to the measured acceleration signal produced by the undriven voice coil of a dual voice-coil woofer. The difference between these two becomes the error signal, which is fed to a power amplifier that drives the other voice coil. This concept only works for closed box systems, where the motion of the driver cone is the sole source of sound produced by the system.

This idea first occurred to me in 1984. I remember the year because I recall where I was working when I contacted one of the major loudspeaker manufacturers that produced dual voice-coil (DVC) woofers to ask about the feasibility. I had a nice conversation with a very

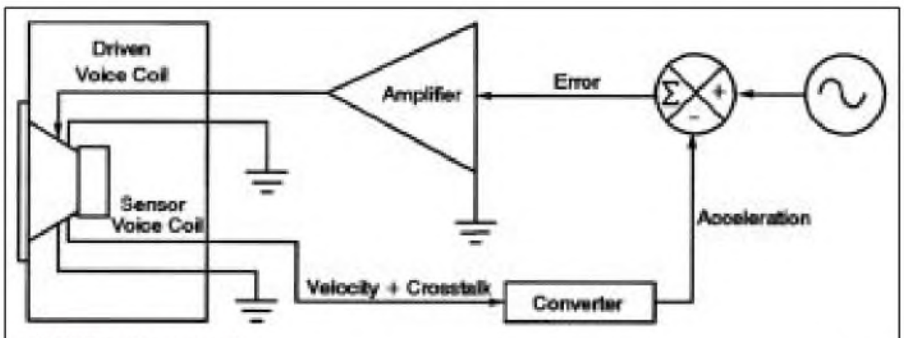


FIGURE 1: Servo woofer.

obliging fellow who was an authority on loudspeaker design. When I asked about the possibility of using one of the coils as a velocity sensor, he replied that it wouldn't work because the sensor signal would be "nonlinear." Since mathematically nonlinear functions are the kiss of death in servo design, I reluctantly tucked the concept away in the back of my mind but never really forgot about it.

Recently, I decided to see whether I could find anything on the Internet concerning the use of a dual voice-coil speaker in a servo configuration (one coil driven and the other used as a sensor). I was thrilled to find a paper on the subject published in 2000 by the University of Michigan[1]. I couldn't wait to download and read it.

The paper was written by four researchers at three major universities, so I was confident of the quality of the work. It answered the questions I had wondered about and filled in almost all of the missing gaps. However, I found one crucial area that I believed I could significantly improve upon.

The main points of the paper (from my perspective) were:

1. The sensor voice coil produces a voltage proportional to velocity that is a mathematically linear function!
2. Sound pressure level (SPL) is proportional to loudspeaker cone acceleration—not velocity. Therefore, the derivative of the velocity signal must be used as the feedback signal in a servo.
3. The sensor voice-coil voltage contains induced voltage (crosstalk) from the driven voice coil.
4. The induced voltage can be filtered out.

EXPLICATION OF POINTS

1. The principle that a conductor moving through a magnetic field produces a voltage proportional to velocity is one of Faraday's laws and is the basis for all generators. What was new information for me was that the loudspeaker voice-coil magnet system produces a linear voltage. In other words, the faster the voice coil moves, the higher the voltage. The term "linear" does not necessarily (or usually) mean a straight line; it means that the process can be modeled with linear differential equations. This is crucial to the servo design.

2. I have always wondered whether SPL was proportional to velocity or acceleration. Reference 1 cites a 1959 Audio Engineering Society paper by J. F. Novak which states that SPL is proportional to acceleration for a speaker with a diameter less than one-third of its highest operating frequency wavelength. For a subwoofer, the passband is ideally 20 to about 100Hz. Therefore, the highest audible frequency it would probably produce is an octave or so above this (200Hz).

   At 200Hz, the wavelength of sound is about 67.8″. One third of this is 22.6″, so the relationship should hold for woofers up to that diameter.

3. It should be obvious that two precision-wound coils in intimate contact with each other will exhibit plenty of mutually induced crosstalk.

4. The authors of reference one used a second-order filter to contour the sensor voice-coil velocity signal to make it more representative of the actual velocity. This is where I believe they arrived at a sub-optimal solution.

In their elegant mathematical model, the equation (after simplifying) that describes the voltage appearing at the sensor voice-coil terminals is:

$$V_s(t) = c(dx/dt) + M(di_1/dt)$$

where $V_s(t)$ = voltage across the sensor voice coil
c = forcing factor or back emf constant of the electromechanical system
dx/dt = velocity of the cone
M = mutual inductance of the two voice coils
di_1/dt = rate of change of current in the driven voice coil

From this equation, the crosstalk component in the sensor voice-coil voltage is produced solely by the rate of change of current in the driven voice coil. If it can somehow be removed, the sensor voice-coil voltage will accurately represent cone velocity. It would seem more straightforward, then, to use a current sensing circuit to directly measure the current in the driven voice coil and then simply subtract it from the voltage present at the sensor voice-coil terminals. When properly scaled, the sum of the current and sensor voice-coil signals should equal cone velocity, and the derivative of this signal should be representative of the driver's sound pressure level. Therefore, the main point of this article is to investigate whether or not this supposition is correct.

PUT TO THE TEST

Before proceeding with design of the servo, I needed to prove my theory. The first step in the process was to verify that the velocity signal is inherently clean and representative of cone velocity if no induced voltages from the driven coil were present. I accomplished this by bolting two 8″ DVC drivers

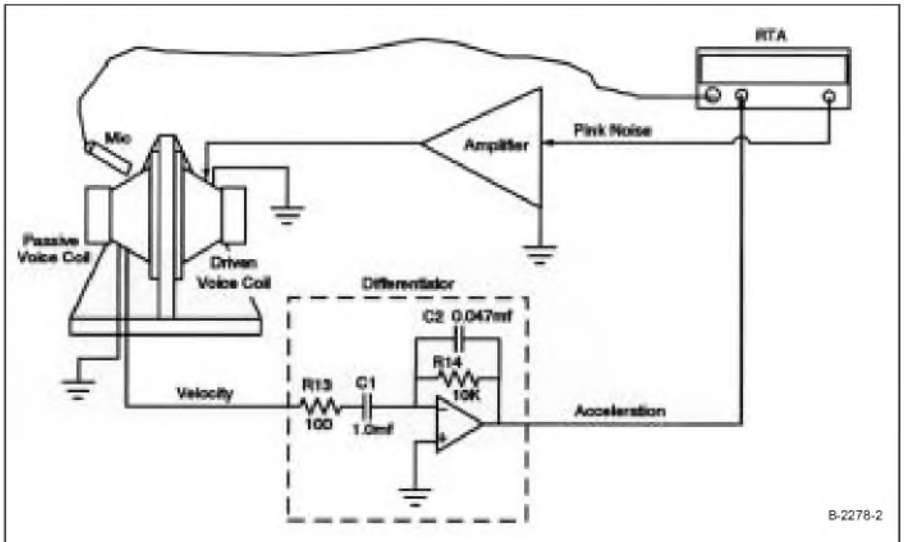


FIGURE 2: Test setup for verification of sensor voice-coil accuracy with no crosstalk.

(Peerless 831858) together face-to face as shown in *Photo 1*. One driver is driven with a signal generator and power amplifier while voltages are measured at one pair of voice-coil terminals on the passive driver.

The wiring diagram for the test rig is shown in *Fig. 2*. Since there are no electrical connections to the passive driver (it's driven by the air trapped between the two), there can be no mutually induced voltage. So whatever voltage is present is produced solely by the motion of the driver's voice coil moving through the magnetic field.

While I could accurately measure SPL with a real-time analyzer, SPL is proportional to acceleration—not velocity. With no way to measure the cone's velocity directly, I could only measure it indirectly after converting it to acceleration. Running the passive driver's output signals through a derivative circuit (referred to as a differentiator) accomplished the analog conversion from velocity to acceleration (*Fig. 2*).

The first test looked at the raw voice-coil signal to see whether it was clean and sinusoidal at all frequencies. The lower trace in *Photo 2* is a representative sample at 30Hz. The upper trace is the same signal after processing through the differentiator. Both appear to be free of anomalies. Since the derivative of Sin(ωt) is ωCos(ωt), the approxi-

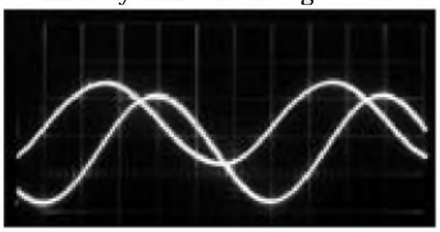


PHOTO 2: Voice-coil signal measurements.

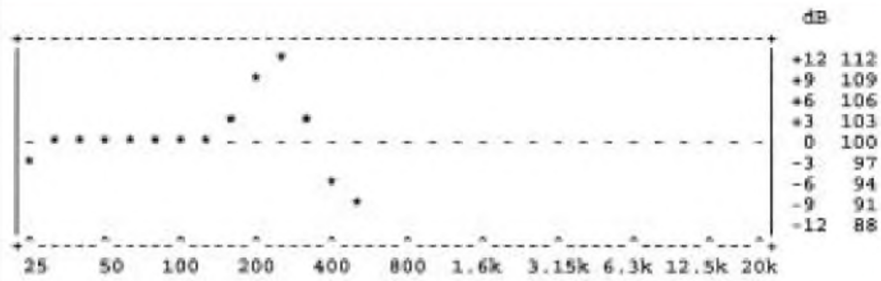


FIGURE 3A: Passive driver acoustic pink-noise response.

B-2278-3a

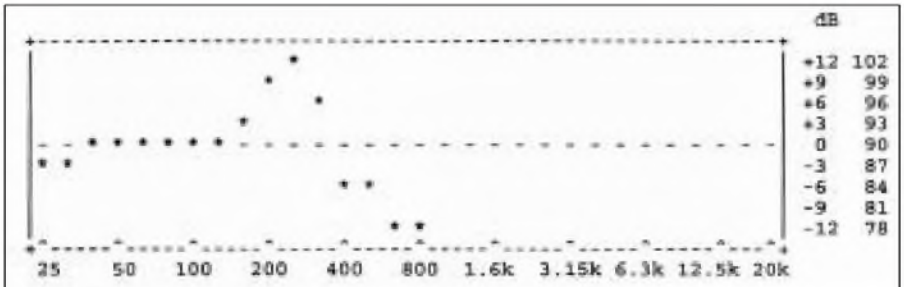


FIGURE 3B: Passive driver voice-coil pink-noise response after differentiator.

B-2278-3b

mate 90° phase shift is one indication that the differentiator is working. Another aspect of this particular derivative is that the magnitude is multiplied by the frequency (ω). So, compared to its input, the output of the differentiator has a constant 90° phase shift and an amplitude which increases with frequency.

If the differentiator is working properly and Novak's theory is correct, the frequency response measured at the passive driver cone will equal the output from the differentiator. To verify this, I ran full-range pink noise through the driven speaker and recorded the sound of the passive driver's cone motion with a calibrated microphone and a real-time analyzer (RTA). I captured the response curve in *Fig. 3a*.

Next, I ran the output from the differentiator into the RTA and obtained the response shown in *Fig. 3b*. The two are essentially identical. From this, I concluded that the signal had to be representative of the passive driver's cone velocity.

If the above sounds like a cakewalk, it wasn't. Active differentiators are inherently noisy, and the output of a differentiator increases linearly with frequency. Therefore, solid-state devices can easily reach saturation voltages if the differentiator is not configured to operate within a fixed range of frequencies. I chose the lowest resistance values possible to reduce the noise and experimented with component values for several days before I was satisfied that I had a clean, stable unit with plenty of headroom.

In the *Fig. 2* schematic, the differentiator operating range is set by the combination of R14 and C1 and R13 and C2. For the values shown, the operating range is 15.9Hz to 33.9kHz. To ensure maximum linearity, it is good practice to maintain a minimum ratio of 100:1 between these two.

SERVO CIRCUIT

The next step was to build the driven voice coil's current-sensing circuit to measure the mutual inductance component and the summer to subtract its output from the raw velocity signal. For this operation, I constructed the circuit shown in *Fig. 4*, which also includes the final summing junction and amplifier, which will act as the controller for the servo. The prototype circuit board is shown in *Photo 3*.

In this circuit, op amp C provides scaling for the output of the sensor voice coil before the input to the first summer—op amp D. The power amplifier positive output is connected as shown to R7, the other side of which connects directly to the positive terminal of the driven voice coil. Current flowing through R7 (which has a value of 0.1Ω) causes a slight voltage drop that is measured by differential op amp A with a gain of one. Its output is scaled by op amp B prior to summing at the input of op amp D.

The output of op amp D represents the cleaned-up velocity signal and is fed to the differentiator—op amp E. When all inputs are scaled properly, the output of op amp E should be equivalent to acceleration and therefore representative of sound pressure level.

To test whether the "accelerometer" worked or not, I removed one of the woofers from the test stand and applied a pink-noise signal to the input of the

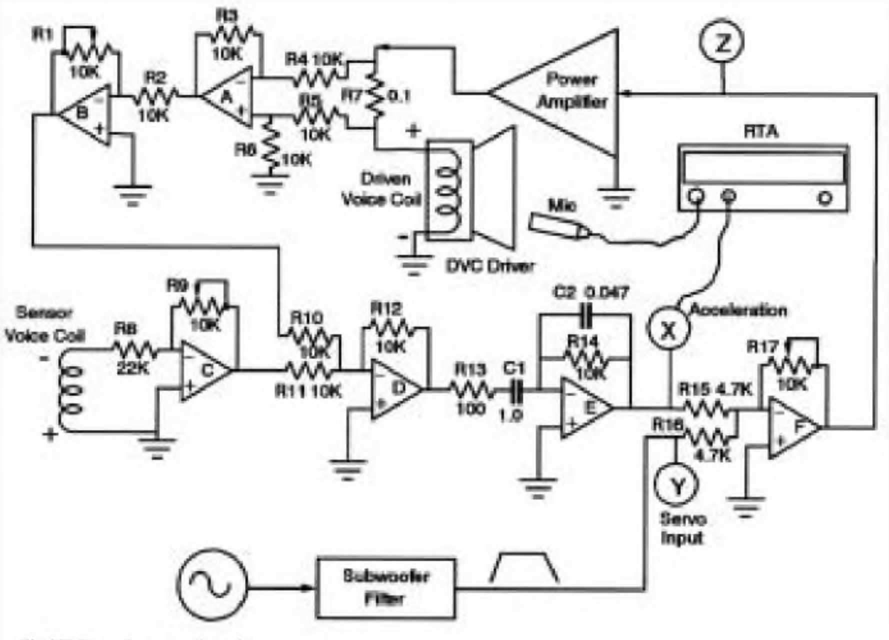


FIGURE 4: Servo circuit.

B-2278-4

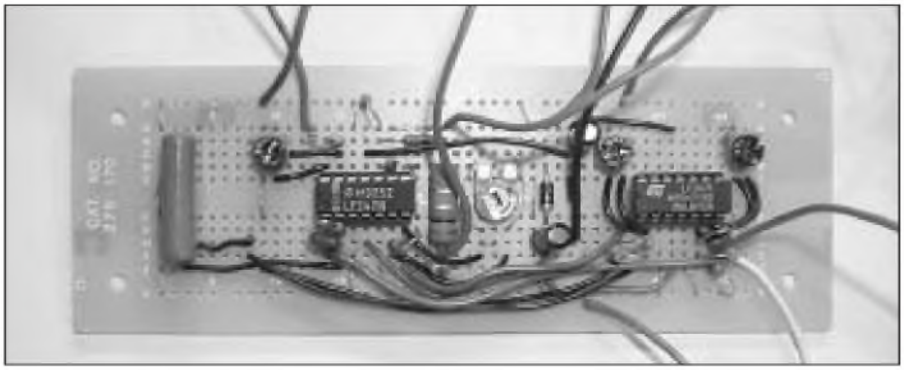


PHOTO 3: Servo circuit board.

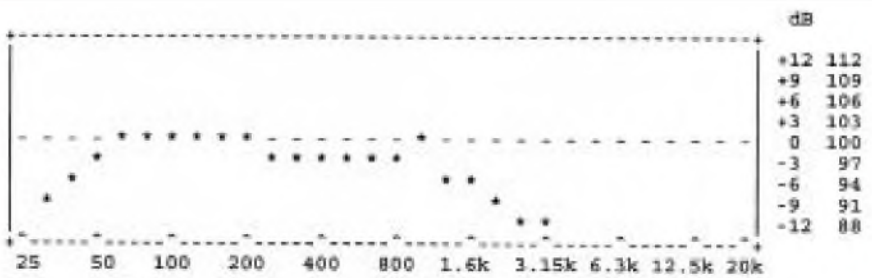


FIGURE 5A: Single driver in test stand acoustic response to pink noise.

B-2278-5a

power amplifier (point Z). With the mike set close to the cone, I captured the response on the RTA (*Fig. 5a*).

Next I connected the output of op amp E (point X) to the RTA and experimented with the scaling pot settings to try to find a combination that would make the two similar. *Figure 5b* is the differentiator output with the current-sensing circuit gain set to minimum. I then increased the current-sensing gain to what appeared to be optimum and obtained the results shown as *Fig. 5c*.

While *Figs. 5a* and *5c* are similar below 200Hz, above this frequency, they diverge. The extended high-frequency output of the sensor voice coil is primarily crosstalk from the active voice coil. The mathematical model indicates that the high-frequency components in the pink noise are easier to transmit by mutual inductance—M(di/dt). Also, phase shifts between the crosstalk and the back emf generated by the woofer's moving mass system allow the higher frequencies to pass through unopposed. Once the high frequencies hit the differentiator, they are multiplied by the frequency and can easily saturate the op-amps.

The presence of all the high-frequency components obscured the data. Clearly, the range of frequencies to be examined would need to be restricted to verify whether the accelerometer worked. I dusted off one of my subwoofer filters, set it for flat response out to 20Hz, and connected it between the pink-noise source and amplifier. I then re-ran the response curves.

Figure 6a is the woofer SPL response measured with the mike. *Figure 6b* is the accelerometer response after intro-

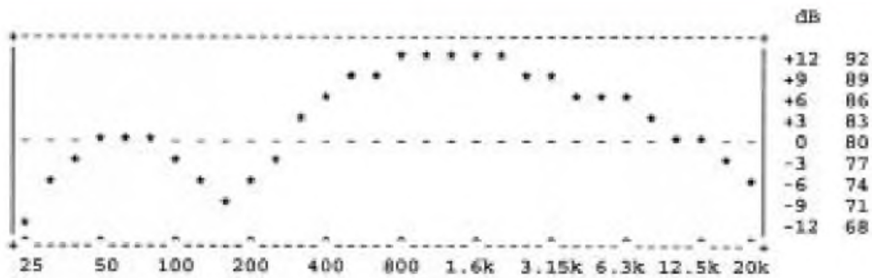


FIGURE 5B: Derivative of sensor voice-coil output with no current sensing. B-2278-5b

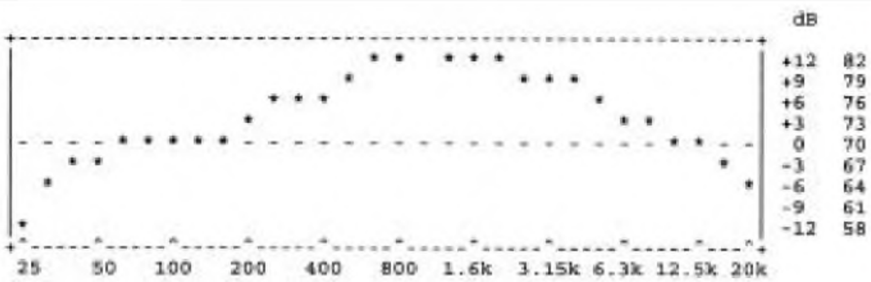
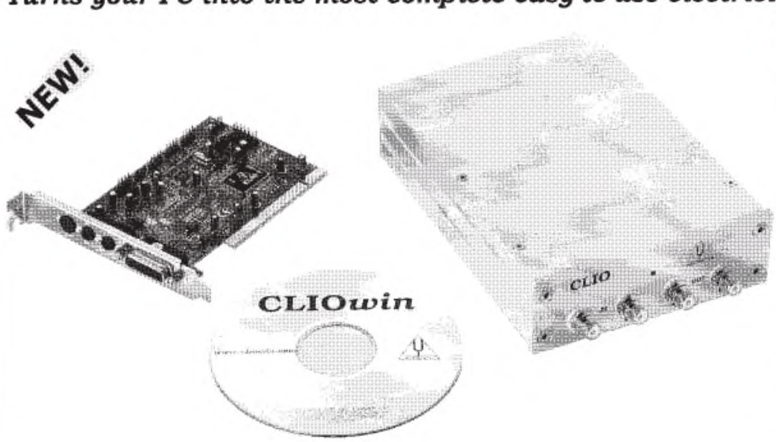
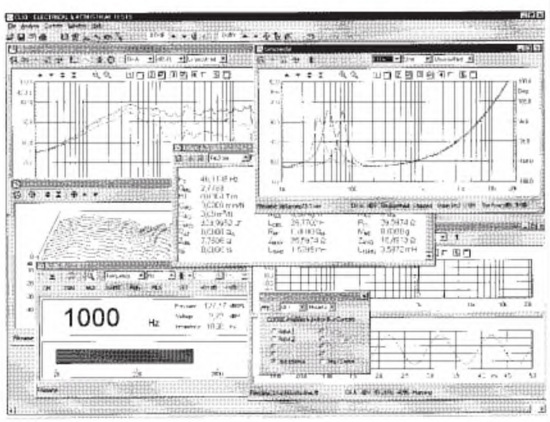


FIGURE 5C: Derivative of sensor voice-coil output summed with current sensor. B-2278-5c





ducing and scaling the input from the current sensor. The accelerometer RTA curve is now similar to the one produced with the mike.

I then connected the sine-wave generator and swept the frequencies up and down looking for points of interest. The microphone waveforms indicated that the woofer generated visible (and audible) distortion at 22Hz. A sample is shown as the lower trace in *Photo 4*. The upper trace is the waveform generated by the accelerometer. Again, the two appear quite similar. I concluded that, since both waveforms and frequency responses are similar, the accelerometer is, in fact, working. The feedback sensor is ready to be employed in a servo-subwoofer.

SUBWOOFER DESIGN

Moving to the design of a subwoofer, the published Thiele-Small parameters for the Peerless 831858 driver used in this experiment are f_S = 22Hz, Q_{TS} = 0.23, V_{AS} =79.8 l, and X_{MAX} = 7mm. With one voice coil driven, Q_{TS} will be about 0.46—nearly ideal for a closed box. The f_S of 22Hz is very low, and the linear excursion rating of 7mm is very high for an 8″ woofer. Both of these parameters are exceptionally good. (Incidentally, this woofer performs outstandingly in normal vented box applications with both voice coils driven. It's by far the best 8″ woofer I have found, and its reasonable price makes it even more exceptional).

In the $1ft^3$ test box, closed-box Q measured 0.8—close enough to the ideal of 0.707. Closed-box resonant frequency was 44Hz.

To close the loop and transform this to a servo-subwoofer, I completed the hookup shown in *Fig. 4* by connecting the output of op amp F to the input of the power amplifier and connecting various signal sources to the servo's input, point Y. I left the accelerometer settings the same as they were after the earlier

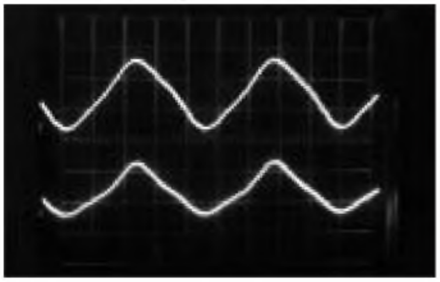


PHOTO 4: Waveform comparison.

"calibration" sessions. Before turning on the power amplifier, I set the final gain setting on R17 to minimum. Now for the moment of truth—I turned on the power amp and began slowly increasing the gain until the system began to oscillate and then backed it down slightly. When I was sure the system was stable, I ran some more tests.

FINAL TESTING

Figure 7a is the pink-noise response curve for the subwoofer filter. If the servo is working properly, it will cause the woofer to reproduce this same curve. *Figure 7b* is the woofer's open-loop SPL response to the filtered pink noise. With a closed-box frequency of 44Hz, the system is down 3dB at approximately 40Hz. *Figure 7c* is the same test with the feedback introduced, except the scale is reduced from 3dB increments to 1dB. Clearly, the –3dB point has been lowered to approximately 25Hz—a significant improvement.

