Discover the Single-Ended Advantage

audioXPRESS

THE AUDIO TECHNOLOGY AUTHORITY

July 2004
US $7.00 Canada $10.00

Celebrating 35 Years

Build a Low-Cost Phono Preamp

Pure Class A Headphone Amp

BUILDING Subwoofer Boxes

www.audioXpress.com

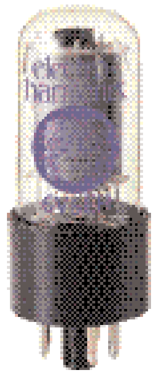
Plus: Testing Audio Interconnects

electro-harmonix

SOULFUL TUBES FOR THE STARS

12AX7EH

Forget about Telefunken! The 12AX7EH provides high gain with ultra low noise and detailed, musical tone that rivals the most desirable NOS types. Unique spiral filament eliminates filament to cathode induced hum common on amps that use AC on the filaments.

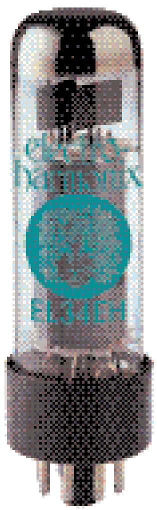


6V6EH

Sweet, singing tone for high plate voltage amps! The all-new 6V6EH combines a specially developed cathode coating, careful alignment of the grids and tri-alloy plate material to create a rugged, wonderful sounding 6V6 that performs flawlessly up to 475 volts. Perfect for high plate voltage amps like the Fender Deluxe Reverb™.

6L6EH

Better than the RCA! Modeled after the vintage RCA 6L6GC "blackplate", the 6L6EH features larger plate dimensions and improved grid structure for increased power handling capabilities. Also features mica spacers with metal springs to eliminate tube rattle and microphonics. The 6L6EH offers tone and performance superior to any other 6L6 or KT66 type.