The response of the servo subwoofer is essentially the same as the filter's response, which is the desired goal. *Fig-*

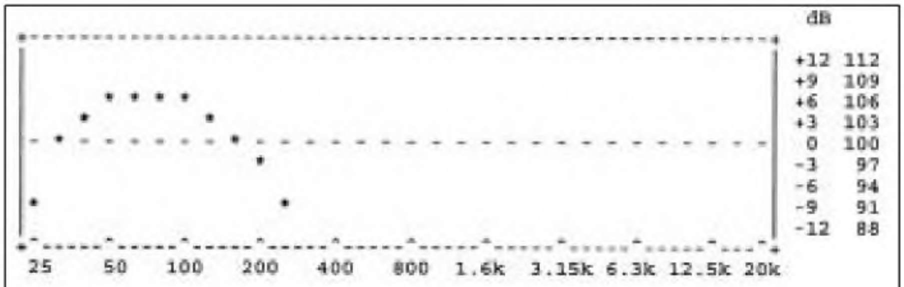


FIGURE 6A: Test stand driver acoustic response to filtered pink noise.

B-2278-6a

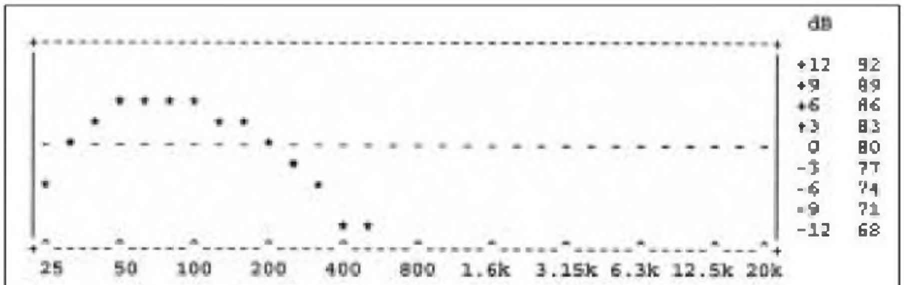


FIGURE 6B: Test stand driver accelerometer response to filtered pink noise.

B-2278-6b

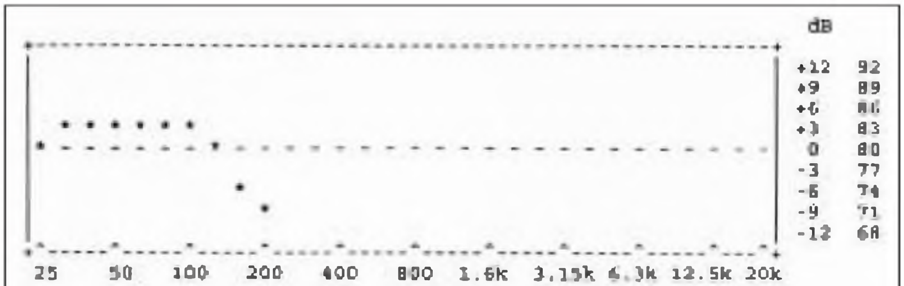


FIGURE 7A: Subwoofer filter pink-noise frequency response.

B-2278-7a

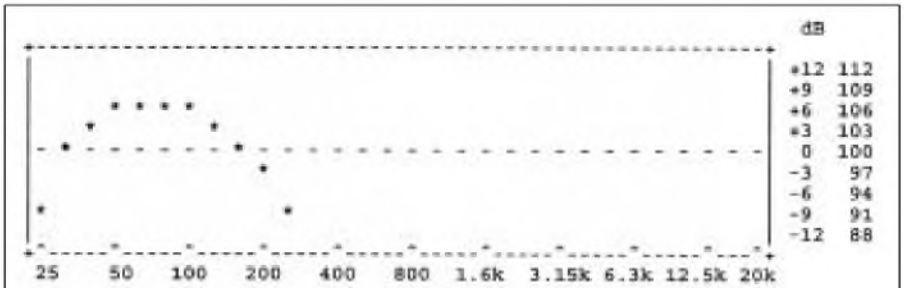


FIGURE 7B: Subwoofer open-loop acoustic response to pink noise.

B-2278-7b

ure 7d is the accelerometer's response during this test and is nearly identical to that of the calibrated microphone.

The last test was to see whether the servo would reduce distortion in a real-world situation. In *Photo 5*, the upper trace is the signal-generator sine wave, again at 22Hz. The lower trace is the subwoofer open-loop waveform captured with a microphone. It is somewhat triangular and produces audible distortion.

Photo 6 shows the same test run in closed-loop mode. The upper trace is the signal-generator waveform, and the lower trace is the subwoofer waveform captured with a microphone. It now more closely resembles the signal generator and is audibly quieter. Therefore, closed-box distortion levels have been audibly reduced.

It's important to remember that this is a worst-case stress test of the system. The magnetic system is operating in its nonlinear range at full excursion. For frequencies above 25Hz, the closed-loop waveform appears distortion-free.

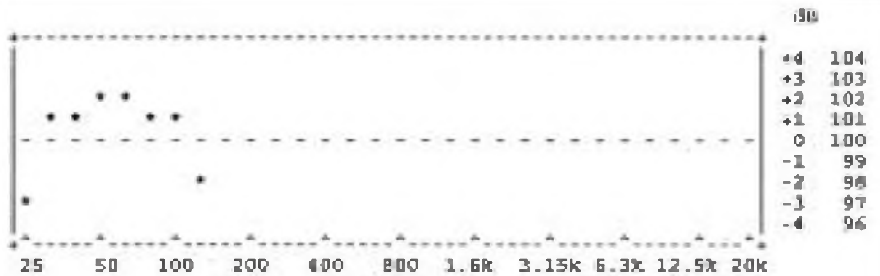


FIGURE 7C: Subwoofer closed-loop acoustic response to pink noise.

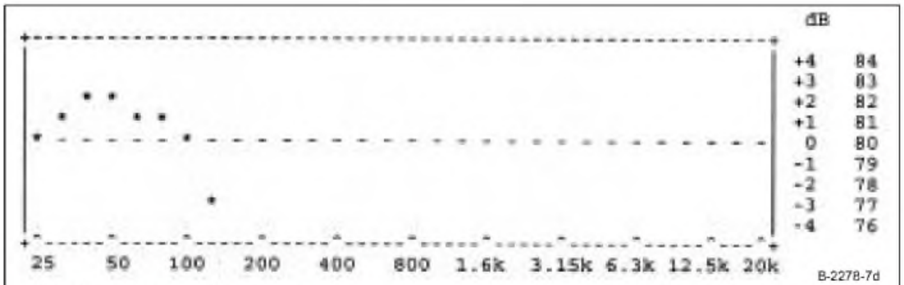


FIGURE 7D: Subwoofer closed-loop response to pink noise measured with accelerometer.

EVALUATION

The last question is, will this thing reproduce music? Is it an improvement? The answer is a definite yes to both questions.

I set up a pair of satellite speakers and hooked up a CD player. I played a number of jazz discs through the system and tested it with and without feedback. As I had hoped, closed-loop bass was much tighter and more defined than open loop.

As a reality check, I asked a friend over to audition the system in my basement. When he saw the maze of wires, he said, "This is a mess!" I played

Steely Dan's *Two Against Nature* CD and turned the feedback on and off.

My friend, John, said that without the feedback, it "sounded like it had an echo." I thought this was interesting since I was using a high-quality woofer in a closed box with a Q of 0.8. It should have sounded tight. The point is that, even to a relatively untrained ear, the "sound" of the servo is much cleaner and more defined than a good open-loop system.

My test rig does suffer from one limitation: it has limited headroom. If I get carried away with the volume control, it makes spitting noises on kick-drum transients. I verified by measurement that this was due to the driver reaching its excursion limits. So even though the driver I chose for my experiments has unusually good T/S parameters, a single 8″ driver doesn't provide enough headroom for reproducing really low bass at high volumes.

If I were building a servo system for my home theater, I would probably use a 15″ driver. A 12″ high-excursion driver would be the absolute minimum. However, in that setup, box size could be objectionably large unless something can be done to mitigate it. Maybe the servo can do just that.

I needed to know how the servo would perform if the box were grossly undersized. If it flattened the response, the implication is that you could put a large driver in a small box as long as you had extra amplifier power to overcome the loss in efficiency.

For this final test, I removed the woofer from the test box and filled the box with wooden blocks, reducing the box volume from 1ft^3 to approximately 0.2ft^3. After this, the closed-box frequency rose to 58Hz. *Figures 8a–d* show the test results.

The subwoofer filter response for this

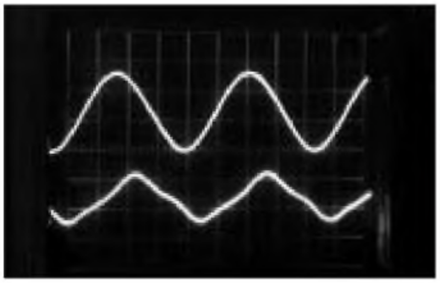


PHOTO 5: Open-loop waveform test.

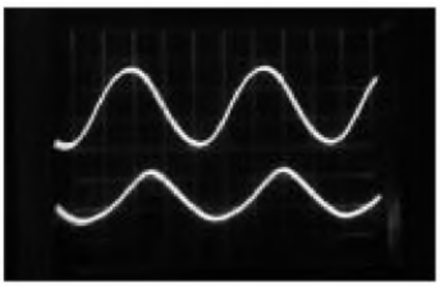


PHOTO 6: Closed-loop waveform test.

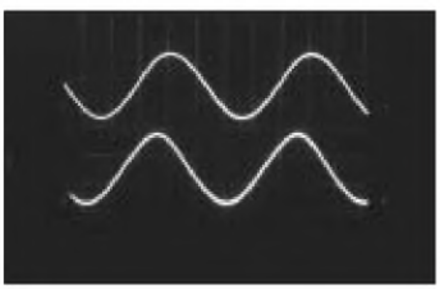


PHOTO 7: Closed-loop waveform test at 30Hz.

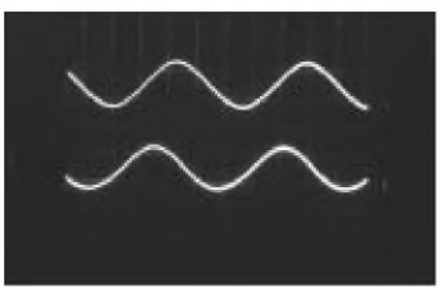


PHOTO 8: Closed-loop waveform test at 20Hz.

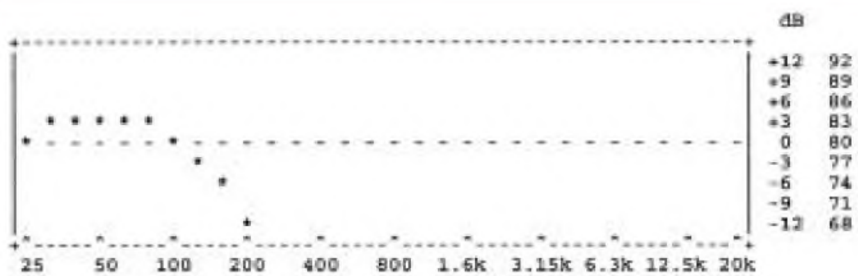


FIGURE 8A: Subwoofer filter pink-noise response.

B-2278-8a

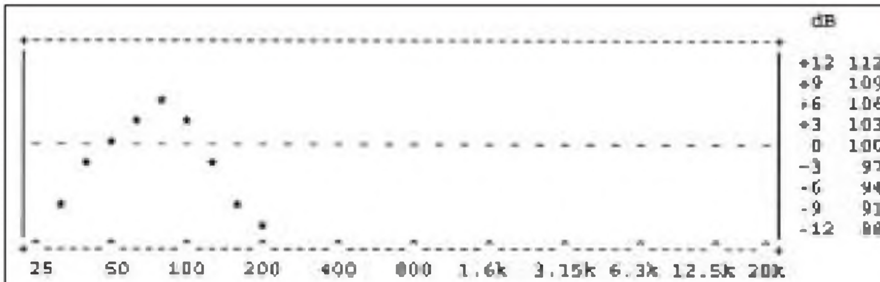


FIGURE 8B: Stuffed box open-loop response to filtered pink noise.

B-2278-8b

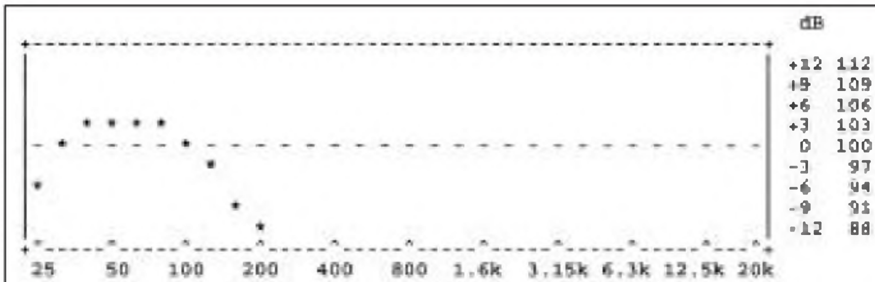


FIGURE 8C: Stuffed box closed-loop response to filtered pink noise.

B-2278-8c

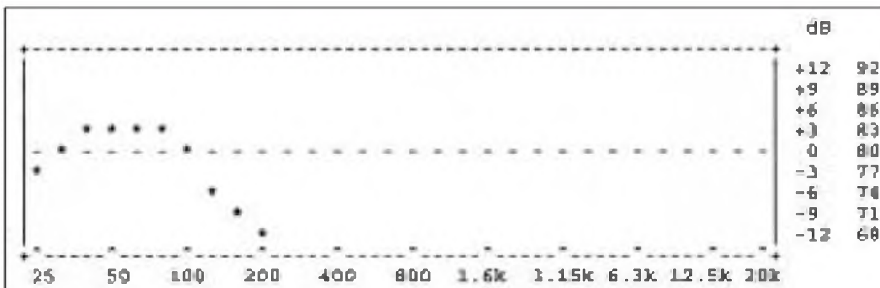


FIGURE 8D: Stuffed box accelerometer response.

B-2278-8d

test is shown for reference as *Fig. 8a.* The SPL open-loop response (*Fig. 8b*) is a "one note" system with a nasty peak. The low-frequency cutoff is somewhere around 70Hz. The closed-loop response (*Fig. 8c*) is a picture-perfect flat response out to about 30Hz. With the stiffened air spring provided by the smaller box volume, distortion correction is outstanding.

Photos 7 and *8* show the closed-loop waveform at 30Hz and 20Hz, respectively. Both appear, at least visually, to be perfectly sinusoidal. From this last test, I am fairly confident that you could construct a compact subwoofer with a large diameter driver that should perform quite well.

That about does it for this adventure. After half a century, the pursuit of good bass is still in progress. In that pursuit, the closed loop woofer has been considered by some to be the "Holy Grail." Maybe it is. ❖

REFERENCE

1. C-Y Chen, G. T-C Chin, C-C Cheng, and H. Peng, "Passive Voice Coil Feedback of Closed Box Subwoofer Systems," *Proc Instn Mech Engs*, Vol. 214, Part C.

Flared Ports

Here's a handy trick to remember when you need flared ports.

By Hilary Paprocki

This construction idea was inspired by Jim Moriyasu's excellent article "A Flared Port Study" (*aX* July '02, p. 28) about flared loudspeaker ports. He showed that sharply terminated ports stress the air passing through them, and at higher levels actually compress the sound level at the port output. Flared ports largely fix this problem, and as to effort, are well within the range of what we audiophiles are willing to adopt for better sound.

It is a common perception that smaller ports are more likely to have compression and noise problems. Some speaker designers just say, "Don't use 'em." But if you're looking at building a pair of Lynn Olson ME2s, or some other speaker design that calls for 1″ ports, you're probably thinking about how you can fix that situation. Piece of cake.

PRETTY FLARES

Just install the port outlet flush with the cabinet panel in the normal way. Grab a ½″ radius corner rounding router bit, screw the guide bushing to the bottom of it, and take a spin around the ID of the port. You'll make the prettiest little flare you've ever seen.

As for the inside end of the port, it's just about as easy. A small piece of scrap wood will make plenty of face area for the inner flare. *Figure 1* shows what it looks like in scale if you put a 3″ piece of half-inch material on a 1″ PVC pipe and cut it with a ½″ rounding bit. You can see from *Fig. 1* that there's plenty of mating surface to hold the parts together. It doesn't even really need to be this big. The slug from a 2.5″ hole saw would make a nice piece for you.

Remember that a flared port must be a little longer than an abrupt one. I'll let the people who understand math tell you how much, but I think that if you cut ½″ radii, you can add ½″ to the total port length. ❖

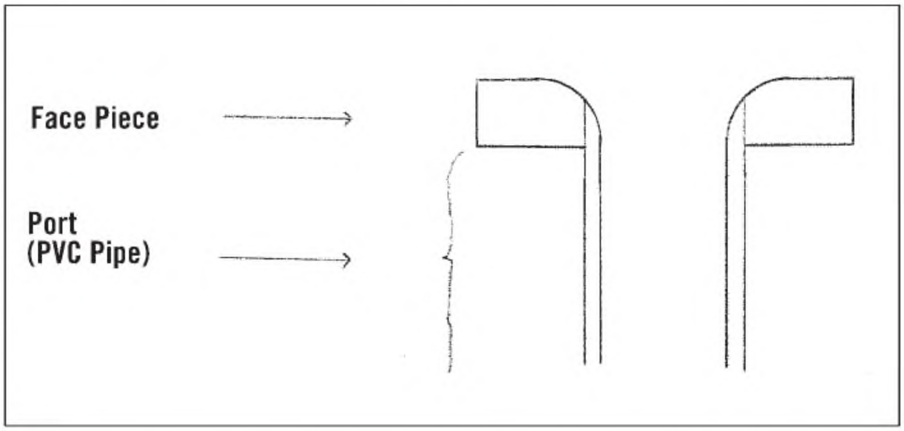


FIGURE 1: Cross-section of flared port.

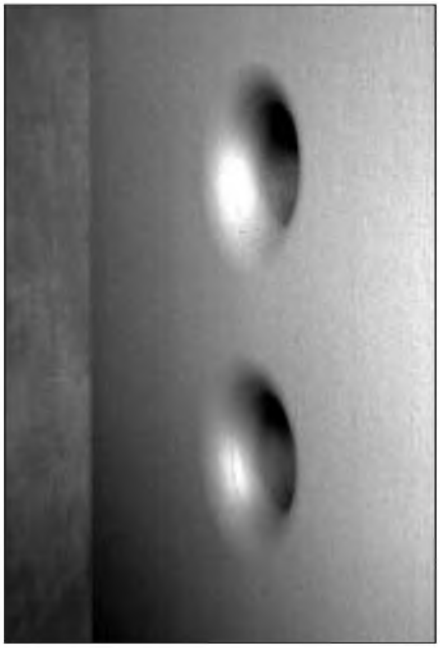


PHOTO 1: Inner end block.

PHOTO 2: Outside of cabinet.


Transformers & Enclosures
In Stock...
Torodial Power:
13 sizes (15 - 1500 VA)
6 VAC - 240 VAC
"Classic" Tube Output:
Single Ended (up to 75 watts),
Push-Pull and Potted (up to 280 watts)
Enclosures:
Aluminum & Steel Chassis,
Diecast Aluminum, Plastic
and Rack Mount
Filament & Low Voltage
Power Transformers:
Open & Enclosed
"Classic"
High Voltage Transformers:
Plate, Plate/Filament Combo
& Filter Chokes (Open & Enclosed)
Contact us for free catalogs & a list of stocking distributors
HAMMOND
MANUFACTURING TM
256 Sonwil Dr. - Cheektowaga, NY 14225 USA
Phone: (716) 651-0086 Fax: (716) 651-0726
394 Edinburgh Rd., N. - Guelph, Ontario N1H 1E5 Canada
Phone: (519) 822-2960 Fax: (519) 822-0715
www.hammondmfg.com

Amplifier Burst Testing

Test the power output of your amp with this simple test-tone circuit.

By John L. Stewart

When your amplifier is driven hard, in most cases its power supply will sag. There are few exceptions. Even a Class-A amplifier pulls a little more current at full output than it does at no signal. As you progress into Class-AB and beyond, the output stage will need more current and your power-supply voltage will drop even further. Because of cost it is customary that only instrumentation amplifiers have a regulated power supply.

Yet, one of the most commonly used checks tests amplifiers at full single-tone CW (continuous wave) output, usually 1kHz. That's unrealistic because no sane person would last long listening in that kind of environment. If your ears didn't fail, then your loudspeaker probably would. If your program material has a 60–80dB (or whatever) dynamic range, then what can you do to get some measure of the real output (headroom) available? What is the output capability of the amp for a short burst?

That's a realistic test because that is how most of your program material is available. There are several ways in which you can produce a tone burst so that you can test your amp in this mode. This article reveals one of them.

This simple gate circuit allows you to apply tone bursts to the amp in test. The signal originates in your existing audio generator. You can set the gate to allow a few cycles of the test tone through and then block the signal. Repetition rate of the tone bursts is set at about 7Hz but could be varied. Now you can make power output measurements while full power supply-voltage is applied to the amplifier. You will need an oscilloscope to observe and measure the test results.

The test works with any amp, whether it is solid or vacuum state. It is possible to build two versions. The simpler depends on your scope having a sweep gate connection, usually found on the rear panel. If that is not available, you can add a three transistor gate driver with synchronizing of the gate to the audio generator source. Nothing is wasted.

You can do all this for less than $100. I built mine in a Hammond 5 × 13.5 × 2 chassis so that it fits right under my scope, where it can stay. Many of the parts came straight out of my junk box!

THE GATE

The gate itself is nothing more than a pair of back-to-back connected Hammond interstage transformers, switched by a pair of diodes (*Fig. 1*). One of the connectors on the rear of my scope is a positive pulse in time with the horizontal sweep. The sweep gate pulse drives the diodes D1 and D2 into conduction and the test tone passes through to the output terminals. Diodes in my final version are very old 1N478s, mostly because I had some. They are germanium, so I thought they might work better because of the low forward drop. I did try a variety of other diodes as well.

The silicon power diode series 1N400X works almost as well. I inserted a 3V reverse bias (two "AA" cells) into that lead so that the tone won't leak through while the sweep gate is absent.

The DPDT switch S3 allows you to select either continuous or burst signal mode. As shown it is in the burst position. The switch section S1c is part of the on-off switch, the rest of which appears in the gate driver schematic (*Fig. 2*). The 4PDT switch S2 allows the test set to be completely bypassed.

Because the circuit is working with a switched signal, some ringing occurs in

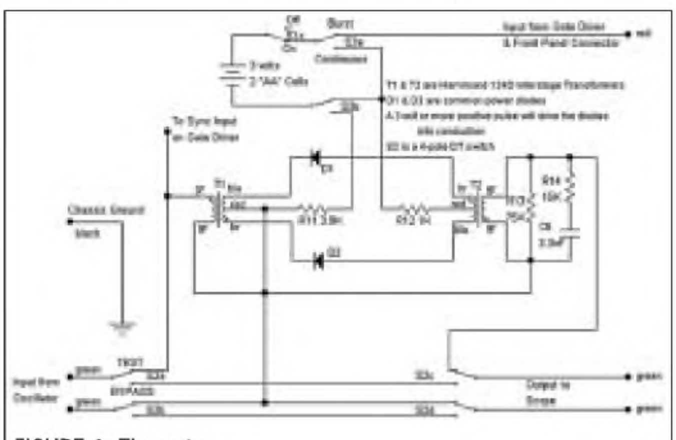


FIGURE 1: The gate.

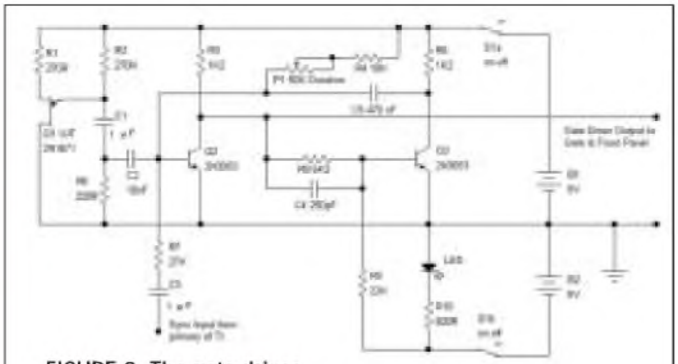
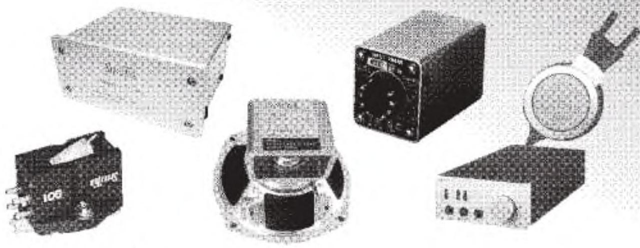


FIGURE 2: The gate driver.

Look!!

Click http://www.eifl.co.jp
or send e-mail to info@eifl.co.jp
for more products information



■Cartridge

Model	Price (US$)
Denon DL-102 (MONO)	150
Denon DL-103 (STEREO)	200
Denon DL-103R (STEREO)	250
Denon DL-103 PRO (STEREO)	350
Shelter Model 501 II (CROWN JEWEL REFERENCE)	750
Shelter Model 901 (CROWN JEWEL SE)	1,400

Postage (Air Economy)	
Area I $18	China, Korea Hong Kong Taiwan
Area II $22	Singapore Malaysia Indonesia
Area III $27	North America Oceania Europe
Area IV $34	Africa South America

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

■Japanease Audio Book

Postage $15

Title		Price(US$)
Attractive Tube Amps Vol. 1&2	(Isamu Asano)	30 each
The Joy of Vintage Tube Amps 1&2	(Tadaatsu Atarashi)	30 each NEW
Direct & Indirect Tube Amps	(Kiyokazu Matsunami)	40 NEW
SE Amps by Transmitting Tubes	(Kouichi Shishido)	50
The Remembrance of Sound Post	(Susumu Sakuma)	30
Classic Valve	(Hisashi Ohtsuka)	40
MJ Selected 300B Amps	(MJ)	30
Top-Sounding Vintage Power Tubes	(Stereo Sound)	30
Output Trans of The World	(Stereo Sound)	30
20TH CENTURY OF AUDIO	(Stereo Sound)	30
Vintage Speaker Units	(Stereo Sound)	30 NEW
Tube Amp Craft Guide	(MJ)	30

■MC STEP UP TRANS

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

** Air Economy

■STAX

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

■Speaker

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

*Price is for a pair **Air Economy

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

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	Postage** I	II	III	IV
XE-20S (SE OPT)	20	2.5, 3.5, 5	20Hz~90kHz	300B,50,2A3	396	47	56	84	113
U-808 (SE OPT)	25	2, 2.5, 3.5, 5	20Hz~65kHz	6L6,50,2A3	242	42	50	73	98
XE-60-5 (PP OPT)	60	5	4Hz~80kHz	300B,KT-88,EL34	620	62	74	115	156
FX-40-5 (PP OPT)	40	5	4Hz~80kHz	2A3,EL34,6L6	320	47	56	84	113
FC-30-3.5S (SE OPT) 〔XE-60-3.5S〕	30	3.5	20Hz~100kHz	300B,50,PX-25	620	62	74	115	156
FC-30-10S (SE OPT) 〔XE-60-10SNF〕	30	10	30Hz~50kHz	211,845	620	62	74	115	156
X-10SF 〔X-10S〕	40	10W/SG Tap	20Hz~55kHz	211,845	1160	90	110	180	251
NC-14 (Interstage)	—	[1+1 : 1+1] 5	25Hz~40kHz	[30mA] 6V6 (T)	264	30	40	50	70
NC-16 (Interstage)	—	[1+1 : 2+2] 7	25Hz~20kHz	[15mA] 6SN7	264	30	40	50	70
NC-20F [NC-20] (Interstage)	—	[1 : 1] 5	18Hz~80kHz	[30mA] 6V6 (T)	640	42	50	73	98
NP-126 (Pre Out)	—	20,10	20Hz~30kHz	[10mA] 6SN7	264	30	40	50	70

Price is for a Pair

** Air Economy

■TAMURA TRANS (All models are available)

Model	W	Pri.Imp(kΩ)	Freq Response	Application	Price (US$)	I	II	III	IV
F-7002 (Permalloy)	10	3.5	15Hz~50kHz	300B,50	836	60	70	110	145
F-7003 (Permalloy)	10	5	15Hz~50kHz	300B,50	836	60	70	110	145
F-2013	40	10	20Hz~50kHz	211,242	786	70	84	133	181
F-5002 (Amorphous)	8	3	10Hz~100kHz	300B,2A3	1276	65	80	120	160

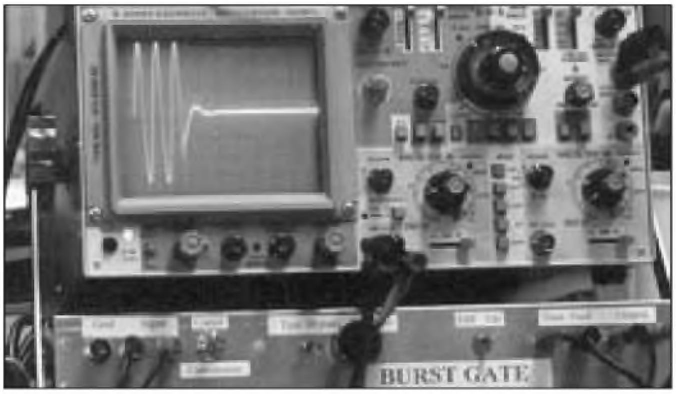
Price is for a Pair

** Air Economy

E·I·F·L EIFL Corporation

1-8, Fujimi 2-chome, Sayama City, Saitama Pref. 350-1306 JAPAN.
Phone:+81-(0)42-956-1178 FAX:+81-(0)42-950-1667
E-mail:info@eifl.co.jp
Wire Transfer:MIZUHO BANK, SWIFT No.MHBKJPJT a/c # 294-9100866

Card Charge 6%



◀PHOTO 1: Scope.

PHOTO 2: Burst front.

the transformers. For the most part this is damped out by the network formed by R13, R14, and C6. If your scope has a negative sweep gate, then you could reverse the diodes and the 3V battery.

GATE DRIVER

If your scope doesn't have a sweep gate or you would like a more comprehensive piece of test equipment, you can drive the gate in a number of different ways. Here is how I did it, mainly because I had these parts in my stockpile (*Fig. 2).*

PARTS LIST

C1	1µF
C2	10nF
C3	1µF
C4	250pF
C5	470nF
C6	3.3nF
R1	270R
R2	270k
R3	1k2
R4	10k
R5	1k2
R6	220R
R7	27k
R8	8k2
R9	22k
R10	820R
R11	3k9
R12	1k
R13	75k
R14	15k
B1, B2	9V
B3	2X "AA" cells and holder
D1, D2	1N400X or see text
D3	red LED
P1	50k pot and knob
Q1	2N1671 (see www.americanmicrosemi.com)
Q2,Q3	2N3053 NPN
S1, S2	4PDT switch
S3	DPDT switch
7 binding posts	
various terminal strips, machine screws, and nuts	
T1, T2	Hammond 124D
Chassis	Hammond 1441-18 (steel) or 1444-18 (aluminum)
Bottom	Hammond 1431-18 (steel) or 1434-18 (aluminum)

All resistors are 1/2W.
All caps are low voltage.
Most parts are available from Antique Electronic Supply, www.tubesandmore.com, and Radio Shack.

The gating pulse is provided by a one-shot multivibrator (MV) consisting of a pair of 2N3053 NPN transistors. However the circuit is not critical, so you can probably use any common NPN transistor here. The multivibrator in turn is triggered by a 2N1671 unijunction transistor. Unijunctions were at one time fairly common, and I found them quite useful. However, they seemed to have disappeared, for the most part, from the market.

The duration of the gate is determined by setting P1, the 50k pot. With P1 set to minimum, the duration is long enough that about three cycles of a 1kHz test tone get through. The gate signal will probably not be synchronized with the audio source, so I have included a connection through C3 and R7 to help stabilize the scope display.

You can also trigger your scope with the signal available from the collector of Q2 and identified on the schematic as the gate driver output. I used a red binding post in order to differentiate from the other front-panel connections. An example of the output burst is shown in *Fig. 3.*

THE UNIJUNCTION

For those who are interested, a unijunction transistor is just that—only one junction, not two as in a regular bipolar. The base is a bar or intrinsic material with connections at each end labeled B1 and B2 (base connections one and two). Ordinarily, the base has a resistance of a few kΩ between its ends, so that little current can flow. The emitter junction is placed part way in from one end of the bar, usually closer to connection B2.

When the bar is supplied with a voltage source, a potential gradient will result along its length. In this case 9V has been used. Not much happens until the 1000nF capacitor C1 charges up to a voltage a bit greater than that which results on the bar where the emitter is attached. As soon as the emitter base junction is forward-biased, current flows and discharges C1 into the base of Q2. That way the one-shot MV is triggered and a gate pulse occurs. The CR time constant I used results in about seven pulses per second.

There are several references to the Unijunction on the web. Refer to http://www.americanmicrosemi.com/tutorials/unijunction.htm for a very good tutorial page and parts source if you decide to use a unijunction in this project.

TEST SETUP and MEASURMENT

The hookup to test your amplifier is very straightforward. The test tone from your audio generator is set to about 1kHz and is not critical. The test set is connected between your tone generator and the

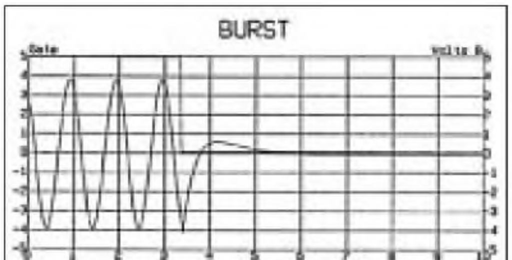


FIGURE 3: Example of output burst.

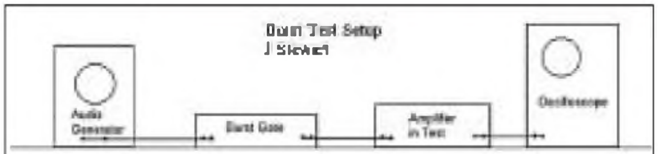


FIGURE 4: Burst test setup.

amplifier you wish to test (*Fig. 4*).

Output from the amplifier goes to the vertical input of your scope. With the test set switched to the ON position there are a number of possibilities. First of all, with switch S2 set to BYPASS you can route your test tone straight through to the amplifier. Alternatively, you could set switch S2 to TEST and to allow a few cycles of the test tone to pass. By varying the position of pot P1 you will be able to pass more or fewer cycles of the test tone.

I wanted to make valid comparisons of amplifier performance both with and without the test tone gated. Because the test set has a bit of attenuation, I have included switch S3, which can set these conditions. You will notice the attenuation if you look at the output of the gate when the BYPASS mode is compared to the CONTINUOUS mode.

The gate introduces about 2% distortion into the signal, but for the intended application this is irrelevant. I included the BYPASS mode so that you could leave the test set connected to the rest of your setup, without having to worry about the gate's residual distortion and attenuation.

Measure the signal amplitude with your scope, much as you would normally do. You are looking for maximum signal output from the amplifier at the clipping point, comparing the BURST and CONTINUOUS modes (S3).

Table 1 shows some results I measured which are fairly typical of amplifiers, especially those running Class-AB and have a power supply with a capacitor input filter. This particular amp is push-pull 6V6GTs running in Class AB2. A 6BQ7 drives the output grids into conduction. The power supply is a small Hammond device rectified by SS diodes and into a capacitor. The test results are quite eye-opening and would not be obvious by other test methods.

When tested by the normal CW method the scope trace had a maximum amplitude of 17.5V at clipping. That translates to 12.4V RMS. The load resistor used measured 7.85Ω cold, so the output was 19.5W.

When tested using the gated tone burst the maximum amplitude measured at clipping on the scope was 20.2V. That translates to about 26W! Another advantage of burst testing is that your load resistor doesn't change value due to heating while in the burst mode.

CONSTRUCTION

I built my burst tester in a small aluminum Hammond chassis which easily fits under my scope. I installed a pair of 8–32x 1.25″ machine screws with locking nuts in the bottom plate, so that the front face would be tilted up. These are placed about 1.5″ to one side of the long dimension of the bottom plate center line with their heads facing down. You may choose some other method to improve access.

TABLE 1

CONDITION	POWER SUPPLY VOLTS	PLATE CURRENT mA	POWER OUTPUT WATTS
No Signal	342	60	zero
Burst	341	61	26.1 burst
CW	281	142	19.5

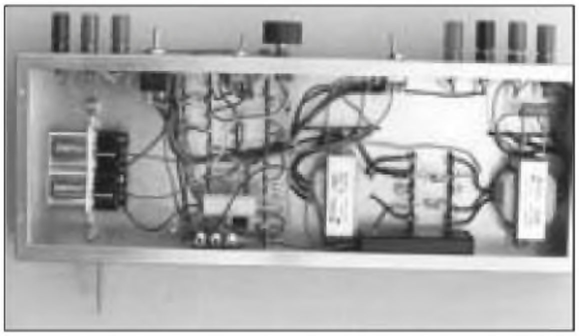


PHOTO 3: Burst bottom.

The pair of double "A" cells (B3) is enclosed in a Radio Shack holder. I used two-sided sticky tape to mount the assembly to the inside of the chassis. If you use alkaline cells at this point, they will probably last for at least five years, their normal shelf life, because the current requirement is minimal.

The two 9V batteries are simply held in place by a short length of #14 solid copper wire with the insulation still attached. The wire is secured to the chassis with 6-32 screws. Again, if you use alkaline batteries, they should be good for 50 to 100 hours' operation in this circuit.

You will enjoy using this simple piece of test equipment and probably be surprised at some of the results. ❖

More on Heater Problems, Tube Flash, and Electron Emission

This article will enlighten you about heater flash, how electron emission works, "sleeping sickness," and the development of high-purity cathode nickel—a breakthrough for tubes. By Bernard Magers

Mark Kelly's excellent article, "Direct Heaters: A Guided Tour" (*audioXpress*, Jan. '03, pp. 28–35) was of much interest to me, and I would like to add some information to this subject.

"Heater" and "filament" are terms often used interchangeably and should be clarified. In tube manufacturing, a heater is usually a tungsten wire, with a passive coating (aluminum oxide) for insulation and inserted inside a cathode sleeve to provide heat.

A filament is a nickel alloy wire coated with an active coating (barium carbonate) to provide emission. A thoriated tungsten filament is made active from thoria in the tungsten. Wire in a light bulb is a filament because the intent is to provide light, not heat. Resistance lamps and ballast lamps have filaments to provide resistance.

In short, for electron tubes, heaters heat; filaments emit.

HEATER FLASH

Heater flash occurs when the uncoated portion of the heater leg, under the cathode, momentarily reaches incandescence during an initial high-current surge through the cold tungsten wire. The coated heater legs are in close contact with the inner walls of the cathode sleeve. Due to conduction and radiation, they heat up more slowly than the uncoated portion. As a result, a leg may flash. Although a heater is a current limiter, it will take several seconds to do so. It may not cause a problem, but for complete protection, a current-limiting supply is recommended.

In the 12AX7, for example, normal heater current is 150mA. During the initial surge through the cold wire, current can reach almost 1A (940mA). As the resistance increases, the flash is extinguished. Not all tubes flash, and the flash may not repeat. Flash depends on tube type, heater design, surge current, wire size or thermal mass, heat treatment, conduction, and radiation. Straight or coiled heaters are both subject to flash.

The heat distribution along a heater—whether it is straight or coiled—is not uniform. The hottest part of the cathode is the middle one-half. The upper and lower one-fourth have a decreasing temperature gradient due to the heatsinking of the mica supports. There is also a sharp gradient from heatsinking where the leg is joined to the heater connector. It is cold near the connector, but heat starts to resume redness about ⅛″ away from it, causing a considerable stress point where the flash is more likely to burn through the wire.

HEATER COATING

Heaters must be insulated to prevent shorting of the legs and electrical contact with the cathode. Alumina coating is best and may be applied by drag, spray, or cataphoretic methods. Alumina 38-500 is a typical ingredient having a particle size of 15–18 microns. A coating suspension contains approximately one-half Alundum, one-fourth aluminum nitrate, and one-fourth water.

Coating is built up on the wire to about 2 mils. After preliminary baking, the wire passes through a wet hydrogen furnace at 1650°C. Hydrogen is the only gas not reactive with tungsten.

Before coating, coiled heaters were wound around a wire mandrel. The finished coil was treated in a chemical bath, dissolving the mandrel out of the coil. Bits of undissolved mandrel have been known to remain in the coil, shorting out several turns and raising havoc with the heater current.

HEATER PROBLEMS

When the heater is positive, breakdown of the coating by electrolysis can cause negative oxygen ions to be drawn to the tungsten, forming aluminum tungsten oxide capable of dissolving the coating and lowering the resistance. After time, the insulation breaks down and the wire fuses to the sleeve, causing a heater-to-cathode short. Small amounts of impurities can be responsible for substantial leakage currents. Just a fraction of a particle of carbon can be very offensive. If the bare legs are in close proximity, a short may occur, momentarily causing a catastrophic flash. A bare leg touching the rim of the cathode sleeve may burn out the cathode tab or cause a heater-to-cathode short.

A standard test for leakage was to apply 100V between heater and cathode. Leakage current may not exceed 10µA. A high leakage path can burn a hole in the coating. Wrong gas pressure may cause an arc and strip coating or burn out a leg. The pressure of the spot-welding electrode can split the wire in half longitudinally and cause a burnout. The presence of cracks, seams, or fissures is most harmful to reliability.

To test reliability, heaters were cycled. A typical heater cycling test placed ten tubes on a test rack and turned the heater on and off for two minutes for 2000 cycles. Heater voltage was raised to 7.5V on a 6.3V heater, and 100V was applied between heater and

ABOUT THE AUTHOR
Bernard D. Magers is the author of *75 Years of Western Electric Tube Manufacturing*. Before retirement, he was a senior engineer for Western Electric, and responsible for the design of the 430C cold-cathode tube, and assisted in design changes in the 374A and 425A tubes.

cathode. One failure required a second sample of 30, and another failure would cause the lot to be rejected.

TUNGSTEN

Tungsten is a refractory metal—stubborn and hard to manage and difficult to fuse, reduce, or work as a metal. It has the ability to retain its physical shape and identity when subjected to very high temperatures. It has a high melting point, mechanical strength, and ductility, to a greater degree than other metals and can operate at temperatures from 2000° to 3000°C. It is not possible to melt tungsten and pour it into molds, so powdered metallurgy techniques are used to produce bar ingots.

Thoriated filaments are prepared by adding 1 to 2% thoria—in the form of a slurry of tungstic acid in water—to the tungsten powder. The slurry is reduced to a metal by hydrogen heat treatment while the thoria is uniformly distributed through the powder.

The powder is compressed under pressures of 25 tons/in^2 and then sintered in hydrogen at 1000° to 1250°C into porous bars. Further sintering is done by passing several thousand amps, raising temperatures close to the melting point. After cooling, a coarse crystalline structure is formed, making it strong but brittle. It is made ductile by hammers striking blows in all directions at 1500°.

The tungsten grains are elongated in an axial direction, resulting in a fibrous structure. The wire can then be drawn with tungsten carbide or diamond dies lubricated with graphite. The completed filaments are "flashed" in vacuum as high as 2200°C for about 40 seconds.

Since the melting point of thorium is 1850°C, any thorium on the surface of the filament evaporates. By reducing the temperature slightly for 30 minutes, metallic thorium in the interior of the filament is given a chance to diffuse to the surface and form a monatomic film. The filament may then be operated at 1700°C, yielding an emission of approximately 3A/cm^2.

Fibrous tungsten will start to grow crystal grains when heated above 1000°C. The size of the grains will increase with increasing temperature, and large grains are formed between a critical range from 2600° to 2800°C. They may extend across the full diameter of the wire, and the crystal boundaries may slip at right angles to the axis of the wire leading to hot spots and burnouts. Tungsten used for filament springs is not heat-treated to prevent loss of ductility.

Consideration should be given to the pumping and aging cycles. For example, 407A, a dual triode like 12AX7, was pumped on a 16-head sealex machine at an index speed of 480 tubes/hour. Initially, glass sealing raises tube temperature, and RF coils heat the plates to 840°C. Heater current is brought up in steps during pumping to 36% above nominal and then to 52% (nominal is 40V, 50mA). This combined heat is 1200° to 1300°C.

During 4½ hours of aging, two "hot shots" from 0 to 60V are given instantaneously, lasting ten minutes each. I mention this to show that some heat treatment of the wire takes place during these steps. The effect on the wire is not clearly defined, but the main concern is that the wire meets final heater-current specifications.

A popular tungsten wire developed by GE for electron tubes was designated

218. It was a "non-sag" wire doped with oxides of sodium, potassium, aluminum, and silicon, permitting the growth of desirable long-grained crystals. Voltage applied to an old, cold heater can snap the wire as it elongates, "squirms" from the heat, or reacts to magnetic forces. Small tungsten wires—up to .030″ in diameter—are identified in terms of weight in milligrams per standard length of a piece of wire 200mm long.

Test specifications for bend support, density, strength, flaws, sag, and electrical parameters can be found in various standards of the American Society for Testing Materials (ASTM) from F213 to F290. Typical suppliers of tungsten were the Elmet Division of Philips, GE, Sylvania, and Westinghouse.

HIGH PURITY NICKEL—A BREAKTHROUGH FOR CATHODES

The cathode sleeve was fabricated from selected nickel and coated with a mixture of barium, strontium, and calcium carbonates. A major breakthrough to extend tube life was the introduction of high-purity nickel by the Bell Telephone Laboratories during the 1950s. It was defined as nickel containing less than 0.005% by weight of all impurities.

In 1905, Wehnelt discovered that alkaline earth oxides possessed remarkable properties for emission. He used platinum as the first core material. Later, various nickels were produced. They all contained impurities such as aluminum, copper, iron, and carbon ranging from 0.05% to 0.35% by weight. At one time, it was believed that emission came from the core material itself but it was non-existent when the coating was removed. It soon became evident that coating and core interacted and core material played an important part in emission characteristics.

During the tube's life, emission slowly declined due to a build-up of interface resistance between the coating and the sleeve. In his book, *Getting the Most Out of Vacuum Tubes*, Robert Tomer refers to this phenomenon as "sleeping sickness." The recommended remedy was to increase the cathode temperature, but the increase in heater voltage shortened tube life.

Not all impurities were detrimental, but silicon was particularly bad. Analysis of the interface showed the formation of barium orthosilicate, an impedance to the electrolytic and chemical reaction between the nickel and the coating. It restricted the amount of free ions, lowered emission, and resulted in a drop of transconductance. An interface can build between coating and sleeve when the tube is either in operation or on standby. (In keeping with Mr. Tomer's human terms, I would say barium orthosilicate was more like the amyloid plaque in Alzheimer's disease!)

Poor coating adherence or peel could also cause interface resistance. It was a separate problem from stripping by low emission space charge and high plate voltage. Peel resulted from a chemical and etching deficiency of the sleeve and was distinguished by early failure—usually less than 1000 hours.

Impurities in the raw material, as well as contamination from the cru-

TABLE 1
PERCENTAGES FOR HIGH-PURITY NICKEL

	CU	MG	FE	MN	SI	TI	C	S
Nickel 220	.10	.08	.10	.20	.05	.02	.08	.008
High-purity	.005	.024	.01	.005	.005	.001	.003	.001

cible during melting and fabrication, produced nickel with a composition that varied over wide limits. It required a "prove-in" procedure where cathodes from a new melt were life-tested against a control melt of known quality to select the best melt available for production. International Nickel's 220 alloy was the most popular.

Performance and component life were directly related to understanding the fundamental physics and chemistry of how much and how long oxide-coated cathodes generate electrons. An investigation to increase tube life was started in the 1950s by Mr. K. M. Olsen of the Bell Labs Metallurgical Research Department. He reasoned that removing the impurities could extend cathode life. The growing need of submarine cable tubes demanded it since undersea work required tube life of 20 years or more to be economical. (The estimated cost to replace a defective tube in undersea equipment was $40,000 to $250,000.)

Between 1956 and 1959, Mr. H. M. Kern of Bell Labs picked up on Olsen's work and developed a process for producing high-purity nickel. It is described briefly as follows: Pure nickel powder was obtained from the Mond Company of England. It was sintered to form nickel slugs by wet hydrogen treatment at 800°C for 16 hours, reducing the oxygen and carbon content to one-tenth of its original value. The sintered slugs were melted down in dry hydrogen in a controlled atmosphere furnace using a clean magnesium crucible.

During melting, a steady flow of hydrogen was maintained at 20ft^3/hr. After a ten-minute period to allow the molten charge to reach a stable temperature of 1500°C, the dry hydrogen was purged from the furnace with dry helium, and all the gases were removed by evacuation. Then dry hydrogen was reintroduced for 15 minutes to effect a further reduction of carbon and oxygen. Helium was reintroduced to establish a flow of 20ft^3/hr, at one atmosphere.

The molten charge at this point was poured into an alundum-coated steel mold. Careful processing provided a nickel of exceptional purity as shown in *Table 1* as maximum percentage by weight.

With high-purity nickel now available, Mr. Kern began a study of this new material for cathode use. He found a pure nickel core was a poor emitter and an impurity had to be deliberately added and rigidly controlled. Although most impurities work well initially, they failed after less than 10,000 hours of use.