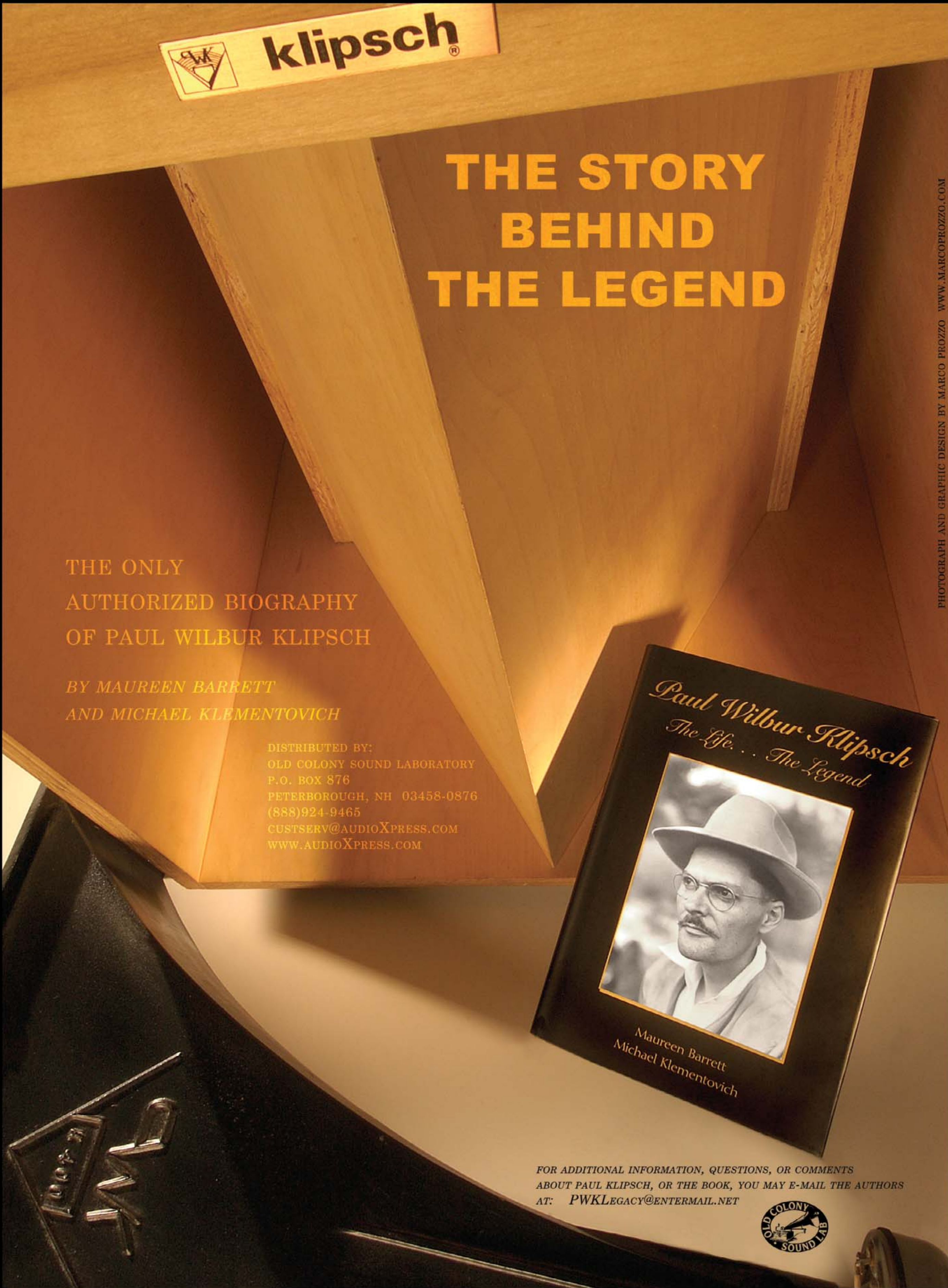


EL34EH

Mullard reincarnated! With the new EL34EH, Marshall™ amp owners can recreate the balanced tone, deep, tight bass, sweet mids, and well-defined highs of the original equipment Mullards. Innovative design and precision construction give the EL34EH exceptionally smooth tone and an ideal balance of pleasing even-order harmonics while virtually eliminating undesirable odd-order harmonics.

www.ehx.com

Available at leading music stores and service shops throughout the world!



PHOTOGRAPH AND GRAPHIC DESIGN BY MARCO PROZZO WWW.MARCOPROZZO.COM

audioXPRESS

CONTENTS

VOLUME 35 NUMBER 7 JULY 2004

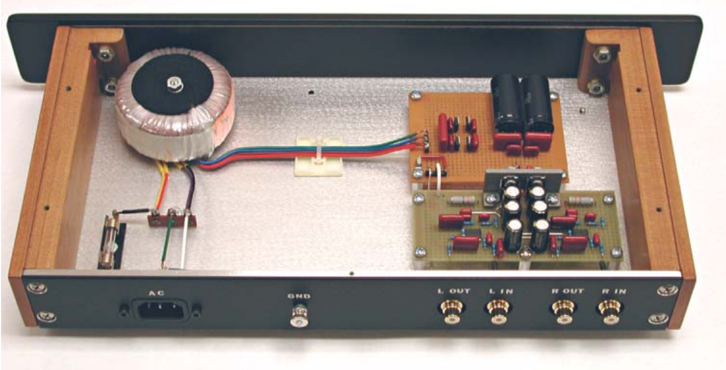
FEATURES

REVIEWS

page 28

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.

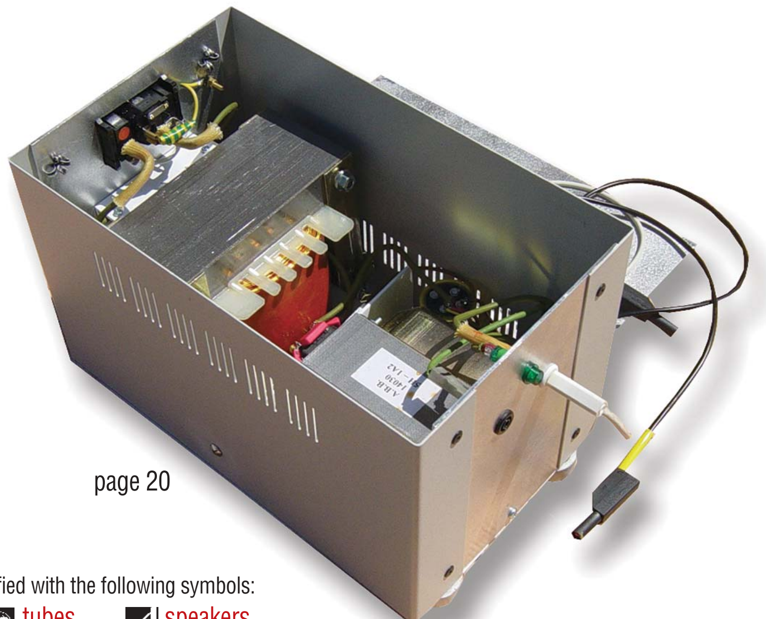
POSTMASTER: Send address changes to: *audioXpress*, PO Box 876, Peterborough, NH 03458-0876.