He tested various additives on long life and found that minimal interface resistance was obtained with 0.024% magnesium or 0.1% zirconium. Although INCO 220 contained Mg, the level was too high. Bell Labs high-purity nickel with magnesium as a controlled additive was designated CA-519, and the Western Electric equivalent was XN-6, which later became known as grade 6.

Grade 6 was fabricated at the Hawthorne plant. In 1959, the Superior Tube Co. in cooperation with WE introduced high-purity alloy X-3012 containing 0.1% zirconium. Both alloys contained 2% tungsten, which allowed higher firing temperature and greater collapse strength.

Other nickels, provided by Driver-Harris, were Cathaloys A-30, A-31, A-32, P-50, and P-51. Those provided by INCO were 200, 220, 225, 233, 399, and 599. The impurities in all of these alloys were up to 20 times greater than that of X-3012 and grade 6. Therein was the beauty of high-purity nickel.

After fabrication, cathode sleeves were tested for thermionic emission, spectrochemical analysis, tensile strength, hardness, grain size, and physical dimensions.

CATHODE EMISSION—HOW IT WORKS

The cathode was coated with a carbonate or nitrate coating because oxides formed hydroxides in moist air and became poor emitters. The coating was mixed in a suspension of amyl acetate and nitrocelulose as a binder. Coating thickness was typically 0.001″ to 0.002″.

Calcium was added (4%–13%) because it activated quickly and abundantly early in life, but it also faded early. A mixture with the longest life possible was made from Ba and Sr carbonates and was referred to as "double carbonates." In April 1963, I attended a meeting at the WE plant in Allentown, Pa., where it was declared that high-purity nickel with a magnesium additive, combined with double carbonates, was "state of the art."

The cathode was activated by heating while the glass envelope was pumped to a vacuum of 10^{-5}mm Hg. Heating began by raising the heater current to about 140% of its operating voltage and turning on RF coils to heat the plate. At 500°C the organic binder, ethyl alcohol, and amyl acetate were driven off by evaporation. The color of the coating changed from white to dark gray, to patchy white, and then to pure white.

As temperature increased from 750 to 850°C, carbon dioxide was driven from the carbonates and ionized blue in color. The beginning of decomposition was signified by a marked increase in exhaust pressure. When complete, barium oxide was formed ($BaCo_3$-Co_2=BaO) and the pressure returned to normal. Gas made coating subject to deactivation by ion bombardment, making vacuum maintenance critical.

During decomposition, the coating material shrunk to a porous structure 20% to 50% of its original volume. Barium oxide was contained in the coating in the form of ionic crystals. The Mg impurity reacted chemically as a reducing agent with BaO at the interface to form MgO. Mg replaced Ba in the BaO oxide molecule by reducing (removing) the O to a vacant site. The Ba atom was now free to move.

The movement of the Ba atom produced a differential gradient. At elevated temperature, the gradient induced diffusion of a continuous new supply of atoms at the interface. This action plus heat elevated the Ba atom into the conduction band. Each atom contains 56 electrons. Heat gave these electrons sufficient energy to break through the potential barrier and escape, producing a positive charge on the metal.

The kinetic energy an electron needs to overcome the charge is the "work function" of the material measured in electron volts. If the temperature is elevated high enough to emit both atoms and electrons, the material is being disintegrated. You might assume that a low work function is ideal for good emission because the lower the barrier the lower the operating temperature. However, materials with low work functions have high vapor pressures and cannot be operated sufficiently hot to obtain adequate emission.

Mr. Kelly says there are reports of a semiconductor with a work function so low it provides high levels of emission at room temperature and that one day we may have tubes with no heater. The reason we probably haven't seen this yet is because a material with such a low work function would have a vapor pressure so high that it would vanish into thin air.

Electrons attracted to the plate gain velocity until they strike the surface and flow as emission current rated in amps/cm^2 of the coating surface. Impact imparts energy and causes secondary electrons to escape. This nuisance was usually eliminated by a suppressor grid.

ADDITIONAL NOTES

1. The act of emission is referred to as donor production and donor loss. An overproduction of donors may cause a deposit, resulting in grid emission or leakage. Gas, geometry, current density, and electrode voltages affect donor loss rate. During tube processing, foreign particles may be deposited on the plate. They will decompose or ionize from electron bombardment and return to deactivate the cathode.

2. During emission, the BaO acts as an N-type, excess impurity semiconductor. When all impurity atoms have diffused and reacted, the production of Ba donors ceases. If impurity atoms exceed the BaO molecules, the end of life will terminate from the coating and not the nickel.

Speaking of semiconductors, Mr. Kelly chides the tube purist (no semiconductors) about the tube connection. To respond, the tube purist can take some solace in the fact that semiconduction is a natural phenomenon found in tubes and was used over 50 years before the semiconductor device was a gleam in the inventor's eye.

3. The reaction that produces Ba is limited by the rate which impurity atoms diffuse through the nickel. Different impurities have different diffusion rates, and knowledge of the rates is essential. Nickel with no impurity additives fails because the concentration in the nickel is too low to maintain an adequate donor-production rate versus the donor losses to the plate.

4. Life of the cathode was terminated by a blocking layer at the interface, using up the BaO molecules in the coating, using up the impurity atoms in the nickel, or poisoning the coating by the ionization of gas or foreign particles.

5. "Brimarizing" equipment at STC/Brimar (Great Britain) was the retrofitting of old tubes with new tubes made of high-purity nickel cathodes.

6. Tube life was increased from 10,000 to over 100,000 hours. I have personally monitored a life sample of 408As for 12 years. Even then the sample was not terminated for failures but for a growing need for life rack space. Although high-purity nickel was developed for submarine cable work, the benefits were extended to all Western Electric tubes. ❖

REFERENCES

H. E. Kern, "Research on Oxide-Coated Cathodes," BTL memo, 1960.

K. M. Olsen, "Hi-Purity Nickel," BTL memorandum, 1960.

Walter H. Kohl, *Materials and Techniques for Electron Tubes*, GTE Technical Series, 2nd Ed., 1962.

Bernard Magers, "Memo for Record," April 16, 1963.

Bernard Magers, *75 Years of Western Electric Tube Manufacturing.*

Bernard Magers, "More on Tube Flash and Heaters," *Tube Collector*, December 2002.

Bernard Magers, "Hi-Purity Nickel, A 1950s Break-Through for WE Tubes," *Tube Collector*, April 2001.

Restoration of Peavey TKO 80 Bass Amp, Part 2

With the driver selected, it's time to take a look at circuit operation of this classic amp.

By Charles Hansen and G. R. Koonce

BY CHARLES HANSEN
TKO 80 AMPLIFIER DESIGN (VINTAGE 1980)

The quasi-complementary (QC) output stage, used in a number of musical instrument and early hi-fi amplifiers, including the subject 1980-vintage TKO 80, was patented by H. C. Lin, for RCA Laboratories in 1956. Most modern power amplifiers, especially high-fidelity designs, use complementary-symmetrical output stages, in which matched complementary NPN/PNP are used in an emitter-follower (EF) output stage configuration. (A similar arrangement of N-channel/P-channel power MOSFETs produces a source-follower output stage.)

By contrast, in the QC output stage, the upper NPN driver operates with its NPN output transistor as a Darlington-connected emitter follower during the positive output half-cycles. The lower PNP driver is an emitter-follower inverter driving its common emitter NPN output transistor during the negative input half cycles. This was a necessary evil back when PNP power transistors were greatly inferior to the NPN types. The lower pair configuration is also sometimes called a complementary feedback pair (CFB).

The QC output stage has a number of design problems compared with the EF output stage. The upper pair of transistors has more headroom than the lower pair. The design is also susceptible to negative side "sticking" at high frequencies or during clipping.

The upper transistors operate as emitter followers and cannot saturate, but the lower output transistor, operating common emitter, can saturate. Once saturated, the transistor no longer responds to the base drive until the stored charge is removed. This can cause common mode conduction as the upper transistor conducts into the saturated lower transistor, causing high peak current through both devices until the lower device again responds to the reduced base drive.

With the upper output transistor configured as an emitter follower, it has less than unity gain. But

The restored Peavey TKO 80 bass amp.

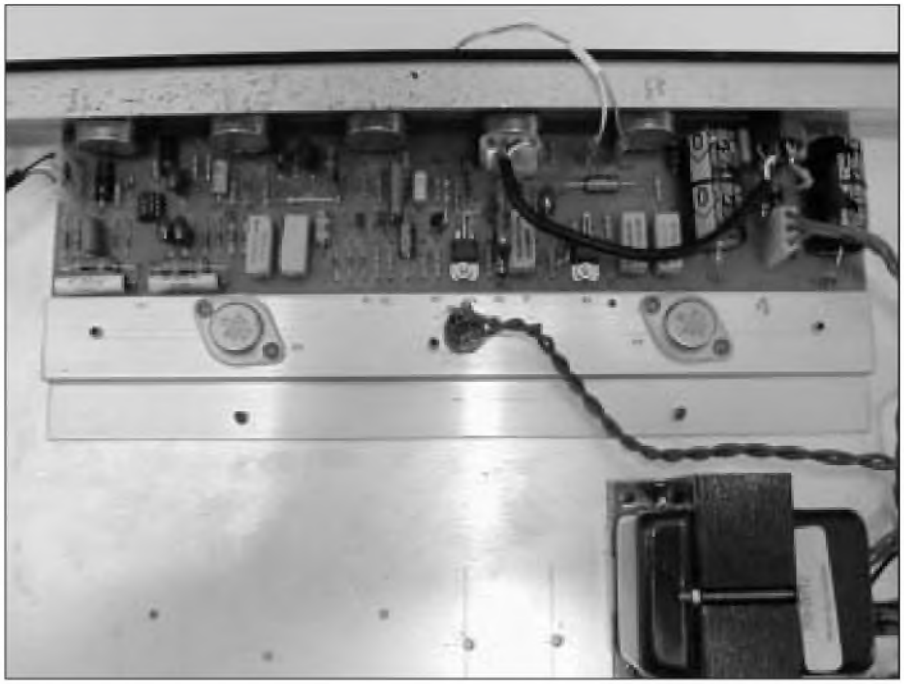


PHOTO 7: Modified amplifier PC board.

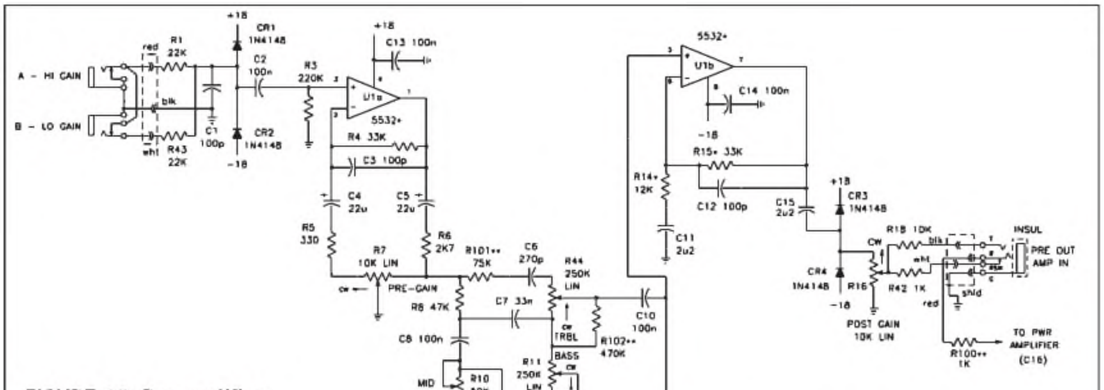


FIGURE 13: Preamplifier section modifications.

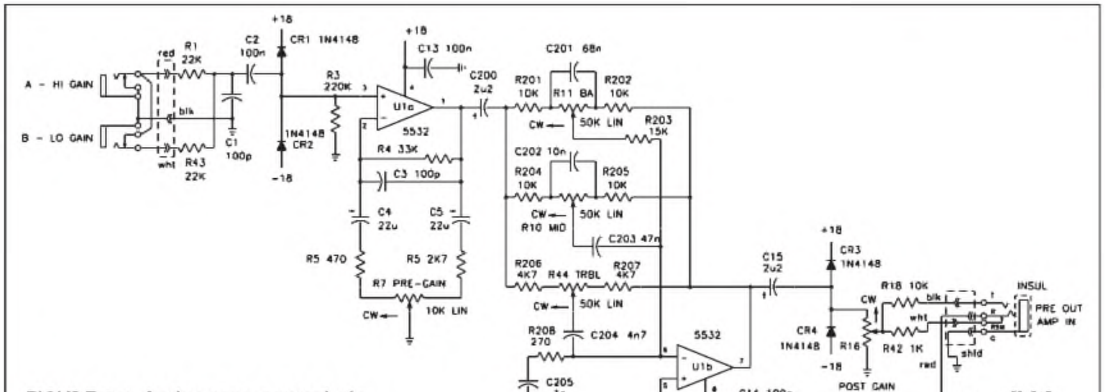


FIGURE 14: Active tone control circuit and response (for information).

the lower common-emitter transistor has the speaker load in its collector and, therefore, has voltage gain and can oscillate at high frequencies. James Bongiorno of S.A.E. found the problem was due to phase shifts within the QC output stage. While this can cause problems in high-fidelity amplifiers, it historically hasn't been a major issue for musical instrument amplifiers, given their limited high-frequency response.

However, even for musical instrument amplification, there remains a fundamental problem. The differing input impedances of the two halves of the QC output stage cause greater overall distortion than in a complementary-symmetrical output stage. In the upper QC pair the impedance presented to the voltage amplifier stage is two base-emitter (b-e) junctions, while in the lower pair the voltage amplifier stage sees just one b-e junction, the PNP driver. This results in asymmetry between the impedances of the two output stage halves, especially at low output currents. This asymmetry produces serious crossover distortion.

The two halves also exhibit different

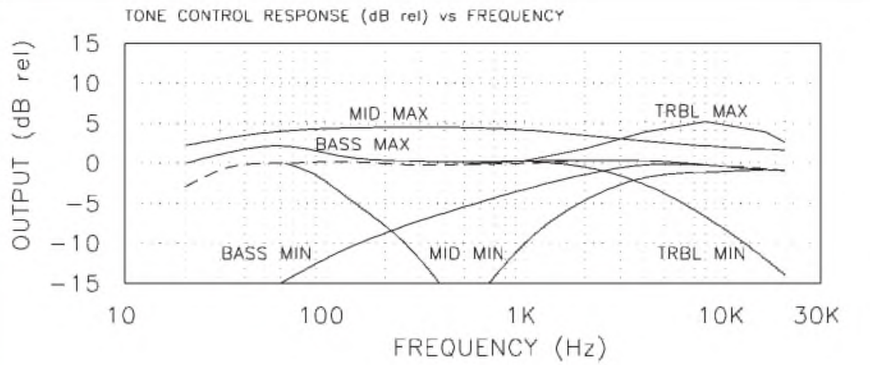


FIGURE 15: Modified preamplifier frequency response and tone control range. A-2248-15

TABLE 8
MODIFICATION PARTS LIST

SYMBOL	DESCRIPTION	QTY	VENDOR
	10″ Peerless 850146	1	PE
	Tolex, black, 54″ × 1 lin ft	3	AES
	3M Super 77 adhesive	1	PE
	Grille cloth, black, 67″ × 1 yd	1	PE
	Acoustic convoluted foam, 1.5″, 24 × 18	1	PE
	Cabinet corner, 2-leg, 1.42 × 0.87 nickel	6	PE
	Cabinet corner, 3-leg, 1.6 × 1.6 nickel	2	PE
	Cabinet feet, 1-1/8 × ½ thick	4	PE
	T-Nuts, 10-32	8	HD
	Machine screws, 10-32 × 1½	8	HD
	T-Nuts, 8-32	6	HD
	Machine screws, 8-32 × 1½	6	HD
	Corner RH screws, stainless, #10 × ¾	16	HD
	Chassis FH screws, stainless, #10-24 × 1½	4	HD
	Chassis screw finishing washers, #10	4	HD
C1, C3, C12, C17	100pF NPO ceramic	4	Mouser
C2, C8, C10, C13, C14, C25–C27	100nF polyester	8	Mouser
C4, C5	22μ 50V, axial aluminum	2	Mouser
C7	33nF polyester	1	Mouser
C9*	3n3 polyester	1	Mouser
C11, C15	2μ2 50V non-polar, axial aluminum	2	Mouser
C16*	680nF polyester	1	Mouser
C19, C20	47μ 50V, axial aluminum	2	Mouser
C21	1nF polyester	1	Mouser
C22*	470pF NPO ceramic	1	Mouser
C28, C31*	100μ 25V, axial aluminum	2	Mouser
C29, C32	4700μ 50V, axial aluminum	2	Mouser
C6	270pF NPO ceramic	2	Mouser
C200**, C201**	470nF 220V polyester film	2	Mouser
C300**, C301**	220pF NPO ceramic	1	Mouser
CR10, CR11*	1N4746A zener, 18V 1W 5%	2	Mouser
CR12–CR15*	1N5402 diode, 3A 200V	4	Mouser
CR300**, CR301**	1N4001 diode, 1A 50V	2	Mouser
Q6*	SK3440/TIP31C NPN transistor	1	Mouser
Q7*	SK3441/TIP32C PNP transistor	1	Mouser
R14*	6k8 5% carbon film	1	Mouser
R15*	22k 5% carbon film	1	Mouser
R27	105Ω 1% metal film	1	Mouser
R32*, R35*, R300**	100Ω 1% metal film	3	Mouser
R100**	1k 5% carbon film	1	Mouser
R101**	75k 5% carbon film	1	Mouser
R102**	470k 5% carbon film	1	Mouser
R301**	1k trimpot 10T cermet	1	Mouser

Notes: *part value change, **new component added transfer characteristics, and it is not possible to minimize the crossover distortion with bias adjustment. The QC output stage has a much lower tolerance of quiescent current to power-supply variation as well.

Doug Self illustrates the crossover distortion problems inherent in the QC output stage with his SPICE simulations[3]. He shows that a major improvement to QC stage symmetry can be made using the Baxandall diode. I. M. Shaw of Wellbrook Engineering Electronics Ltd. first proposed the use of a single diode in the emitter of the PNP driver stage in June 1969. Peter Baxandall (the designer of the bass-treble "Baxandall" tone control circuit) proposed the alternate form in Sept. 1969, with a compensating diode in the bias string.

The Baxandall diode, in conjunction with the PNP driver b-e junction, mimics the Vbe drop in the paired b-e junctions of the upper EF pair. The crossover distortion characteristics are quite close to those of the complementary-symmetrical designs.

I decided to perform the following modifications to the circuitry (*Table 8*).

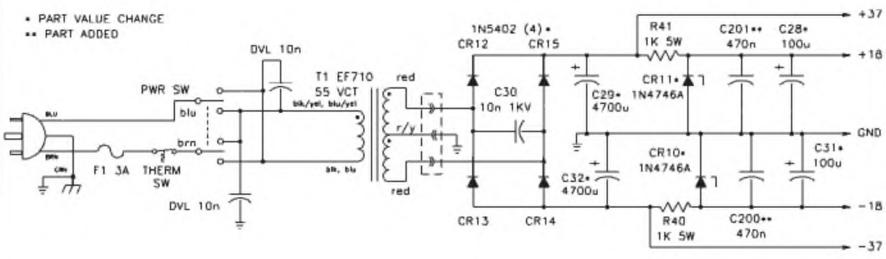


FIGURE 16: Power supply modifications.

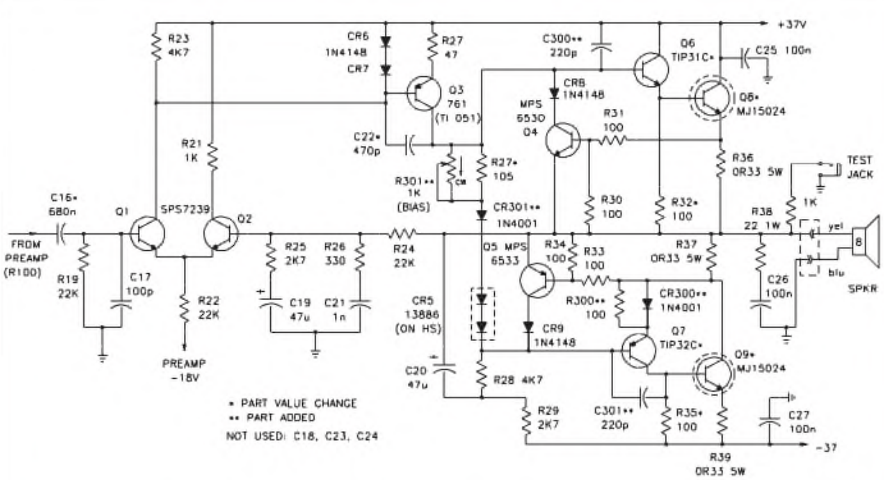


FIGURE 17: Power amplifier section modifications.

ELECTRONICS MODIFICATION DETAILS

The preamplifier section is designed around a dual op amp. Someone had replaced the TL072 in my unit with a NE5532A (*Fig. 13*). In view of the age of the amplifier, I replaced every aluminum cap with a new one. I also replaced many of the small axial ceramic caps in the signal path with new polyester film types of the same value.

As I mentioned earlier, the standard bass/mid/treble controls are passive, and interact like mad. I spent some time redesigning the three-band active tone control circuit in the *1980 National Semiconductor Audio/Radio Handbook*[4] from a "hi-fi" design to one more suitable for a bass amp. The handbook circuit has ±20dB control centered at 1kHz. I lowered the center to about 600Hz and reduced the authority to about ±12dB. SPICE simulations show the response is flat with all three controls centered.

However, when I tried to find a way to conveniently implement this circuit on the original PC board, I thought it required far too much modification. The pots are soldered to the circuit board, so I would either need to cut all the pot tracks to isolate them electrically, or disturb the nice neat one-piece mechanical PC board package. Instead, I made a few changes to the original passive tone control circuit. While I ultimately did not install the active tone control circuit, I show the schematic in *Fig. 14* for those who are interested.

I wanted the response to be as flat as possible with the pots mechanically centered ("5" on the 0–10 silk-screened scales). An hour or so of fiddling with SPICE simulations rendered the changes shown in *Fig. 13.* The treble control was quite peaky in the original design, so I added R101 to limit the boost. R101 and R102 combine to make the treble control response flat at "5."

C9 was 10nF on the service schematic, but I found 15nF was actually installed at the factory. I changed it to 3n3 to make the midrange control flat at "5" and reduce the two dips in response adjacent to its control range (Fig. 6). The bass control was already flat in the center of its rotation. I added R100 to put some series impedance between the rewired preamp out jack and the new power amp section input.

The tone control range for the modified passive tone controls is shown in *Fig. 15.* As you can see, there is a lot more cut than boost available, and U1b is needed to recover the loss of gain through the tone controls. R14 was a high 33k, so I changed R14 to 12k and R15 to 33k for a bit lower noise without loading down the ±18V supply. With the three controls centered, the response, shown in the dashed line in *Fig. 15*, is now flat within ±0.5dB from 25Hz to 15kHz, a big improvement over Fig. 6.

I also modified the power supply (*Fig. 16*), replacing the main 2200μF reservoir caps with 4700μF. There were six holes in the PC board under these caps, so I tied them down with nylon lacing tape to secure them for the rough handling a musical instrument amp sometimes receives. The Peavey schematic lists the main rectifier diodes as 1N4003, but they were actually 1N5393 types. Since 50W into 8Ω requires 2.5A RMS, or 1.25A RMS per diode, these 1.5A diodes were marginal. I used three amp 1N5402s and spaced them about 7/16″ off the PC board for better cooling.

The old preamp section obtained its ±15V supply from the power amp HV rails using 1k 5W resistors and zener diodes. Zeners are quite noisy and the distortion tests showed a fair amount of 60Hz as well (Fig. 3). While it was feasible to remove the resistor and zener and install linear regulators, the ±37V rails were too close to the maximum input voltage limit for IC regulators.

Instead, I changed the zeners to 18V (the 5532 is rated for ±22V) for more input stage headroom. I also changed C28 and C31 from 47μF to 100μF in the same package size, and added film caps C200 and C201 across the zeners to re-

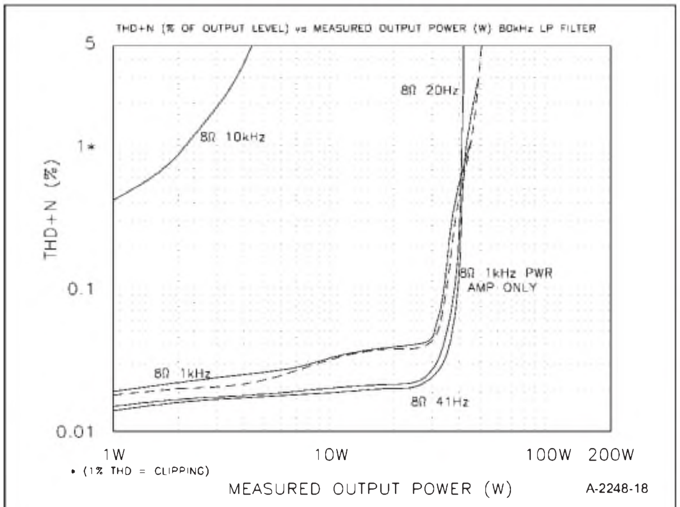


FIGURE 18: Modified amplifier THD+N versus output power.