page 6

DEPARTMENTS

IN EVERY ISSUE



page 20

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**
Richard Campbell **Joseph D'Appolito**
Vance Dickason **Bill Fitzmaurice**
Gary Galo **Charles Hansen**
G.R. Koonce **James Moriyasu**
Nelson Pass **Richard Pierce**
Paul Stamler **Reg Williamson**

Vice President
Karen Hebert

Dennis Brisson Assistant Publisher
Amity Palmer Editorial Assistant
Richard Surrette Editorial Assistant
Tina Hoppock Graphics Director
Ben Leece Production Assistant
Laurel Humphrey Marketing Director
Lora Cunningham Marketing Assistant
Kelly Bennett Customer Service
Sharon LeClair 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: peter@smmarketing.us
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. All subscriptions begin with the current issue. 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.

Robert M. Bullock III

Robert M. Bullock III

It is with deep sadness that we announce the untimely death of one of audio's best friends. He passed away on February 23, 2004, at his home in Oxford, Ohio. He was 66.

Dr. Bullock taught mathematics at Miami University in Oxford from 1966 until his retirement as Professor Emeritus in 1999. He taught previously for one year at Cincinnati Country Day School and for three years at the University of Cincinnati, where he earned bachelor's, master's, and doctoral degrees. He served in the U.S. Air Force from 1955–59.

Bob Bullock loved loudspeakers and loudspeaker technology. He enjoyed building loudspeakers and was an early advocate for the work of Neville Thiele and Richard Small. He was an early subscriber to *Audio Amateur* and one of the first contributors to *Speaker Builder* magazine, launched in 1980. His review of the third edition of "The Loudspeaker Design Cookbook," by (the then unknown) Vance Dickason set the staff on a search for the book's author.

Before the end of *SB*'s first year, Bob wrote the first of a long series of definitive articles on box design. These supplied excellent alignment tables, making box design all that much easier. Eventually, teaming up with Bob White, he wrote some of the first software for speaker modeling. He was also one of the first Contributing Editors for *Speaker Builder* and later to *audioXpress.*

He continued his series of *Speaker Builder* articles, covering five on alignments in addition to six more on computer- and calculator-based design. He answered dozens of letters from readers, as well. His papers delivered before the Audio Engineering Society were exceptionally well received and referenced by others. Eventually, in 1991, we gathered all his work, along with that of Bob White, into a book titled "Bullock on Boxes," which is still in print. He was a very quiet, self-effacing man with a keen sense of humor. His interest was always on the project at hand and seldom on his own person.

He belonged to the Society of Industrial and Applied Mathematics, the Mathematical Association of America and the Audio Engineering Society. He is survived by his wife Evonne, of Oxford, two children Robert M. (Lynn) Bullock IV, of Fairfield, Ohio, and Tanja (Mark) Bisesi, as well as a brother, Michael (Pat) Bullock of Mt. Pleasant, Tenn. He is also survived by two grandchildren, Abby and Allen Bisesi (Bloomington, Ind.), and two step-grandchildren, Luke and Corey Crawford (Fairfield, Ohio.)—E.T.D. ❖

nextgen™

The exact solution for all digital connections

- the perfect match for all digital audio and video links (75Ω)
- true wideband RCA type connectors
- free from interferences
- with pure copper or silver contacts
- ultra rapid signal transfer

Ph: 801-621-1500
www.wbtusa.com

WBT

A High-Quality MM IC Preamp

This author achieves "analog ecstasy on a shoestring budget" with this low-cost phono preamp design.

By Dan Stanley

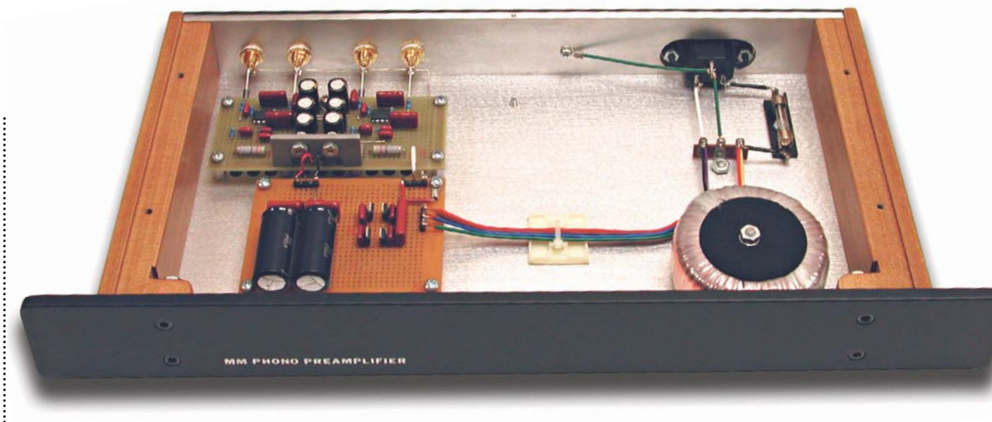


PHOTO 1: Front overall view of preamplifier.

If you are like me, you were probably won over to CD sound reproduction by the medium's convenience, nonexistent noise, and promise of "perfect sound forever." I gladly abandoned (sold) most of my vinyl records, with their surface noise, inner groove distortions, and time-consuming cleaning rituals. However, in the years since my switchover, I have had opportunities to listen to records with increasingly better turntables, cartridges, and electronics.

While I still like the convenience of my CD discs, I must admit that today's good vinyl playback equipment can produce a degree of musical realism and satisfaction that my CD player (even with extensive mods) often cannot match. In an attempt to recapture some of the magic of listening to vinyl, I decided to pull out my audio parts box and build a dedicated phono preamplifier for the remaining records that I (thankfully!) still possess.

SELECTING A CIRCUIT DESIGN

I started out with the goal of building a design that would be simple, very inexpensive (under $150), and capable of relatively high quality performance. During my searches and queries, I ran across a very promising configuration that uses a single high-performance op amp with an active RIAA feedback network. Highlights of this design include a feedback network that plateaus the RIAA curve above 50kHz. A great many records have been cut with this sort of equalization modification to save cutter heads from burnout damage, and taking this into account will

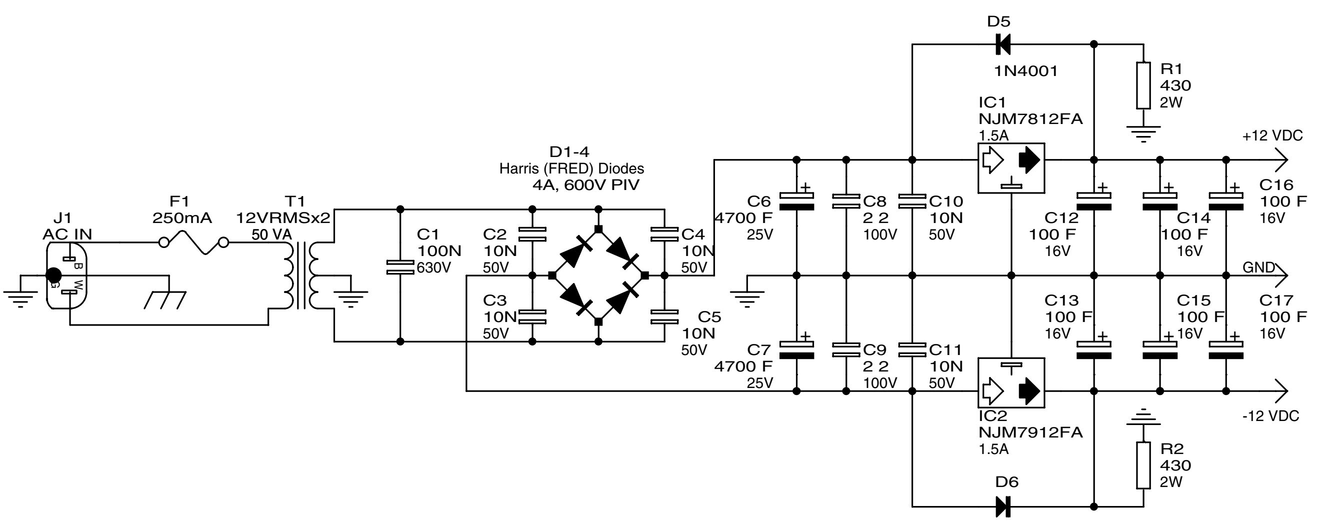


FIGURE 1: Preamplifier power supply.

NEW !!