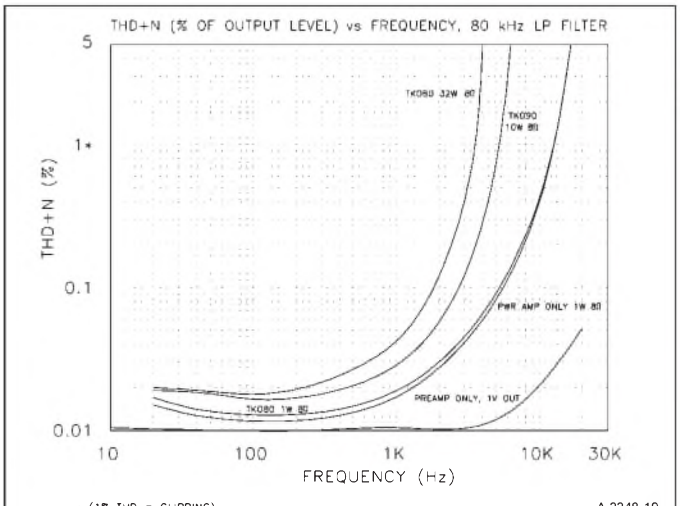


FIGURE 19: Modified amplifier THD+N versus frequency.

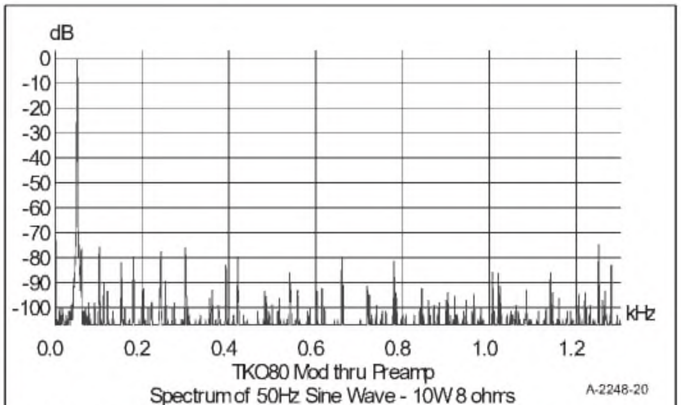


FIGURE 20: Modified amplifier spectrum of 50Hz sine wave.

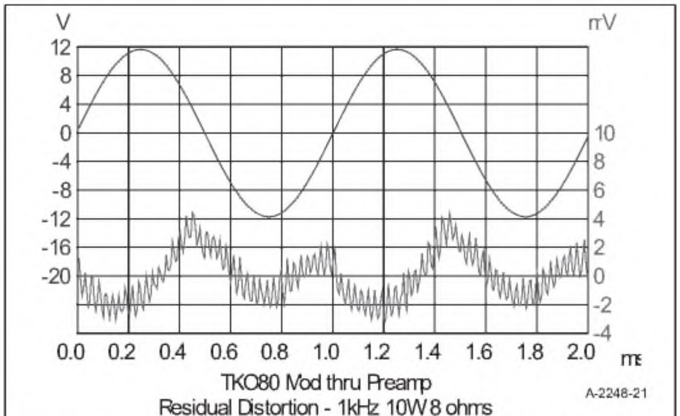


FIGURE 21: Modified amplifier 1kHz residual distortion signal.

duce some of their inherent noise.

In the schematic for the modified TKO 80 power amplifier section (*Fig. 17*), the whole front end is upside down from the usual design: an NPN long-tailed input stage followed by that PNP last voltage amplifier. The only other amp I could find with this topology is the 1970s Gibson SG-812. Interestingly, it used a Baxandall diode.

According to a replacement part research, some of the transistors are now obsolete. Q1 and Q2 had Peavey house number SPS7239, selected devices by Motorola. The Peavey schematic showed them as 2N3904s. This long-tail differential pair uses a single-tail resistor, R22, and thus has poor CMRR and PSRR when compared with a constant-current tail.

The single-tail resistor topology is used in quite a few guitar and bass amps, usually with the tail resistor connected to the power amplifier negative rail supply (or positive rail in the case of PNP input stages) through a diode and resistor, and filtered with an aluminum cap. The TKO 80 design connects the tail resistor to the op-amp negative rail. While it saves the aforementioned parts, it also risks injecting zener noise into the input circuit—not a very good trade-off for less than a dollar's worth of parts. Due to restrictions in the PC board layout in this area, I left the tail resistor alone, hoping for the best with the added film caps.

Test measurements showed the input stage was not DC balanced either. Q2 has twice the current of Q1. This imbalance produces second harmonics, which may well be the intent in a musi-

PHOTO 8: Finished TKO 80 bass amp without grille.

cal instrument amplifier—more tube-like.

Q4 was a standard Motorola MPS6530, which you could also replace with a 2N3904. Q5 was a standard Motorola MPS6533, but you could replace it with a 2N3906. Q3 was labeled with house number TI 051. This was Peavey part #70400761, but equivalent to the 2N5400 according to research.

Q6 (EP430) and Q7 (EP431) were standard TI parts at one time. They weren't listed in the NTE or SK replacement parts indexes, but the TIP31C and TIP32C, respectively, would serve as improved alternates. Before making any modifications to any electronics gear, be sure to identify an alternate replacement part for all the obsolete semiconductors, in case you damage one of them.

I changed the values of a number of components, marked with an asterisk in *Fig. 17.* I increased C16 from 100nF to 680nF to lower the LF –3dB point. R32 and R35 were 47Ω, which I increased to 100Ω. I replaced 0.1μF axial ceramic caps C25 and C27 with 100nF polyester film types. This compromised the room available for the preamp out jack, but it still fit above the power supply diodes and between the reservoir caps.

The added components are marked with a double asterisk. Douglas Self suggested that it would be wise for me to add small caps from base-collector on the two driver transistors to stop HF oscillation, and every other likely place until the amplifier is completely stable. When I modeled these caps using SPICE, I could not detect any change in response. Doug thought these caps might be producing some second-order

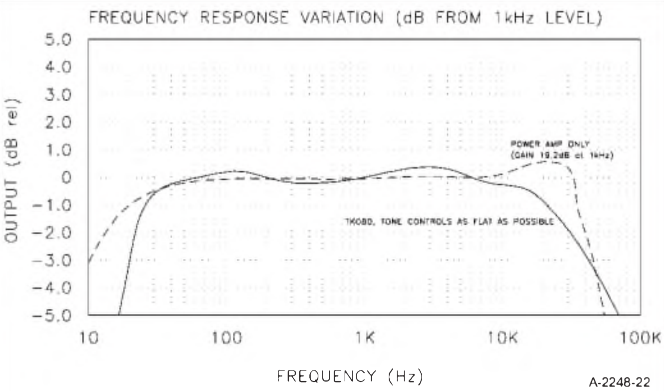


FIGURE 22: Modified amplifier frequency response.

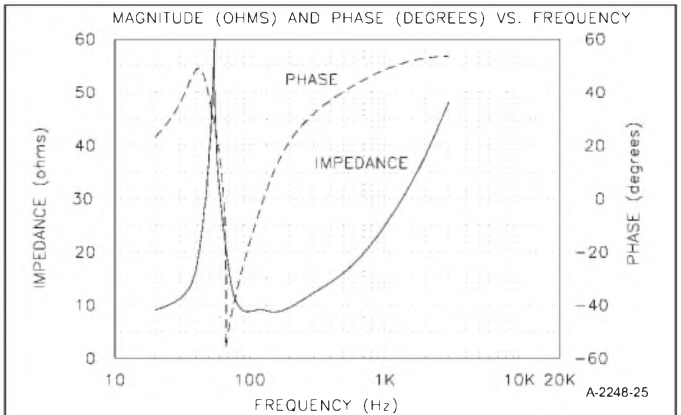


FIGURE 25: QX108 CB electrical impedance and phase.

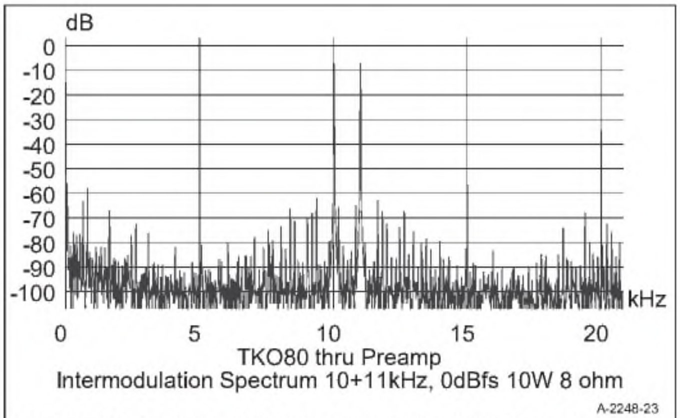


FIGURE 23: 10kHz + 1kHz intermodulation distortion, original.

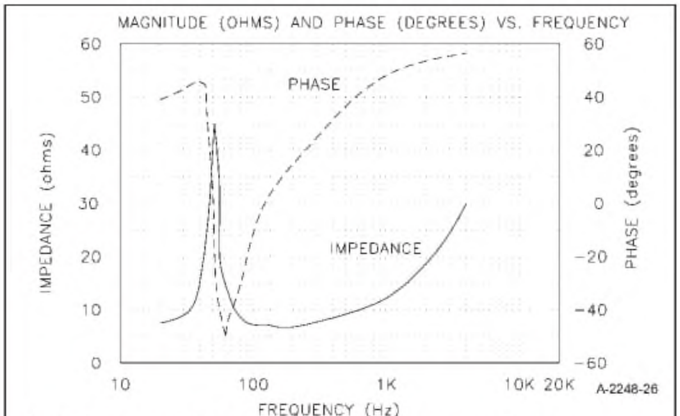


FIGURE 26: Peerless CB electrical impedance and phase.

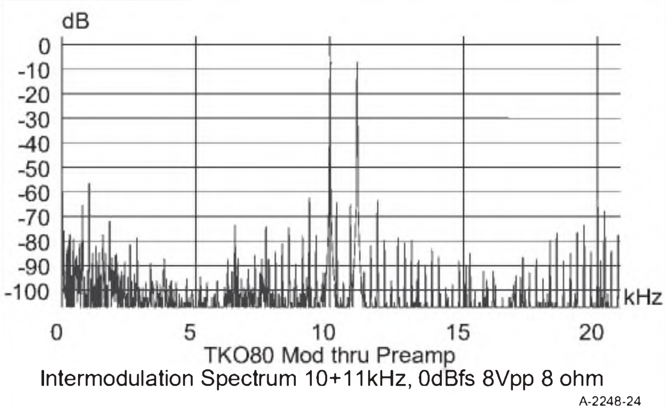


FIGURE 24: 10kHz + 1kHz intermodulation distortion, modified design.

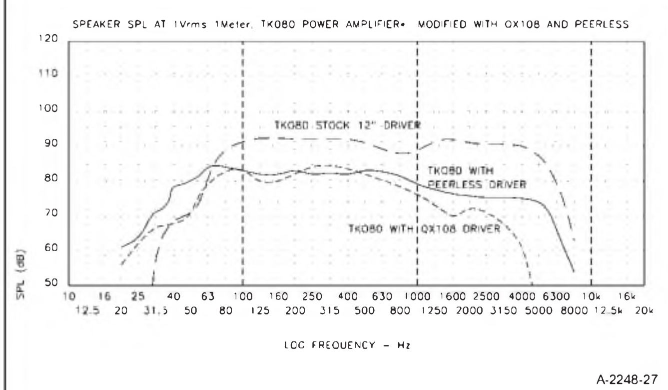


FIGURE 27: Modified TKO 80 loudspeaker frequency response.

effect not modeled by SPICE. In any event, I added C300 and C301 as he suggested.

The 1N4001 Baxandall diode CR300 mimics the upper NPN output transistor b-e junction. A 100Ω metal film resistor (R300), equal to the collector resistor in the PNP driver, is connected across the Baxandall diode. I had to lift the leads of Q7 from the PC board to install these parts and C301.

While I was at it, I tested Q7 and found its h_{FE} was 11, on the low side of the specified 10–50 range. I decided to replace Q6 and Q7 with the TIP devices mentioned previously. The h_{FE} for the ones I received were in the 17–21 range. This would help to unload Q3 a bit.

The Baxandall diode increases the bias voltage requirement by one junction, so I added 1N4001 diode CR301 in series with the bias string. The only way to add CR301 was to lift one lead of R27 and solder it in series.

An added penalty in QC output stage efficiency is caused by the need for the three 0.33Ω power resistors in series with the two NPN output transistors. The complementary-symmetrical output stage requires only two ballast resistors—one in each emitter—for thermal stability. In addition to the emitter ballast resistors, the lower QC NPN output transistor also requires a collector resistor to help make the lower CE output stage mirror the top EF half.

During measurements of the stock driver parameters, there were spikes in the output when the 12″ driver was near its resonant frequency. There are no inductive catch diodes from the speaker output to the supply rails in the TKO 80. Unfortunately, there is also no convenient place to connect them on the circuit board.

The power amp harmonic distortion is now mainly due to the last voltage amplifier (LVA) stage, Q3. The LVA is loaded by bootstrap capacitor C20 instead of a constant-current sink. This makes the low-frequency gain somewhat dependent on the varying impedance of the driver. There is another factor at work here with the QC output stage—the lower driver and output transistor do not have unity gain, and this introduces some asymmetry into the LVA collector loading. The higher gains of the TIP drivers help here.

I tried to eliminate the 60kHz frequency peaking in the power amp section by experimentally changing the value of dominant pole compensation capacitor C22 in the voltage amplifier stage. C22 was originally 100pF, and SPICE simulations suggested it should be 470pF, but the cap value is highly dependent on the gain of Q3, since the value of the b-c capacitor is multiplied by the transistor h_{FE}.

I removed C24, a 1nF cap from the collector of Q3 to ground. SPICE simulations showed it had very little effect on the HF rolloff, so it may have been added for stability, but I believed that C300 and C301, recommended by Doug Self, were a better solution. They theoretically also produce a low-pass pole in the driver stage response, but their effect on the 60kHz frequency peak was negligible. It was C22 that got rid of the HF peaking, with only a minor +0.5dB peak at 20kHz and a –3dB point at 45kHz.

Since the Baxandall change would probably alter the output stage bias, I decided to change R27 to 105Ω and add 1k 10-turn trimpot R301 across it. This would allow me to adjust the output stage bias to minimize the distortion.

I adjusted the parallel combination of R27 and R301 to 47Ω, the original value of R27 alone. Some crossover distortion spikes were still evident, so I increased R301 until they disappeared, which resulted in a parallel resistance of 75Ω. Then I adjusted R301 to its maximum resistance (highest bias current) and let the amplifier operate at 16W (1/3 power) for an hour. The 1/3 power test point I used is supposed to produce the most thermal stress. At the end of this period, the output transistors were hot, but certainly not too hot to touch.

Finally, I reduced R301 for minimum steady-state THD+N at 1kHz, 10W. The final parallel resistance was 87Ω, and the bias voltage across R36 read 35mV or about 100mA idle current. In the event the pot wiper ever opened, the maximum value of the R27-R301 combination would not exceed 100Ω, keeping the output stage from overheating. Finally, I ruggedized the added parts with a bit of silicone rubber sealer.

For those interested in the quasi-complementary amplifier design, I have listed a number of additional references[5–10] at the end of this article. The modified circuit board is shown in *Photo 7*.

MEASUREMENTS FOR THE MODIFIED AMP

The amplifier 1kHz output impedance dropped slightly from 0.19Ω to 0.14Ω after the modification. This will improve the damping factor. Power output at 1% THD has increased a bit: from 48W to 49.5W at 1kHz, and from 36W to 43W at 20Hz. At the 41Hz E_3, it now makes 49W versus the 43W of the old design. Hum and noise dropped from 4mV to 0.35mV.

Figure 18 shows the THD+N versus output power for the modified amplifier. The 10kHz curve exceeds 1% THD at just over 2W. The 10kHz output waveform is triangular beyond this point, and the residual distortion signal is an asymmetrical second harmonic.

The increased HF distortion as compared with the original design is probably due to slew rate limiting caused by the larger compensation cap, C22. The input stage has a rather low transconductance to begin with, and Q3 has a limited current capability when compared to high-fidelity last voltage amplifier designs.

The THD+N versus frequency (*Fig. 19*) is much better, with no 60Hz component. The distortion is quite good below 2kHz, but increases rapidly at higher frequencies. The distortion is lowest at the bass frequencies, and stays below 1% out to 3.5kHz, which is near the upper limit of the driver response. The spectrum of a 50Hz sine wave at 10W into 8Ω in *Fig. 20* shows much lower odd harmonics when compared to Fig. 4, and the 60Hz component is reduced to –76dB.

Figure 21 shows the distortion residual waveform of the modified amplifier for 10W into 8Ω at 1kHz. THD+N at this test point is 0.031%. The upper waveform is the amplifier output signal, and the lower waveform is the monitor output (after the distortion test set notch filter), not to scale. This distortion residual signal now consists of mainly the second harmonic overlaid with low-level noise. This is a major improvement as compared with Fig. 5.

Measurements of frequency response (*Fig. 22*) show the response dropping to –3dB at 16Hz and 35kHz, with the three controls flat (centered). The response peaking at 60kHz is gone, with only a small ½dB peak at 20kHz. The 10kHz square-wave response bears this out, with one barely perceptible 50μs half-cycle on the leading edge.

Figure 23 shows the original TKO 80 amplifier output spectrum reproducing a combined 10kHz + 11kHz intermodulation distortion (IMD) signal at 12V pp into 8Ω. The 1kHz IMD product is 0.13%, while the 9kHz and 12kHz products are a bit lower at 0.08%.

You can also see multiples of the 60Hz AC line frequency throughout the spectrum. Compare it with *Fig. 24*, the same IMD signal with the modified amplifier. The 60Hz multiples are much lower, while the 1kHz, 9kHz, and 12kHz products are about the same level. This response is probably due to the nonlinear design of the differential input circuit in the power amplifier section.

Figure 25 shows the electrical impedance and phase of the QX108 installed in the Peavey cabinet. *Figure 26* shows the same data for the Peerless 850146. Finally, the acoustic response of the QX108 and Peerless 850146 measured with the sound level meter is shown in *Fig. 27*, compared with the stock 12″ driver and its original amplification.

GR'S SUMMARY ON DRIVER SELECTION

Evaluation of a variety of drivers available for this application resulted in three prime candidates, #10, #11, and #14. These were all 10″ drivers—#10 and #11 were woofers by Peerless, and #14 was a subwoofer by Phoenix Gold. We initially selected the Phoenix Gold QX108 because of its attractive price and large rated X_{MAX} as a subwoofer driver.

When the QX108 driver arrived, tests showed it not to be the driver advertised in terms of its T/S parameters. Design and listening showed it would not do the intended job. The cure was one of the Peerless units, which did meet its advertised parameters and performed as predicted.

This tale demonstrates one of the frustrations of home speaker building. You take a scientific approach to selecting a driver only to find the maker does not deliver what he has advertised. The best available design software won't help you if it is fed data that does not match that of the actual driver. I know of no cure for this except to stay with speaker manufacturers that you, or others, have found to demonstrate consistent quality in meeting their published specifications.

When something like this happens, you want to report your disappointment to the vendor so he will know his customers would prefer he sell quality products. This problem is the reason that I try to never do a final enclosure design based on published catalog parameters. Home builders without test capability should surely stay with manufacturers known to deliver a quality product.

CH'S CONCLUSION

The finished unit is shown in *Photo 8*. I am quite happy with the performance of this restored/modified TKO 80. Restoration was a big investment in time, and the not-insignificant amount of $270. Whether or not it was cost-effective is also up for debate—you can buy a brand new 50W Peavey Basic 112 bass amp for $360, with a 12″ driver in a ported enclosure and more preamp features.

Despite the performance problem with the QX108, I had a lot of fun, and I gained a tremendous amount of knowledge and experience about driver selection from G.R. Koonce. My TKO 80 now sounds the way I want it to sound, which is pretty much the goal of the Audio Amateur philosophy. ❖

REFERENCES

3. Self, Doug, *Audio Power Amplifier Design Handbook*, published by Newnes, 2nd Ed. 2000; p. 112, Figures 5.11 and 5.12; p. 127, Figure 5.25 (available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 888-924-9465, www.audioXpress.com, e-mail:custserv@audioXpress.com).

4. *1980 National Semiconductor Audio/Radio Handbook*, "Three Band Active Tone Control (Bass, Midrange, and Treble)," Figure 2.14.18, page 2-55, reprinted by Audio Amateur Press.

5. 1966 RCA Transistor Manual, "High-Fidelity 70-Watt Audio Power Amplifier," pp. 424–425.

6. Motorola Application Note AN-484A, 1975, "Basic Design of Medium Power Audio Amplifiers."

7. Darr, Jack; *Electric Guitar Amplifier Handbook, 4th Edition*, 1977, published by Howard W. Sams.

8. Sondermeyer and Gold, "Brute-70," *Popular Electronics*, date unknown.

9. Tymerski, Richard, "Playmaster 300 Watt Amplifier," *Electronics Australia*, May 1980, pp. 39–41; June 1980, pp. 45–61; July 1980, pp. 52–57.

10. D. Sweeney and S. Mantz, "An Informal History of Solid-State Amps," *Audio* magazine, June 1988, pp. 46–55.

An Unorthodox Two-Way, Part 3

In the third and final installment of this project, the author concentrates on crossover construction, measurement, and listening impressions.

By Jon Mark Hancock

In order to fit the electrical crossover conveniently into the cabinets, I split the networks up into three boards, using 6″ × 8″ quarter-inch hardboard as the construction base. The completed unit is shown in *Photo 18*. I've developed the habit in the last several projects of using Euro power connectors for interconnecting the crossover boards with the drivers and input connectors; this makes experimental work much easier than soldering and unsoldering connections, and they're available at your friendly local Radio Shack, as well as conventional distributors.

CROSSOVER CONSTRUCTION

Construction is simple (*Photos 19*, *20*, and *21*); I use hot glue to mount the components to the board and wire up the networks with AWG 14 wire.

For these speakers I obtained most of the inductors from distributors for Solen. If you wind your own coils—as I sometimes do—I suggest using AWG 14 wire to get a similar DC resistance. I used a combination of GE polypropylene and Solen polypropylene capacitors. Just for fun, I measured the ESR (equivalent series resistance) of the GE and Solen caps, and they're both quite low—under 10mΩ. In combination with the low dielectric absorption coefficient for polypropylene, the electrical behavior is quite good.

There are more expensive film capacitors available, which do have their supporters, but unless you're using very high-grade electronics and cables, you may not hear a difference in your system. I leave the upgrading of crossover components to the judgment of the individual constructor.

Note that the tweeter crossover is shown without the tweeter level pad installed and wired. In order to handle the power required by 8–9dB attenuation, and to have some flexibility in adjusting the attenuation, I used an unusual construction for the L-pad with multiple parallel resistors (*Photo 22*). It features paralleled 15Ω and 12.5Ω Mills resistors, five in each leg, with an additional 10Ω in parallel on the shunt leg to fine-tune the attenuation. You may choose to omit that shunt if you want a slightly brighter sound.

PHOTO 18: Finished speaker on stand.

If this is too much to deal with, there are high power 4Ω L-pads available from a few sources, such as Phoenix Gold, but while I've sometimes used

PHOTO 19: Low-pass crossover showing component locations.

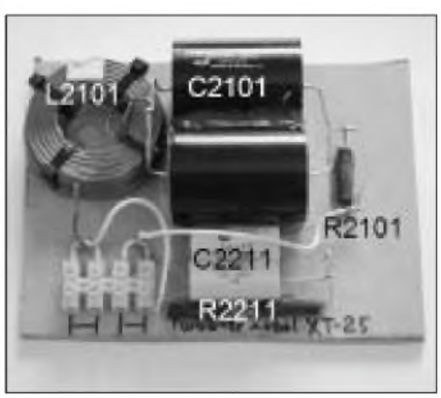


PHOTO 20: Tweeter Zobel (conjugate network impedance equalizer) board.

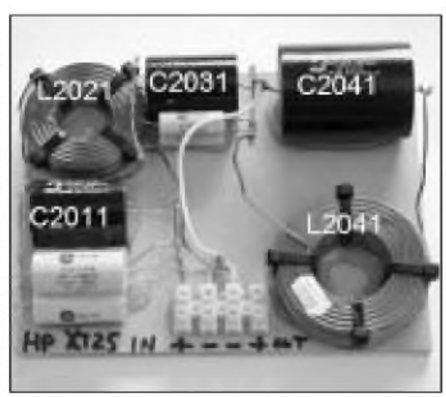


PHOTO 21: High-pass crossover network without resistor pad network.

them to experiment and determine the range of levels, I don't usually leave these in the finished speaker. Also, technically, a 4Ω L-pad is about a 25% impedance mismatch to the Vifa tweeter. In practice, this doesn't make a very significant variation in the response, and if you wanted the flexibility of adjusting levels for adverse acoustics (overly reverberant in the presence region and highs), this might be the ticket to consider.

I used a popular "DB-Cup" assembly for the input connection for which the Woodstyle enclosure is pre-cut. After testing it I made some modifications. Testing it, you may ask? Something I try to avoid in connectors and wiring is the use of any ferrous materials. The tabs for connecting the wiring to the binding posts as supplied can be picked up by a speaker magnet, so they've got to go.

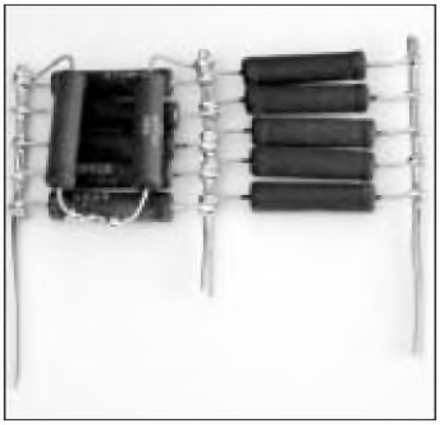


PHOTO 22: L-pad network from Mills resistors.

PHOTO 23: Input cup with new binding posts.

This prompted me to look a little closer at the binding posts themselves, which, though nicely plated, may not be the best quality base metal. In the end, I replaced the binding posts with some Vampire posts, and directly soldered the Cardas hookup wire to these posts (*Photo 23*). In testing wire for home-brew power amps, I've found in the past I prefer the subjective qualities of the Cardas cable hookup wire, and so I use it in speaker construction also. Naturally, you can substitute your favorite hookup wire—I used 13 AWG for crossover inputs and woofer wiring, and the 15 AWG for the tweeter boards and connections.

ASSEMBLY AND MEASUREMENT

At this point I knew I was in the home stretch—for the experienced builders among you, the hard work is done, and it should be easy to get the speakers working correctly, unless you forget to wire the tweeters in the correct phase, or don't adequately seal the drivers when mounting them.

Depending on your confidence level, you may do as I do and mount the drivers first with long leads coming out of the reflex port, and check out everything with the crossover boards wired up externally (*Fig. 24*). This is highly recommended as a "sanity" check before putting everything together; it's much easier to work on the crossover boards before putting them in the cabinet, if you find a wiring error.

As mentioned during the cabinet construction, the first of the crossover boards to mount and connect is the tweeter Zobel board. This contains the impedance equalizing networks formed by the LCR combination of L2101, C2101, and R2101, which controls the impedance rise at f_S, and the RC network formed by R2211 and C2211, which flattens the impedance rise due to the voice-coil inductance. Cut and connect output and input leads before mounting it. I arranged the terminal connections so that the "positive" input and output were adjacent on one side, and the nominal "negative" on the other.

Actual polarity at the board doesn't matter; this is a network that is connected in parallel with the tweeter, and the connection to the tweeter is just a "pass through." However, you must be careful about the absolute polarity for the tweeter; note from the crossover diagram that the tweeter is connected in reverse phase to the woofer for an eighth-order all-pass network, just as it would be for a second-order. Since the tweeters are mounted in what is effec-

PHOTO 24: Woofer LP crossover on bottom wall of enclosure.

PHOTO 25: Enclosure for first tests, without diffraction control treatments.

PHOTO 26: Testing felt layout for diffraction reduction.

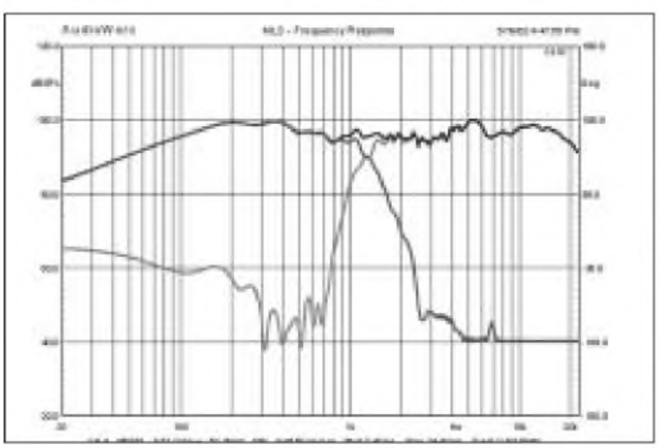


FIGURE 24: Composite LP and HP measurement in standard WS123 test box, checking crossover boards.

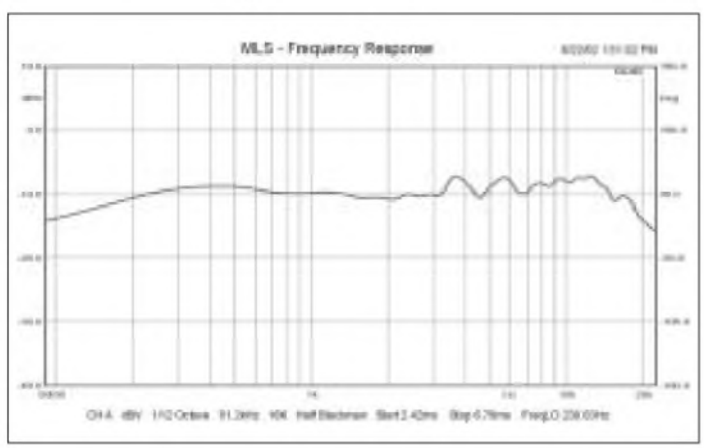


FIGURE 25: Measured response without diffraction treatment.

tively a sub enclosure, it will be necessary to drill holes sufficient to pass the tweeter wiring; I angled these to one side so that a large flat area was still available for mounting the Zobel board.

You can attach the crossover boards with "Industrial Velcro," which is available at most hardware stores as well as Radio Shack. Or, if you've tested everything and you're feeling brave, just hot-glue it in place (guilty as charged). Make the input and output connections before mounting the board, as there's very little space for working by hand in this area.

I mounted the woofer crossover boards in the bottom wall of the enclosure, towards the back and away from the front panel (*Photo 24*). I mounted the tweeter crossover board on the back panel's oak brace just below the input cup hole. Connect the leads from the Zobel network, being careful to observe correct phase.

Prior to connecting and mounting the woofer, you must install internal damping materials. I hot-glued heavy felt on the side walls behind the woofer, attaching it by using a series of closely spaced thin beads (speed is of the essence, even when working with slow-setting hot glue). In the area behind the tweeter, and also directly behind the

PHOTO 27: Completed cabinet with felt attached with hot glue.

woofer, I installed small folded blocks of polyester quilt batting. This helps with absorbing the midrange backwave, and damps the LF alignment to a degree. Keep this material away from the port opening in the cabinet, or the reflex action may be impeded and low frequency output will suffer.

I use self-adhesive weather-stripping foam for sealing the drivers. I drill pilot holes which are just slightly under the diameter of the body (excluding threads) of the mounting screws. I usually use 1″ drywall screws for driver mounting in MDF, though you may prefer black finished round-head screws, which are included in the mounting kits supplied by Parts Express with the woofers they sell.

INITIAL TEST AND DIFFRACTION CONTROL

After completing the assembly of the first MkIV cabinet (*Photo 25*), I initially checked the frequency response (*Fig. 25*). Your first inclination upon seeing this might be to reduce the tweeter level or fine tune its voicing, but noting the elevated regions in a series of ripples in the 3kHz to 14kHz range, I suspected diffraction effects at the edge were inducing this problem. As I had hoped for after using the Baffle Diffraction Simulator to plan the driver layout, they are distributed over a wide frequency range and at relatively low amplitude.

Photo 26 shows my experimentation using soft felt to reduce edge effects. I cut up scrap pieces of various sizes and experimented with the felt cutout in the area of the tweeter. A circular cutout would be the worst of course, since it would result in a diffraction effect at a uniform distance and peaking at one frequency.

Though I also tried a square cutout as Wilson Audio uses and I have previously employed with the Focal Tc120dx2, I found that the shallow diamond shape shown here seemed to work best in minimizing the diffraction ripples (*Fig. 26*). This is reflected in the finalized configuration implemented for the felt on the front baffle (*Photo 27*). You could use double-sided foam tape to attach the felt, but very satisfactory (if hard to reverse) results will be obtained with that popular standby, hot glue.

Photo 27 shows the final configuration; note that the raised surface provided by the additional ¼″ sub-panel and the felt makes a fairly good blend to the standard Woodstyle grille frame, which has a slight bevel undercut on the inside/top, reducing the height of the grille frame at the inner edge.

ADDITIONAL MEASUREMENTS

I made some additional measurements to check and verify performance against the design goals. *Figure 27* shows the measured impedance curve. As expected from the tweeter Zobels used, the upper range is quite flat, though one friend, on seeing the low frequency variation and minimum impedance, quipped that "this isn't very SET friendly."

The LF impedance curve shows some modification from initial test box measurements, cutting the impedance peak from about 55Ω to about half that value, and reducing the Q of the curves. I believe this is due to the batting used behind the woofer and behind the tweeter area for additional damping. You can see the effects of

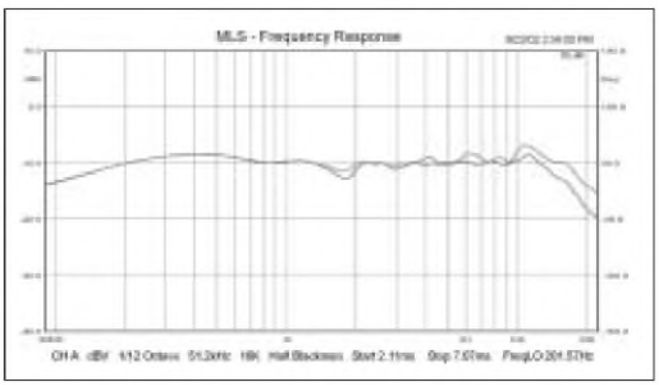


FIGURE 26: Measured response 0 and 15° off-axis after diffraction treatment.

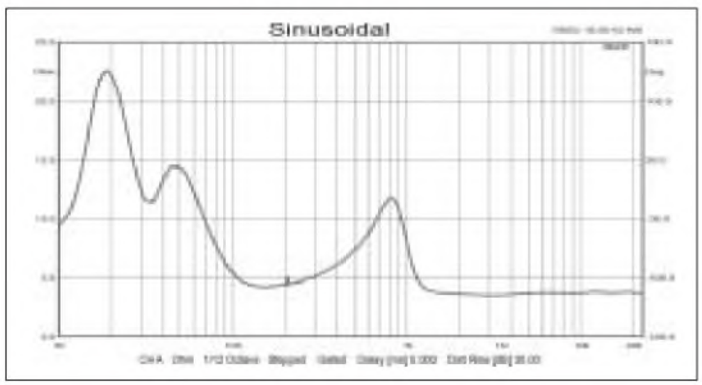


FIGURE 27: Impedance plot of completed system.

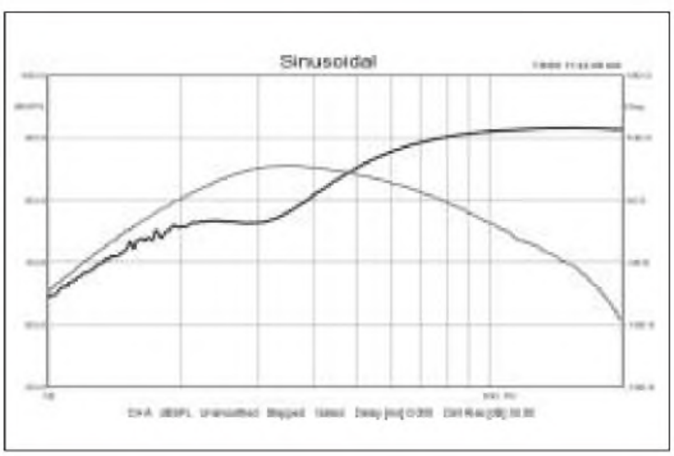


FIGURE 28: Near-field woofer and port response.

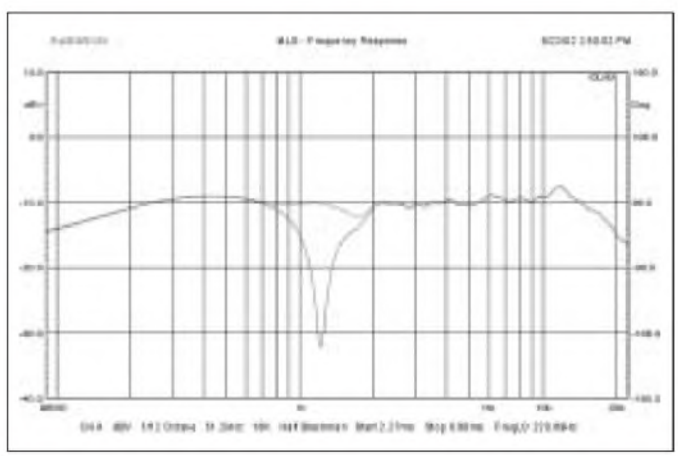


FIGURE 29: Normal and out-of-phase response (null depth test).

this damping in the plot of near-field woofer and port output (*Fig. 28*). This somewhat "over-damped" alignment has good bass extension and weight, but without the heaviness or lack of articulation often attributed to ported designs.

I also made plots of the response with the tweeter in "normal" phase (actually reverse polarity) and for the "out-of-phase" condition as the classic check on crossover alignment. *Figure 29* shows a deep and narrow null, which you would expect from this high slope design. For the MkIV version I centered the design axis between the woofer and tweeter, believing it was more useful to have good off-axis behavior above the primary listening axis, such as when standing up, compared with the performance well below the listening axis, which might only be encountered if lying on the floor. *Figure 30* shows the resulting vertical response window, with the dip in response in the crossover region occurring at 15° below the woofer axis. Performance in the crossover region is smooth both for seating position and standing at reasonable distances.

Figure 31 shows a series of sweeps at 15° increments on the horizontal axis. The dispersion limitations of a 1″ nominal tweeter above 10kHz become readily apparent from this plot. However, this also shows the wisdom in general of not exceeding roughly 1200Hz for the crossover of an 8″ driver, as the behavior in the upper midrange and presence region (1–3kHz) remains remarkably consistent from on-axis to 60° off-axis.

Note that a different test microphone was used for this measurement, a Behringer ECM8000. I believe that this consistency on- and off-axis goes a long way towards explaining the large sweet spot these speakers exhibit, as well as their overall sense of transparency compared with many two-way systems using smaller mid-woofers but higher crossover points.

SETUP AND EQUIPMENT

All of the versions including the MkIV described here underwent listening checks during the crossover evaluation process. By the time I was finishing the MkIV, my ability to correlate what I heard with specific measurements had improved considerably. Much of the listening evaluation for the MkIV crossover occurred in the test box phase, with only a very minor adjustment (lowering) to the shunt impedance of the L-pad in the final crossover.

My program sources are all digital, including a Sony SCD777ES SACD player, an APN Audio MP-DAC II, and an experimental DAC using a CS8420 for 2× sample rate conversion (44.1 to 88.2kHz) and re-clocking, with a CS4397 24/192 converter and transformer-coupled connection to a non-loop feedback discrete Class A balanced output stage.

The preamplifier used for evaluations was a Marchand PR41 passive unit, using Shallco switch-based attenuators, which has little sound of its own, as long as short interconnects are used. Interconnects from the digital sources to preamplifier were Jon Risch recipe cables (the second version with mixed core materials) constructed with WBT connectors. Cardas interconnects were used from the preamplifier to power amplifier. Speaker cables are based on Kimber 8TC with WBT connectors.

I tried both a conventional power amplifier with high damping factor (an Aragon 8008X3B) and a non-loop-feedback design, the Ayre V-5 in my listening tests. In no sense did the bottom end control seem to suffer with the Ayre amplifier, though its measurable damping factor is nowhere near as high as the Aragon. Because of its pristine midrange and high-frequency behavior (very "un-solid state" in character), I used the Ayre V-5 for the majority of listening evaluations. The V-5's frequency response extends to 200kHz, so I don't think the smoothness in the upper range can be attributed to rolled-off frequency response, as some claim for vacuum tube amplifiers.

As discussed earlier in the design phase, setup with regard to boundaries plays a crucial role for the in-room performance of any speaker. Fortunately, if you're not fond of throwing together your own MathCAD doc or Excel worksheet to analyze your setup, there are some excellent programs available at reasonable cost to enable you to take as cursory or detailed a look at this issue as you like.

I've found Room Optimizer from RPG Acoustics very helpful. As its name implies, Room Optimizer can actually help you optimize the setup of the room, and not just by telling you how mediocre your first thoughts for speaker placement were! Using constraints you choose, it will perform a search for optimization of both the speaker and listener position in the room.

It does this by analyzing both the boundary and modal response issues in the room, and within the constraints of the configuration setup, explores a variety of combinations of locations to identify the overall smoothest response in the optimization range for the low frequencies between 20Hz and 300Hz. For the "optimized" layout, it will also suggest locations for room treatment and damping to minimize response irregularities stemming from multiple path comb filtering.

What's worked best for me with Room Optimizer is defining a starting point based on my MathCAD calculations for locations next to the major boundaries. Then, enter a range of location for the speakers, and make one mirror dependent on the other for a symmetrical stereo layout. Define an area for the preferred listener locations, then let the optimization process have a go at it and see what turns up.

Figure 32 shows the results of a positioning analysis I ran for this system in one room, including suggested positioning for wall treatments to reduce early reflections. *Figure 33* shows the comparison in the modeled frequency response during optimization, including the best- and worst-case results.

Keep in mind that the optimizer is an idiot savant—it's just a "dumb as dirt" number cruncher. If you give it some good clues and starting points to work from, it can weed through a lot of possibilities and allow you to explore some "what if?" scenarios pretty quickly (literally, just a couple of minutes on a PIII or Athlon system). Such as, "What if I rotated the listening axis in my room 90°—What benefits or problems would result?" A comprehensive look at ways to use Room Optimizer could be a complete article on its own, and is well beyond the scope of this article. The white paper available on RPG Acoustics' website[8] gives a thorough overview of the capabilities of the program and the basic techniques used at its core.

In my own setup, I have a few additional "problems," such as a front projection screen behind and between the speakers—when not viewing video, I

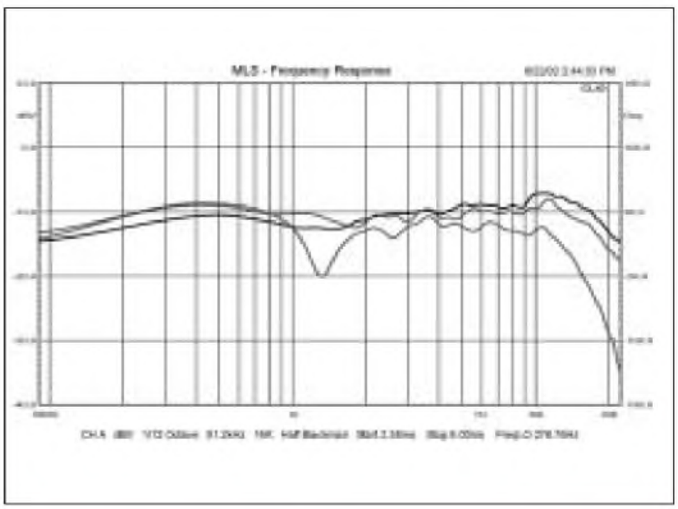


FIGURE 30: Vertical off-axis response (+15, 0, −15°) optimized for seating and standing listening (don't sit on floor).

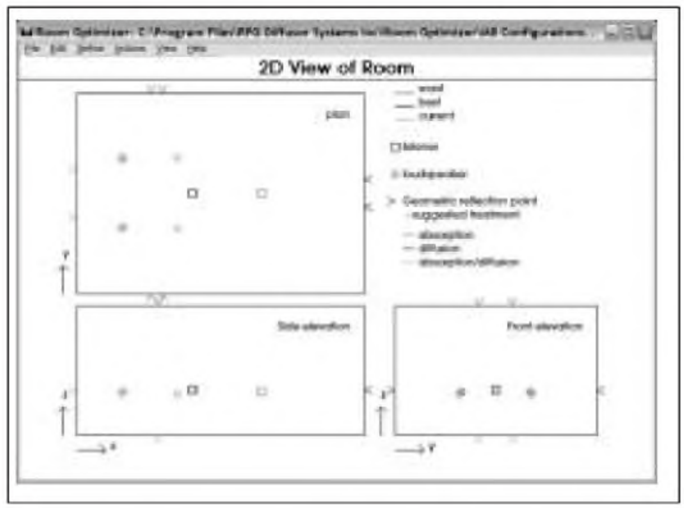


FIGURE 32: Analysis of positioning of speakers, listener, and treatments using RPG Room Optimizer.

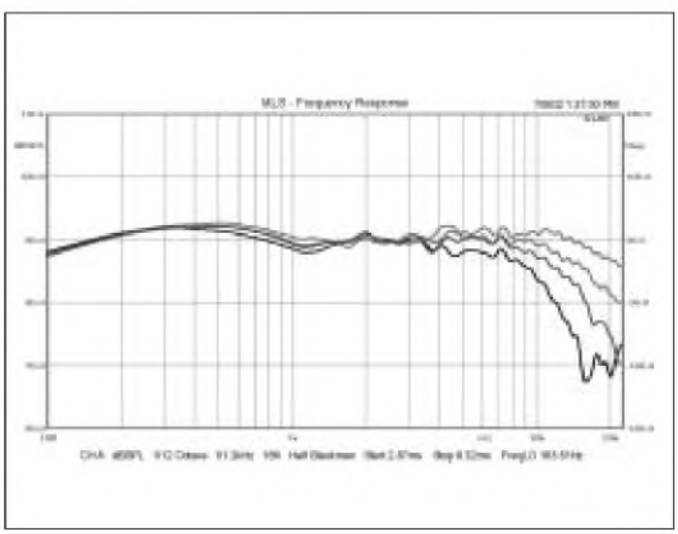


FIGURE 31: Overlaid horizontal on-axis and off-axis sweeps (see text).

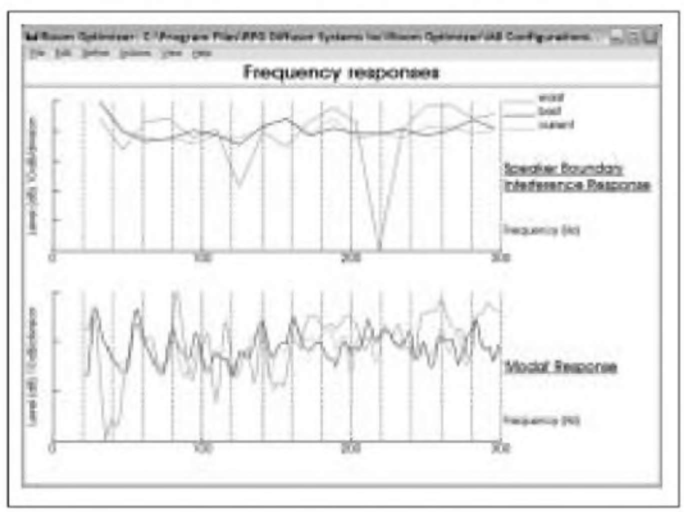


FIGURE 33: Comparison of "worst," "best," and "current" positions after optimization pass in Room Optimizer.

drape the fixed screen with a white down comforter.

LISTENING IMPRESSIONS

Vocal music, both popular and classical, constitutes a large part of my listening preferences. Because of our innate familiarity with the human voice, it tends to be fairly revealing of coloration in a wide part of the musical spectrum, from upper midbass through the presence region and top end. Solo voice works well, but combined voices probably work even better, as they're more revealing of possible intermodulation issues—and quickly separate the wheat from the chaff in the ability to resolve individual vocals with natural timbre.

Modern high-quality recordings of pop such as Keb Mo's *Slow Down* and Alison Krauss's *Forget About It* sound very good, with the expected clarity, body, and dimensionality. I've found listeners with little previous exposure or interest in Bluegrass entranced by the tracks "It Don't Matter Now" and the title cut "Forget About It." The acappella vocals of Jonatha Brooke and Jennifer Kimball of the Story on several cuts from *Grace in Gravity* float in space, while at the same time make a strong personal connection for me.

I've been very pleased in the last few years about how some older recordings fare that have been carefully remastered, such as Maddy Prior's *Woman in the Wings*, which is a collection of original songs from the vocalist who is probably best known for her long-time career with Steeleye Span, a British band doing rock arrangements of traditional British folk songs and original compositions. In this recording, her original songs showcase her voice in a variety of styles including traditional jazz influences; the recording is so natural and vivid sounding that it's a startling contrast for those familiar with the processing and reedy tonality on her work with Steeleye Span and in other projects. In this regard it sounds like a much more recent recording than its original 1979 release date would suggest. Perhaps the production values Ian Anderson brought to the studio as producer made the difference, complete with very competent backing performances by a host of musicians from late '70s incarnations of Jethro Tull.

The Hunter by Jennifer Warnes is a modern well-produced album in her ultra-clean style that gives a good demonstration of natural, relaxed midrange with detailed top end reproduction, while including a good workout in the bottom end from the low 30s and up. Cuts such as "Big Noise New York" demonstrate Warnes' command and assurance as both a writer and performer.

Recent releases from Telarc have raised the bar for recording quality and have also featured some surprisingly good performances compared with many "audiophile" recordings. I especially appreciate their new DSD recordings released on hybrid CD/SACD disks, because they give the listener a choice in playback equipment. Though I'm no music critic, I think the new release of Carl Orff's *Carmina Burana* could well come to be the performance and recording by

which others are judged, and likely found wanting. This opinion comes from auditioning it on a "reference" system, which has been assembled at a friend's over the years.

Obviously, it would be pretty ludicrous to compare this recording on an 8″ two-way to a four-way system with active crossovers using an IB sub with a dozen long-throw 12s, two pairs of Acoustat 1+1 electrostatic panels from 100Hz to 600Hz, a pair of Bohlender-Graebener RD75 ribbons from 600Hz to 7kHz, and a Technics leaf ribbon line array covering the top end. Yet, on this "modest" two-way there's a surprising clarity and transparency in vocal reproduction that clearly reveals the recordings' merits, and even more surprising is the impact achieved with the remarkable percussion on this recording. This is assuming relatively reasonable playback levels, considering the limitations of a speaker of this size. Otherwise, increasing levels of even-order distortion from the Vifa tweeter as well as dynamic compression of the woofer on high-level bass transients will result—but at playback levels over 100dB.

All in all, the performance of these moderate-size speakers has reached, and perhaps exceeded, the level of transparency and musicality I was hoping for. With four iterations of the design completed, it's clearly a triumph of perspiration over inspiration; there's no substitute for development and refinement. In the form presented here, they're not for every room and situation, because they are designed and benefit from spacing well out from the walls.

My friend with the "reference system" was pleased enough with the results that we've designed and built a version specific to boundary loading on a wall for use in his home, using a smaller cabinet and removing baffle step compensation in the crossover filters. Further development underway includes a floor-standing MTM version, being constructed as I complete this article, and a dipole system using these drivers in the midrange and high-frequency area.

I'd like to take this opportunity to thank Charles Hansen (formerly of Avalon, currently of Ayre Acoustics, not

PARTS AND VENDORS

Woodstyle W123REV enclosures	Madisound
GE polypropylene caps	Madisound
HiVi Research M8a woofer	Parts Express
Vifa XT25 XXX tweeter	Madisound, Parts Express
Solen capacitors	" "
Solen perfect lay inductors	" "
DB binding post cups	" "
Vampire or Cardas binding posts	Welborne Labs
Cardas hookup wire	Welborne Labs
Mills resistors	Part Express, Welborne Labs
1 × 3 oak board—4′	Home improvement stores
1 × 4 oak board—10′	Home improvement stores
1 × 6 oak board—17′	Home improvement stores
¼″ smooth-sided hard board	" "
Two-part slow-setting epoxy	" "
Slow-setting hot glue and gun	" "
Polyester batting, 1 pkg, as for quilting	Fabric stores
⅛″–½″ hard felt	Gasket supply firms
⅛″ soft felt	Local hardware stores

the Charles Hansen frequently published in *audioXpress*) for the time he gave discussing the design of the Avalon Eclipse, as well as some suggestions with regard to measurement techniques. I'd also like to thank Thomas Waale, who suggested some of the bracing techniques used in the MkIII and MkIV versions, as well as providing the use of his main listening room for evaluating the MkIII version in a larger environment than my own home would permit. I'd also like to "complain" to Edward Dell about the huge stack of *Speaker Builder* magazines I have at home and can't bear to give away or dispose of.* For these guys, it's all about the love of music and its reproduction, and sharing that with their colleagues, friends, and readers. ❖

**Speaker Builder's* issues are all being scanned for CD availability.

SOURCES

Madisound
PO Box 44283
Madison, WI 53744-4283
608-831-3433
FAX 608-831-3771
info@madisound.com
www.madisound.com

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

Welborne Labs
PO Box 260198
Littleton, CO 80163-0198
303-470-6585
FAX 303-791-5783

RECORDING REFERENCES

1. Carl Orff, *Carmina Burana*, Atlanta Symphony Orchestra, Donald Runnicles conducting, Part 1, Telarc Hybrid SACD-60575.
2. Debussy, *Iberia* from Images, Cincinnati Symphony Orchestra, conducted by Jesus Lopez-Cobos, Telarc Hybrid SACD-60574.
3. Gabriel Faure, *Piano Trio in D Minor*, the Florestan Trio, Hyperion Records SACDA67114.
4. Harry James, *Best of Harry James*, Track XX, Sheffield Labs.
5. Patricia Barber, *Companion*, Track 7, First Impressions Music, FIM XRCD 027.
6. Arvo Part, *Frateres*, The Orchestra of Flanders, conducted by Rudolph Werthen, Telarc CD-80387.
7. Oscar Peterson, Ray Brown, Milt Jackson, *The Very Tall Band*, Telarc CD-83443-SA.
8. The Ravi Shankar Project, *Tana Mana*, Tracks 4 and 8, RCA 2016-2-P.
9. George Fredrick Handel, *Messiah*, Swedish Radio Symphony Orchestra, conducted by Anders Ohrwall, FIM Music FM SACD 039.
10. Jennifer Warnes, *The Hunter*, Tracks 2, 3, and 9, HDCD XRCD2 (Japanese Import) CDA1065.
11. The Story (Jonatha Brooke and Jennifer Kimball), *Grace in Gravity*, Tracks 1, 10, and 11, Elektra 961321-2.
12. Allison Krause, *Forget About It*, Tracks 2 and 8, Hybrid CD/SACD, Rounder SACD 11661-0465-6.
13. Keb Mo, *Slow Down*, Tracks 3, 6, 9, BK069376.
14. Jonatha Brooke, *Steady Pull*, Tracks 1, 2; CD BDR60801-2, DVD-A BDR-DV-61001.
15. Maddy Prior, *Woman in the Wings*, Tracks 2, 8, 11, BGOCD215
16. Oregon, *Beyond Words*, Tracks 2, 4, 6; Chesky Records JD130.

REFERENCE

8. RPG Acoustics, http://www.rpginc.com, White papers for Room Optimizer, Room Sizer.

MEASUREMENT NOTES

Measurements were made with the CLIO system from Audiomatica, using both DOS 4.5 software and CLIOWIN. An HP microphone preamplifier and B&K 4133 capsule were used except Figs. 52 and 55, made with a Behringer ECM8000 microphone and MAudio DMP3 microphone preamplifier. In room MLS measurements used relatively short windows to reduce wall and floor boundary interactions, limiting the measurement accuracy below 400Hz.

Xpress Mail

CORRECTION

In a private communication, reader D.S. Jenkins kindly informed me of some errors in my article on "The ABCs of Filters" (*aX* 9/03).

The first error is in the equation given in line six of the center column of page 45: $V = I(R + j\omega L - j/\omega X_c)$. The second term in parentheses should be $j/\omega C$.

Second, an exponent sneaked out somewhere between my computer and the printed copy in the units for the newton, given near the bottom of the middle column on page 45. A newton is a kilogram × meter/second^2, not a kilogram × meter/second.

In addition, I made a more serious error in the last two equations on page 45: I changed horses in midstream! I have stated most of the equations in mechanical variables, but have used volume velocity, which is an acoustical quantity. The *U* in the force equation should be lower-case, indicating mechanical velocity in m/s, and this should be reflected in the multiplier corresponding to *u* in the last equation, which should be m/s, rather than m^3/s.

I would like to publicly thank reader Jenkins for his careful reading of my article, and to apologize to other readers for any confusion this may have caused. I should have proofread more carefully.

Richard Honeycutt
Lexington, N.C.

SIX-CHANNEL VOLUME CONTROL

I would like to thank Dennis Colin for his six-channel volume/balance control design published in Feb. '03 (p. 6). However, I seek his comments on the following, which in no way is intended to be a criticism.

I have built a circuit based on his design using the SSM2018T, but with modifications. I used a supply voltage of ±15V, which seemed to be a preferred design parameter of Analog Devices (from the data sheet) and derived the control voltage from a 5V regulator IC with a 2k fixed and variable 5k voltage divider and the balance voltage from a 20k pot across the ±15V supply. This seems to work very well and gives both gain and attenuation, which I have found to be necessary with DVDs, especially for the center channel. This significantly simplifies the original design, and the only disadvantage I have found is a transient increase in gain at switch-off.

Also, Dennis biases the SSM2018T into Class-A but does not re-trim the device for this class of bias. The manufacturer indicates that the device is designed and trimmed for Class-AB and that distortion will actually be higher in Class-A unless external trimming is used to compensate for the change in bias. Comments, please.

David Allen
Queensland, Australia

Dennis Colin responds.

I'm gratified to hear that you've built this circuit, and happy that it works well and satisfies your application.

To answer your questions/comments:

1. *My choice of ±12V supply was simply to reduce the IC dissipation somewhat. In a test, I found no noticeable degradation over the AD data sheet's range of ±5V to ±18V.*
2. *Your use of a separate (5V) regulator for the control voltage is advantageous in isolating potential interference from the ±15V audio supply.*
3. *Using ±15V across the balance pots, while not providing temperature compensation, is still very stable: with a balance range of ±12.5dB, at maximum setting the gain change is only about ±0.16dB from +60° to +80°F. And after using the unit with many DVDs, I agree with your observation about this extra range of balance (+ and –) needed, especially on the center channel.*
4. *I can only guess regarding the transient gain increase at switch-off. If the negative 15V supply doesn't discharge as quickly as the positive one, the balance pots affecting all channels will be momentarily biased negative, raising the gain. You could add more capacitance to the +15V regulator output to equalize the voltage decay times, if this is the cause.*

5. Class-A biasing—I somewhat arbitrarily chose this, reasoning that even a minute amount of AB (crossover type) distortion is much more objectionable than the (lower-order) Class-A distortion. Note that the data sheet offers no specific bias current to pin 12 for Class-A. I chose 22k from ±12V to pin 12, resulting in about 0.51mA bias current. I roughly calculated this to be just more than the maximum current "steered" into the transconductance array halves by the audio signal.

I don't have a distortion analyzer; that's why I didn't attempt re-trimming for Class-A. On page 10 of the data sheet (Rev. B), THD is quoted as 0.05% in untrimmed Class-A. My previous experience with Class-A small-signal transconductance stages has been that low-level asymmetry distortion is almost purely second harmonic; 0.05% second harmonic is most likely completely inaudible (when this is at peak signal and it decreases rapidly at lower levels, as it does here in Class-A, not necessarily so in AB). But you could certainly implement AD's symmetry trim and/or use AB biasing.

One other point. AD states that AB biasing results in lower noise. They're correct, but not by much. I looked at output noise with the help of an AD797 op amp at X100 (+40dB), feeding a scope and a Fluke 8010A true-RMS DVM on 200mV AC range. This provides 1µV resolution with 220nV noise floor regarding the control unit's audio output noise. This increased only about 0.5dB from Class-AB to Class-A biasing.

I very much appreciate your letter, comments, and circuit simplification. Also, it's gratifying to know that there are now at least two of these circuits properly working on the planet, on opposite sides thereof!

DIGITAL RIAA PLAYBACK II

I thank the editors of *audioXpress* for publishing my letter and Gary Galo's response about digital RIAA playback equalization in the May 2003 issue (p. 61). My error curves were not presented to display the best possible digital RIAA filter design, but to show that no weirdness appears in the digital filter; simply because it is digital would prevent it from being a proper RIAA playback filter in terms of both amplitude and phase.

Certainly you can add a small corrective filter to make the error between 10kHz and 20kHz as small as you desire. *Figure 2* shows the result of adding just such a correction to the basic digital RIAA filter whose error is shown in *Fig. 1*. Both figures are the difference in dB between the digital filter and the ideal RIAA equalization.

Up to nearly 20kHz, the error is less than 0.05dB. The error increases a bit above 20kHz, but any anti-aliasing filter designed for 44.1kHz sampling will eliminate this.

So, even if you sample at 44.1kHz instead of 96kHz, you can make the RIAA playback equalization as accurate as you desire in the digital domain.

Victor Staggs
Torrance, Calif.

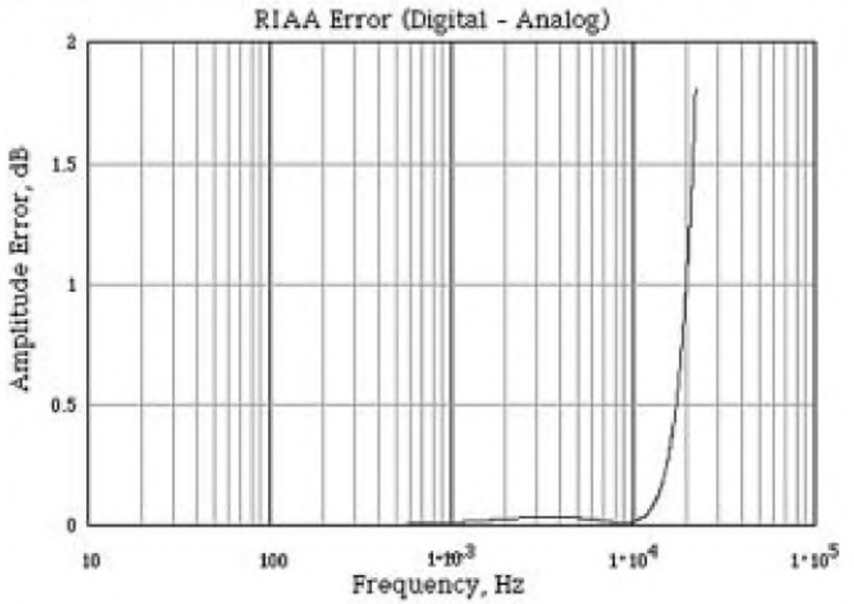


FIGURE 1: RIAA amplitude error using simplest digital RIAA playback filter. XM-1003-1

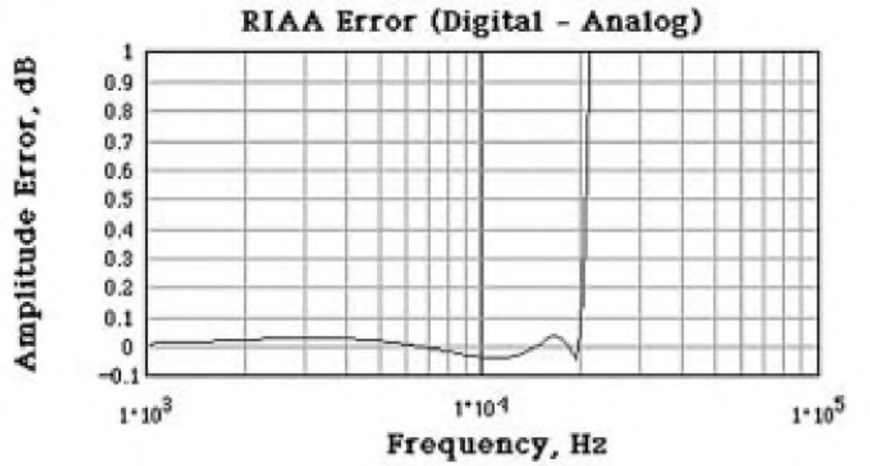


FIGURE 2: RIAA amplitude error using enhanced digital RIAA playback filter. XM-1003-2

DIGITAL RIAA PLAYBACK III

Here is my last installment concerning designing a nearly perfect RIAA playback equalizer to be applied in the digital domain as a software DSP filter. This one should meet Gary Galo's most stringent requirements even up to 22.05kHz for a sample rate of 44.1kHz.

Figure 3 shows that an RIAA playback filter response error of less than or equal to 0.05dB can be achieved to above 20kHz. Any response error above 20kHz will be reduced to zero along with the frequency content of the sound file itself due to the usual anti-aliasing filter applied during digitization of the analog sound. As in my original letter (see "Xpress Mail," May '03 *aX*, p. 63), the excess phase error in the case of this playback filter is about 15° at 22kHz, which will be inaudible.

If you visit http://www.sonyplugins.com/ on the Web, and select Products>EQ, you will note that Sony makes a software DSP equalizer that runs in the digidesign DSP card. Sony notes that they take steps to preserve the correct frequency response of their Oxford EQ right up to the Nyquist frequency, which they call "Fully decramped HF response." My *Fig. 3* shows that you can also do this for the RIAA playback equalizer.

Victor Staggs
Torrance, Calif.

TAKING IT TO THE STREETS

Please express my appreciation to Charles Hansen for his budget milliohmmeter bridge in the May '03 issue (p. 18). I constructed the unit essentially as presented and it works very satisfactorily. You may be amused to learn that my primary use of it so far has been in my alternate hobby of streetcar restoration, because resistances of less than 1Ω are commonly used in the propulsion circuits of early vehicles.

My only modification of the unit has been to house bridge and battery together in a metal case (while hoping that

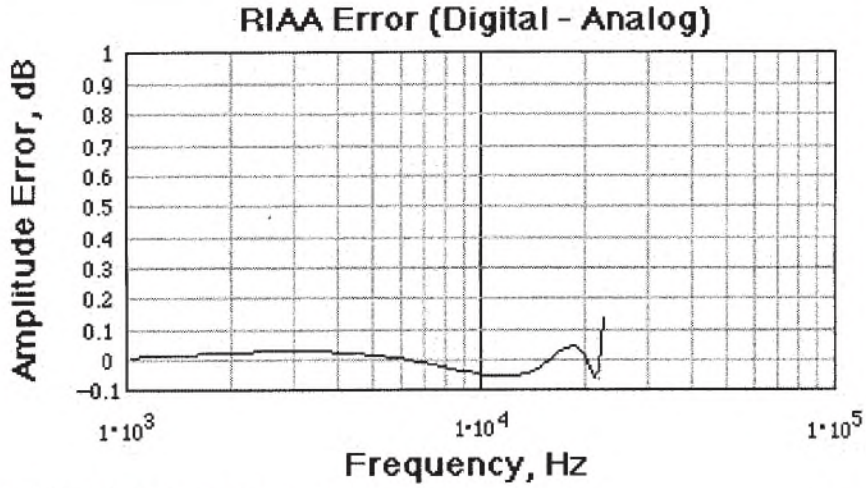


FIGURE 3: RIAA playback response.

XM-1103-3

corrosion will not prove to be too much of a problem) and to include a 12V transformer with a bridge rectifier in series with a 10Ω current limiting resistor. I replaced S1 (the on/off switch) with an SPST switch, allowing either operation of the unit or charging of the battery.

David Noyes
dnoyes1@comcast.net

Charles Hansen responds.

Thanks for the feedback. I always wonder which of my test equipment articles are of interest to the readers. David made some nice practical modifications to the milliohmmeter bridge to suit his unusual application.

TUBE PROJECT

First, I would like to thank you for your magazine. I have been subscribing since *Glass Audio*. I am very pleased with the transition to *audioXpress*.

I was reading Pete Millet's article, "An Affordable Full-Range Speaker Project" (*audioXpress* June '03). His project seems affordable and very worthwhile. Mr. Millet mentioned his 20W single-ended amp using 813 tubes in triode mode. I would like to get any information he can share about his amp.

I, too, have some 813 tubes I would like to use in an SE amp, but I can't find any characteristic curves for triode connection with these tubes. If he has these curves, I would like to get a copy. If not, I would like to know what operating parameters are used: plate voltage, bias voltage, self bias or fixed, driving grid swing voltage, effective plate load, and output transformer primary impedance.

Again, thanks to *audioXpress* and Pete Millet for his article.

Jim Dungan
jwd12ax7@aol.com

Peter Millet responds.

My 813 triode-connected amps have generated quite a bit of interest. I've recently post-

ed some info about them, including a link to 813 triode-mode curves, on my website at pmillet.addr.com—follow the link under "projects" to the 813 SE triode amp.

The triode curves, as well as considerable inspiration, came from a gentleman in England named Paul Leclercq, who built an 813 triode-mode push-pull amp. Thanks, Paul!

I'm operating the 813 with a plate voltage of about 950V and plate current of 100mA, with fixed bias of about –80V. The 813 is connected as a triode (screen grid to plate, suppressor grid to ground) and is loaded by a 5000Ω OPT (a bit higher load would work better, but 5k was what I had).

I'm using a single high Gm pentode, a 12HG7, to drive the 813. The driver needs to deliver 160V P-P to clip the 813; the 12HG7 circuit can provide around 280V P-P, with a gain of 40. I'm using a supply voltage of 400V, plate load resistor of 4000Ω, cathode resistor of 50Ω (unbypassed), and regulated screen voltage of 150V. Details about the driver are available on my website as well.

The 813 is a great tube, high powered and not too expensive (yet). Good luck with your project!

ENHANCED TV

Well done, Darcy Staggs! ("Guest Editorial," p. 4, and "Xpress Mail," p. 66, June '03 *aX*). I had thought the poor-quality TV picture problem peculiar to my local conditions in Derbyshire, England.

Due to my low-lying location in a hilly area, I use a tall aerial mast with a mast-head RF amplifier, whose phantom power supply is permanently powered. This lifts 12V through the aerial downlead with what transpired to be one of the noisiest supplies I have ever encountered. Cleaning it up with a cheap RF choke (the type usually found in switch-mode supplies) and better caps not only improved the picture quality, but also the sound of my audio system even when the TV is off. My boys' Sony PS2 is also better left unplugged when not in use.

My audio system is supplied via its own dedicated spur from the fuse box, including a Russ Andrews Silencer, so the problem would be worse with audio and video supplied through the same mains ring.

It is sad that 30 years ago some of the finest rectifier tubes were those designed for TV deflection (try them in your amplifier; they're quieter and more linear) and yet today's televisions are encumbered with unnecessary features but fail to make image quality their number one priority. I shall try Mr. Staggs' suggestions as soon as my TV is out of warranty.

Through many experiments and hundreds of hours listening, my friend Graham Nalty (of Black Rhodium cables, formerly Sonic Link & Audiokits) developed a capacitor bypassing strategy that I have yet to improve upon (and I have tried). He found that a rule of thumb is to bypass with 1/100 value ratios until you reach a small silver-mica value. Use the first bypass to reach a measured value for the whole clump that is closer toleranced than the main big value capacitor. Ultra-low esr is even more valuable with the frequencies handled by video circuits than audio.

Mark Wheeler
Derbyshire, England

Darcy Staggs responds:

Thanks for sharing your TV experience with audioXpress readers. After learning what it takes you to get RF into your television set, I don't feel so badly about my 35-mile range.

Your experiences with improved capacitor quality are about to bring you results in your video which you never will have expected. Please let us know at the end of it all what you see. I know PAL, at 625 lines, can look beautiful, from the years I lived in Stockholm, Sweden.

My 1996 Zenith picture tube has recently sustained a shorted green gun, making the picture mainly green and yellow. As I write, however, I expect the arrival of a rebuilt picture tube (!) from Des Moines, Iowa. These are fascinating times.

CAPACITOR VIOLENCE

Darcy Staggs (June '03 *aX*, pp. 4 and 66) provides an innovative (if not adventurous) method for improving the performance of video equipment. My only caveat is that you should never replace aluminum power-supply reservoir filter caps with solid tantalum types. Solid tantalum pentoxide does not have the electrolyte mobility of the chemistry used in aluminum caps, and cannot withstand high inrush currents. A hot-spot will develop where the current density is highest, leading to dielectric failure.

Although some military types (high-frequency CSR21) can handle up to 1/2A, the conventional solid tantalums are limited to 1/3A max. When used as primary filter caps, solid tantalums can explode if asked to handle high currents. I have seen this happen a few times when new engineers, despite prior warning, tried to take advantage of the smaller size of tantalums in a power supply or snubber circuit, only to be greeted by a catastrophic capacitor failure.

Chuck Hansen
Ocean, N.J.

Darcy Staggs responds:

Charles Hansen raises an important issue in his description of the limited suitability of tantalum capacitors as primary power-supply filters, and experimenters should incorporate his caveats in their plans.

The power-supply rails I have bypassed with tantalum are in the low-current, signal-processing portion of the video circuits. Since the integrated circuits involved are usually CMOS and therefore use little current, tantalum has been succeeding. These capacitors have been in service for over two years now.

In a way, there is a natural selection process that has kept tantalums out of the riskier applications in my TV and DVD player. Tantalum caps have shown their best performance scattered around the boards bypassing the many supply pins, sometimes several per IC. By this point the currents are spread out, and have gone through one or two stages of regulation.

My TV set is the only object where appreciable currents occur, usually associated with higher voltages, which I have left alone, with the exception of larger aluminum electrolytics to smooth the unregulated 215V DC gun voltage.

The Zenith documentation clearly marks all safety-related components, which remain undisturbed. Some of those components are current-limiting resistors.

One more related TV tidbit. As I mentioned in the previous letter, I needed to replace the picture tube in my Zenith, and thanks to the Internet, found Hawkeye Picture Tube in Des Moines, Iowa, which markets rebuilt picture tubes. This has been 100% successful.

MORE TV TALES

I was much intrigued by the Guest Editorial (June '03 *aX*, p. 4), which reminded me of an earlier visit to NYC when I was sounding off to Charlie Repka about the poor quality of US TV (I exclude content, which is another matter!). He assured me when I saw a professional monitor I would change my mind. Sure enough, he arranged a visit to CBS and proudly showed me a large monitor showing a test card. Thereafter, there was a heated argument . . . what he saw as a superb red, I saw as dark orange. To my eyes, flesh tones on programs never looked right!

So I don't think you can blame it all on the current receivers because my experience is quite the reverse with reputable makes. Here we are in the middle of a revolution, moving with indecent speed to digital transmission of everything—radio as well as TV. The latter because the BBC broke the iron grip of Rupert Murdoch with the move to digital terrestrial TV, which we are currently enjoying. In four months, it has grown to 55 stations, including all the

DIY SHOWCASE

PHOTO 1: Just wanted to say how much I enjoy reading your magazine and show you pictures of my completed THOR speakers. Keep up the good work.

David Schneiderman
splus11@attbi.com

public service radio stations—and all free to air.

Reg Williamson
Kidsgrove Staffs, UK

8417 RECOLLECTIONS

I noticed some letters in the July issue about the 8417 tube, so I thought I'd throw in my two cents.

In the early '80s I did a lot of testing, looking for the ultimate guitar amp output tube. My favorite was the 8417—warm and alive sounding. This was in a single-ended triode configuration. Initially this amp had two output tubes in parallel, but after listening tests, I decided I liked a single tube better. This was in a guitar amp where a little distortion is OK or sometimes even desirable. I don't think much of paralleling tubes.

So I bought ten GE 8417s from a distributor in Oakland, Calif. He got them directly from the factory in Kentucky.

They had the sound for sure, but they failed regularly, even though I wasn't running them hard at all. I had to give up on them. If I had known Westinghouse made them I would have given them a try, but I've never even seen one.

8417 was the last power receiving tube developed in this country. By the time it reached the marketplace, there wasn't much demand for it—transistors had nearly taken over.

One point I don't see mentioned much in your magazine is that it takes considerable skill and specialized equipment to make good vacuum tubes. Audio tubes, in particular, must be made with low microphonics and quality internal construction. If any of the tube elements move, it will degrade sonic performance.

To my ears, there is only a small percentage of tubes that have the magic presumably everyone is looking for. It's not a type either—it's how carefully an individual manufacturer made their particular iteration.

The 8417 was a terrific design, but there wasn't enough of a market for manufacturers to really buckle down and make them properly. It was a relatively hard tube to make, I gather.

Rick Bergman
Missoula, Mont.

MORE ON THOR

I would like to congratulate Mr. Rinaldo for his work on "cleaning up" the cabinet aesthetics for the THOR loudspeaker ("Showcase," Aug. '03, p. 58). In particular, I find his treatment of the base particularly attractive and may well use this approach in the future. Of course, the appearance of the unit only serves to enhance the superb sound

PHOTO 1: THOR in Peruvian walnut.

PHOTO 2: THOR 1 with grille.

PHOTO 3: Froy 3 in oiled walnut finish.

qualities that Dr. D'Appolito built into the design of THOR (to whom most of the credit must be attributed).

Having "lived" with a pair of THORs for over a year now (*Photos 1* and *2*), I can attest to the superb performance of these units. I have never heard units (at any price) which image so incredibly well—the units themselves completely disappear in the sound field and the transition from woofer to tweeter is totally indiscernible. As for the timbre of the instruments (particularly woodwinds), the THORs rank with the very best I have heard in my 45 years of "hi-fi-ing." Nor should anyone worry about the low end cutoff, which in my listening room goes down to 30Hz at –3dB (admittedly they are placed only 18″ from the back wall).

I do have one constructive suggestion for the crossover unit (which is mounted on the back of my units). On my initial build I used fast Solen capacitors in the notch filter which did not suppress the woofer cone resonance sufficiently well for a smooth response. On measurement I was getting a notch depth of only –18dB at 4300Hz (relative to 1kHz) on both units; I then changed the capacitors to high-current (expensive) Multicaps (from the Parts Connexion) which dropped the notch to –30dB at 4300Hz with a very definite improvement in sound quality—now all the caps in the crossover are high-current Multicaps with very pleasing results—obviously the crossover components have a very definite effect on the performance of these units.

I have only one "complaint" about the THORs—I can no longer sit and read while playing music—the sheer presentation of the music demands your attention like no other system I have heard!

As a side issue, I have also recently built a pair of Froy 3s (*Photo 3*), based on the success of the SEAS units in the THORs. Unfortunately, although relatively good (apart from the bass extension and the "smoothness" of the treble around 8kHz), they do not measure up to the THORs' magic—maybe a little D'Appolito magic is required!

Tom Stirling
stirling@ptd.net

HELP WANTED

A cry for help! The left side of the special cable from the base of my Grace tonearm has a discontinuity in the centerline. I've tried to repair it but to no avail. Might you know where I can mail-order a new assembly? I live in a small town with no local audio supplier.

A.J. Steen
601 N. Kirby St. Sp. 71
Hemet, CA 92545-5910

During a recent holiday weekend trip, I came across a brand of loudspeakers that I never heard of before. The brand name is "Royal Copenhagen." These drivers are made in Denmark. Can you give me any feedback on these loudspeakers and if they are still in production?

Steve Lukas
Slukas9020@aol.com

Are there any specs for the RCA LC1as anywhere? I want to build a proper enclosure for them, but those who have knowledge of them tell me not to use one of the old designs—that it can be done better—but I can't seem to get specs together in order to come up with an update. Any info or way to go would be appreciated. Thanks.

John J. Via
2118 Acklen Ave # 12
Nashville TN 37212
615-260-9569

I have an octal, black metal solid-state plug-in rectifier used (possibly) as a direct replacement for some vacuum tube(s). I can't seem to find any information about it. The top of the can has the following: "S-5018" stamped in white paint as the model number (probably), the letters "JC," in white, above and to the right of the model number, and to the left of "JC," in white, is the schematic symbol of what I think is the old silicon-controlled rectifier symbol: a diode symbol with an "S-hook" replacing the anode lead. If anyone has some info on this device, I would appreciate hearing from you. Thank you. ❖

John Agugliaro
JAGUGL4546@aol.com

Ask aX

BOX DESIGN

When calculating the dimensions for the bass port in a vented system, do the port dimensions change if you put the port in the back of the cabinet as opposed to the front?

Kyle Hamilton
Kyle@remi.com

G.R. Koonce responds:

The short answer to Mr. Hamilton's question is no. The calculated port dimensions, based on box tuning, do not change with the port's location. However, locating the port on the rear of the cabinet may modify the system response.

Readers whose stock of Speaker Builder *magazines goes back a ways will find previous work covering this topic. See "Ask Speaker Builder"* (SB*, 2/91, p. 78) for a way to estimate any response change for rear port placement.*

The system response change is due to additional phase shift in the port response. This is caused by the time delay as the port's response travels around the box from the rear to the front. In general, the response change due to rear port placement can be ignored unless the box-tuned frequency (f_B) is rather high or the enclosure's dimensions are very large.

There are some advantages to rear port placement as follows:

- *A lot more room is generally available. Many times the front panel is already full of drivers and finding room for the port is a problem. Putting the port on the enclosure's rear can avoid the problems of having the port too close to the box walls or of making the front panel any weaker.*
- *It is outside the grille cloth covered area. Putting grille cloth over the port can reduce the port Q of the vented box, changing the system response. See my earlier article, "Box Models: Benson Versus Small"* (SB *3/95, p. 14, and corrected figure #7 in* SB *4/95, p. 54) for measured effects of grille cloth on port Q and port Q effect on system response. If you do put the port in the grille area, then use as large a port as possible, keep the grille cloth back from the end of the port, and use very transparent grille cloth. If you desire to cover a rear-mounted port, use a rather open screen material instead of grille cloth.*
- *Suppression of high-frequency port leakage. High frequencies are restricted to radiating to only that side of an enclosure where they originate. Low frequencies tend to "diffract" around the enclosure and thus radiate in all directions. High-frequency leakage from a rear-mounted port will thus be suppressed for a vented box that sets well away from all walls.*
- *It might allow a larger diameter duct. I have always believed that ports should be as large as you can make them, and you may be able to use a bigger port if rear mounted. See "How Good Is Your Port?" by Bohdan Raczynski* (audioXpress *9/01, p. 24) for excellent coverage on this topic. The port diameter used by most builders—myself included—is clearly too small. If rear mounting allows a larger or better shaped port, that alone is enough reason to do it.*

The down side to a rear-mounted port is whether you can control the application of the speaker system. I have steered away from rear-mounted ports because I worried about how the new owner would use the speakers. Clearly, if you push the boxes right tight against the rear wall, use them in a bookcase surrounded by books, or—worse yet—build them into the wall, then the rear port is a major problem. For speakers for your own use, where you can control the enclosure placement, the rear port shows the advantages I've listed.

One point to consider is that a rear port complicates testing the boxes. Normally the homebuilder tests low-frequency performance via near-field testing. If you rear-mount the port, you need special software to compute the true system response based on vector summation of the driver and port near-field responses. This software must include effects of the time delay and amplitude suppression on the port response travelling around the box.

Clearly, the rear port prevents testing via burying the box in the ground and also complicates ground-plane testing by modifying the apparent box dimensions and thus the port response suppression. If testing is not planned, then this problem disappears. ❖

New Chips on the Block

By Charles Hansen

AD1959 PLL/Multibit Sigma-Delta DAC

The Analog Devices AD1959 PLL/Multibit Sigma-Delta DAC is a complete high-performance single-chip stereo digital audio playback system. It comprises a multi-bit sigma-delta modulator, digital interpolation filters, and analog output drive circuitry. The AD1959 is the first PLLDAC to achieve <100psec jitter and 105dB dynamic range in a single chip. The AD1959 provides three clock outputs: a buffered 27MHz, a 33.868MHz, and a programmable 256/384/512/768 Fs clock. These clocks are used to drive the DVD video encoder, CD-DSP, and audio DACs, respectively.

Other features include an on-chip stereo attenuator and mute, programmed through an SPI-compatible serial control port. The AD1959 is fully compatible with all known DVD formats including 192kHz, as well as 96kHz sample frequencies and 24-bits. It also is backwards compatible by supporting 50/15ms digital de-emphasis at 32kHz and 48kHz sample rates, and MPEG decoding.

Device pricing $5.50 US (1k pieces)
28-lead SSOP package

Analog Devices Inc.
804 Woburn St.
Wilmington, MA 01887
(781) 937-1428
FAX (781) 937-1021
www.analog.com

The Weber VST Copper Cap Rectifiers

Weber Copper Cap Rectifiers are solid-state rectifier direct plug-in replacements for vintage tube rectifiers. Each model emulates the forward conduction resistance curve of the tube type it replaces and also has linear in-rush current limiting to emulate the warm-up time of a tube rectifier. The envelopes are solid hard-drawn copper with thermal capacity that allows a 100% continuous overload condition.

The Copper Cap emulates a tube rectifier, but it does not require filament power. In the case of a 5U4, for instance, this is a savings of 15W. The power transformer will run cooler and last longer. All Copper Cap models are rated at 2000 PIV.

Pricing: $8 to $20 (standard models). Fast-recovery (FRED) diodes are available in all models by special order.

Weber VST
329 East Firmin St.
Kokomo, IN 46902
(765) 452-1249
www.webervst.com\ccap.html

Classifieds

VENDORS

AUDIOCLASSICS.com Buys-Sells-Trades-Repairs McIntosh, B&W, Krell, Klipsch, Tannoy, CJ, VPI, Sunfire, Marantz, Levinson, Lexicon & more. **800-321-2834**.

American Science & Surplus. Audio, electronic and mechanical components, connectors, wiring, kits, magnets, speakers, tools, hardware, and more! Call for a free catalog: 847-647-0011 or shop **www.sciplus.com**.

Canada's source for AudioQuest, QED, Kimber, WBT, other DIY parts and accessories. **www.ozenterprises.net**

Build your own loudspeakers with top components from

www.E-Speakers.com

www.kleintechsys.com

COST EFFECTIVE SPEAKER CABLE

High Purity (>99.99%Cu) 12 Gauge, Prepared for Instant Hookup, NEXT DAY SHIPPING, On-Line Ordering, 20 to 80 feet Lengths! Don't Sacrifice Frequency Reproduction or Power Loss. Don't be fooled by high cost "magic" Cable.
www.kleintechsys.com cable@kleintechsys.com p.561.969.2298

Creative Sound Solutions
www.creativesound.ca

Introducing the ELF Spires
...the taller brother to the SuperELF and the ELF2.0.
See website for details.

BILLINGTON EXPORT LIMITED

All types of audio tubes. 300B 6DJ8 ECC81 ECC83 KT88 Mullard GEC Sylvania. Discount for large quantity. Billington UK. Tel (0)1403 784961. Fax (0)1403 783519. Email/website www.bel-tubes.co.uk

WWW.AUDIOELECTRONICSUPPLY.COM—We are your supply source for all electronic parts—tubes, transformers, capacitors, potentiometers, connectors, and so on, as well as DIY kits and assembled products. Call **(919) 355-0014** or e-mail us at **info@audioelectronicsupply.com**.

Outboard DAC, Headphone amp, Ribbon/Millennium power amps
www.borbelyaudio.com
Borbely Audio kits in USA:
www.audiokits.com
Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio/

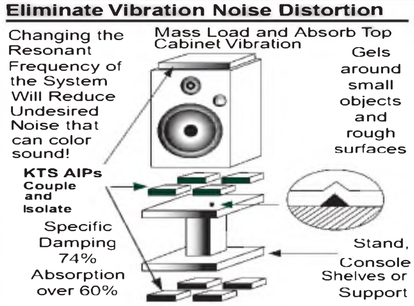
Principles of Power, tube audio books, kits, FAQ **www.londonpower.com.**

Welborne Labs

The largest selection of tubes, audio kits, and parts. It's all online at
http://www.welbornelabs.com

gear racks, media drawers and more
Factory direct since 1984
www.rackittm.com

Eliminate Vibration Noise Distortion



Secures Cabinets/Components to Wood Tile Metal
Non-Toxic, Non-Destructive Removable
www.kleintechsys.com P. 561-969-2298
email: support@kleintechsys.com

THORENS

The New Generation TD800 Series

See all the new products at www.thorens.com
Distributed by:
Trian Electronics 608-850-3600 or www.triancorp.com

www.classifiedaudio.com
High end audio.
Buy it. Sell it. Revel in it.

FARADAY loudspeakers eliminate cabinet vibrations without bracing or damping! See reviews at: **www.faradaysound.co.uk**

Sonic Craft is your high-end speaker source! Accuton, AudioCap, Axon, Alpha-Core, Blackhole 5, Cardas, Goertz, Mills, Sonicap, and kits are on the way! Call 940-689-9800, or see **www.soniccraft.com**

WWW.TESTEQUIPMENTDEPOT.COM
Large supply of New & Pre-Owned Electronic Test Equipment—Oscilloscopes, Power Supplies, Spectrum Analyzers, DMMs, Network Analyzers, and lots more. Call us at **800-275-6261**

WANTED

WANTED Dynaco MKII or MKIII. Also, Pat-4 preamp, any Eico Amp HF-20, HF-22, HF-50, HF-60. W.R. Smitherman **336-699-8699**, 9401 Hwy 67, East Bend, NC 27018

tubes?
www.tubes.it

Want the best price/performance kitset audiophile amp on the planet? Need service too? Can't be done? Visit:
www.aksaonline.com

Precision Port™ *Be Different - Sound Better*

CLEANER SOUND
LESS DISTORTION
EASY INSTALLATION
GREATER SOUND PRESSURE LEVELS
DESIGNED AND MADE IN THE USA

Precision Sound Products, Inc.
Phone: 815-599-0662
Fax: 309-279-0359 www.psp-inc.com

Visit our website:
www.audioXpress.com

Yard Sale

FOR SALE

The original DR Horn prototypes. DR8, $149/pair; DR10, $299/pair; DR 10a, $149; DR12a $199; DR 15a, $249. The same cabs from the pages of *aX*. Must pick up in Laconia, NH. Also, JBL E140, $199 pair; JBL 2446/2370a, $199/pair. Bill Fitzmaurice, fitzmauricew@aol.com.

Drivers: Early 1960s West German, Helcro 10″ (2 ea.); Helcro 5″ × 7″ (2 ea.); Telfunken 3″ Cone Tweeter (2 ea.); room filling sound on 1W, $275.00 O.B.O.; other high efficiency drivers, call. Large qty. of oil caps, call. N.A.D 3020 Amp, $165.00. Call Jack (828) 883-3115.

DENON DRA-825R Receiver w/video inputs, $70.00; Toshiba ST-420 Tuner, $50.00; Technics SL-5 tangent tracking turntable, $75.00 w/cartridge; Technics SL-6, $75.00 w/cartridge; Technics SLQ-300 Turntable, $85.00, several homebuilt speakers using premium drivers and parts—please inquire—selling cheap. Other stuff for sale also—please inquire. Shay Aboor, 436 N. Washington St., Apt 2B, Hobart, IN 46342, (219) 942-4746 (anytime).

Four Dynaudio 30W100 12″ woofers, $250/pr; two Dynaudio 24W75 240mm woofers, $50/pr; two Cabasse 21NDC 8″ woofers, $200/pr; one pair Aria 5s with Accuton tweeter, oak, no grille, $450/pr; one pair Aria 3s with Accuton tweeter and midrange, white, no grille, $600/pr. Alan at 360-829-1155 or aporter@tx3.com.

WANTED

Old *Audio Engineering* magazines. I am missing July and Sept. of 1947, plus Feb. 1951. Also missing many *Audio* magazines from 1964 to 1971. Mike Stosich, (708) 431-4560, EsotericTT@aol.com.

Power Transformer for Carver Professional PM-175 Power Amp or information regarding where one could be purchased. (804) 780-0331 or roncollier@erols.com.

McGee Radio 1982 Catalog. Willing to pay. e-mail perry@perrymarshall.com.

AD INDEX

Glass Shard

Mu-Stage Split-Load Phase Inverter

(Adapted from *Radio-Electronics*, March 1960)

Audio design engineers are constantly seeking new types of phase inverters that will provide higher gain and lower distortion than existing types. High gain means that fewer voltage amplifiers are required when the amplifier is to be driven by low-level signals. With fewer stages, phase shift is reduced and heavier negative feedback can be used while maintaining a high margin of stability.

Some designers use a high-gain inverter with comparatively high distortion and then reduce distortion with negative feedback. Others prefer a low-distortion inverter, even if its gain is very low. Typical examples of these two schools of thought are the various forms of paraphase and long-tail inverters frequently used in high-quality amplifiers. A typical floating paraphase inverter with an ECC83/12AX7 has a gain of 58 to 62 with distortion ranging from 3.5 to 5.5%. A long-tail pair with the same tube and range of plate voltages has a gain of 25–27 with 1.8% distortion.

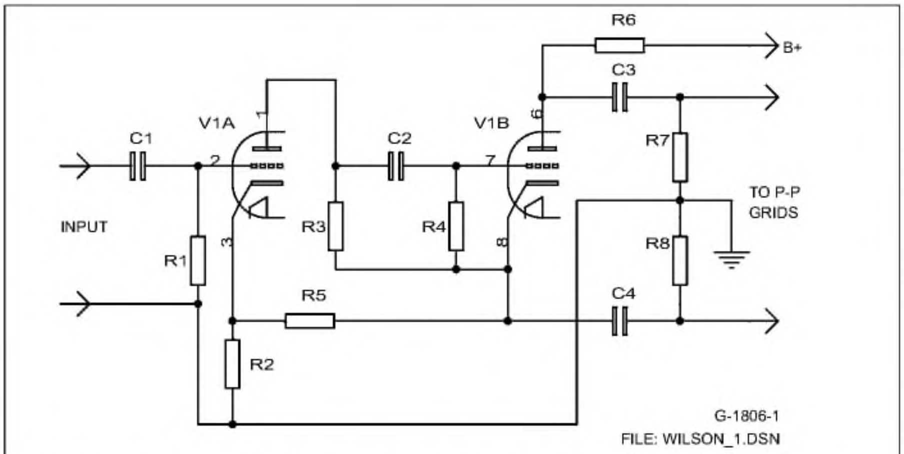


FIGURE 1: This circuit provides gains up to 800 with attenuation of high frequencies. High-frequency losses are reduced when gain is lowered to 200–300.

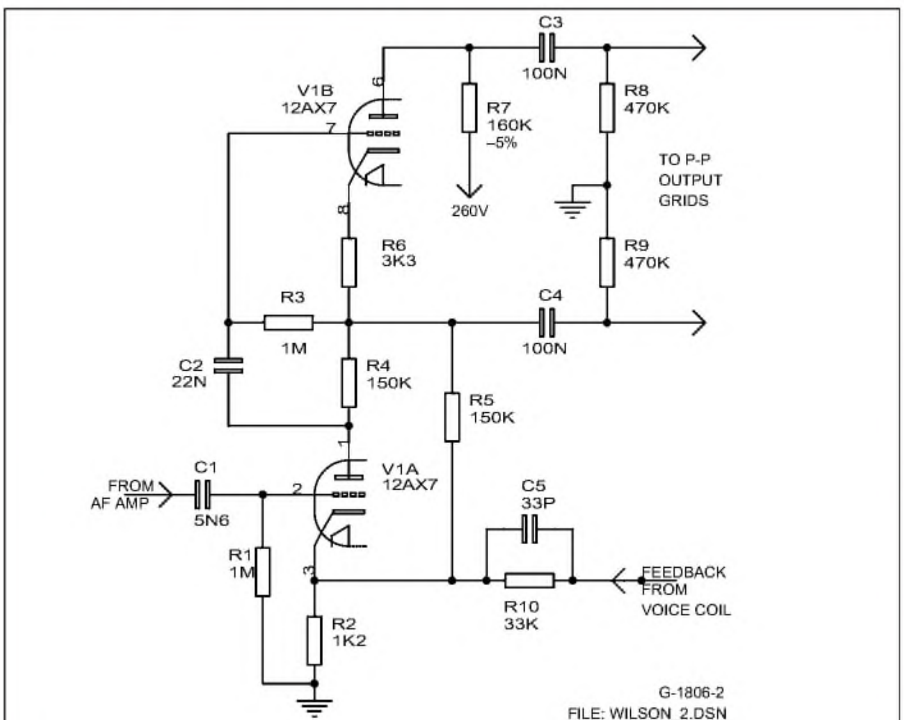


FIGURE 2: High-gain phase inverter as used in Admiral's 5T4-A amplifier.

Design engineers at Amperex Electronic Corp. have developed a new form of split-load inverter in which a weird combination of positive and negative feedback is used to obtain exceptionally high gain with reasonably low distortion. The circuit in *Fig. 1* provides gains up to 800 with considerable attenuation of frequencies in the upper end of the audio range. High-frequency losses are reduced in practical circuits when gain is reduced to 200–300.

In *Fig. 1* the plate of V1-a is fed through R3 from V1-b's cathode. Negative feedback is applied to V1-a by R2, the unbypassed cathode resistor. The low-potential end of cathode resistor R5 is returned to V1-a's cathode and develops a positive feedback voltage across R2. The positive feedback voltage across R2 exceeds the negative feedback voltage and V1-a would tend to be unstable if its plate load resistor (R3) were not connected to V1-b's cathode. However, since R3 returns to V1-b's cathode and V1-a's plate and V1-b's cathode are in phase, enough negative feedback is applied to give the circuit complete stability.

Figure 2 is the circuit—including schematic layout—as used in Admiral's 5T4-A amplifier chassis. Gain is around 200. The circuit is further stabilized and its response widened by applying negative feedback to V1-a's cathode from the output transformer's secondary. ❖

Courtesy of D.G. Wilson
Monrovia, Calif.