DH LABS

SILVER SONIC™

HIGH PERFORMANCE AUDIO CABLES

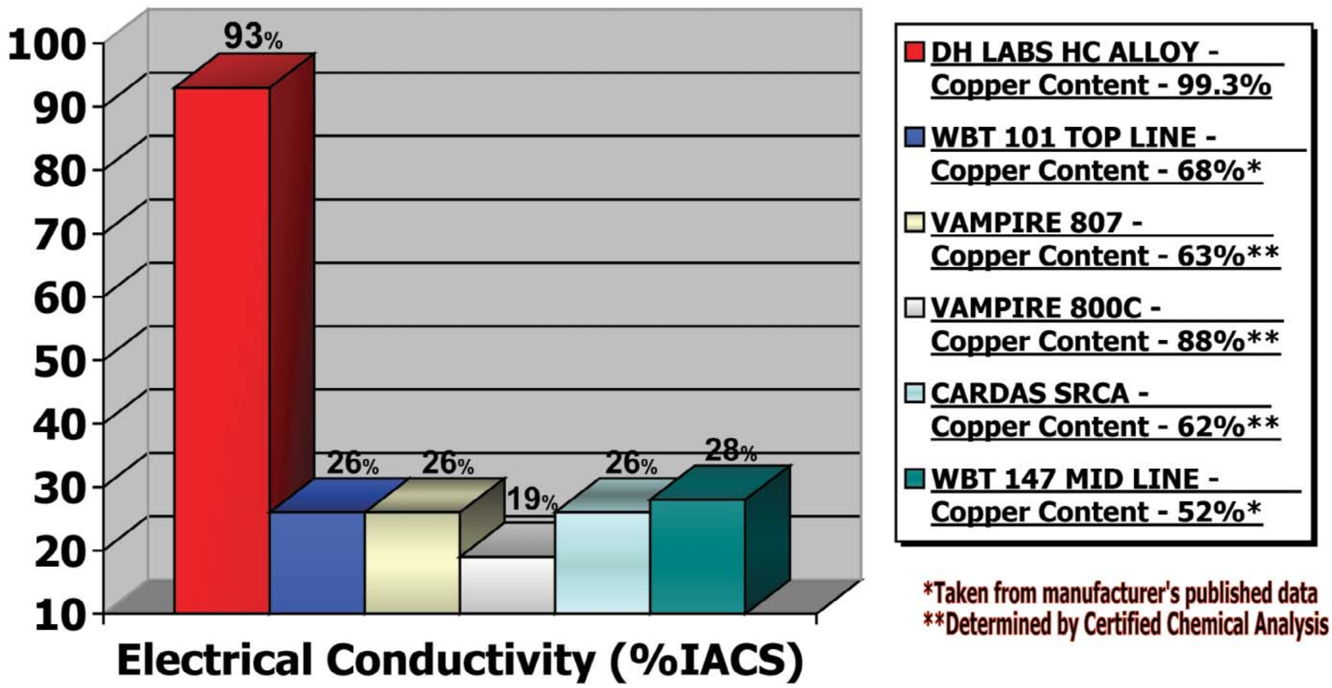
The Ultimate RCA Connector

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

add a sense of openness and air to record playback.

This design also maintains a reasonable network loading of the op amp at higher frequencies, and the ability to couple the input signal without a capacitor.

THE POWER SUPPLY

The power system is a key factor in any audio component, and it can make a very significant sonic difference. The best sonics with this design will be produced using 12V DC lead acid batteries, provided that they have a very low internal resistance. I corresponded with one audio enthusiast who used batteries, and even installed a trickle charger in the preamp housing to maintain them. Due to cost and size considerations, I decided to use an AC-derived supply, which would also allow the unit to remain on for very extended break-in and warm-up periods.

My supply design, which is a balancing act between cost and performance, is shown in *Fig. 1*. The AC is input via a rear-mounted IEC connector, with the earth connection attached to the preamp case for shielding. The earth ground is also routed to a rear panel lug for a turntable ground wire, and there is a circuit board jumper connecting the power supply ground to the earth ground. The fused power is routed to a 50VA toroidal transformer capable of sourcing over 4A of current.

The secondary output is noise filtered and then rectified by a full wave bridge made up of 4A super-fast soft-recovery diodes, which have 0.01µF film capacitor noise snubbers in parallel. The rectified voltages are smoothed by low impedance 4700µF electrolytic capacitors, which have 2.2µF and 0.01µF film capacitor bypasses. This configuration produces approximately ±17V DC (under load from the circuit) to feed the voltage regulators on the preamplifier board.

A few notes are in order about the raw DC supply system. I used a toroidal-type transformer to avoid the expense of an outboard housing, cable, and connectors. You should still keep the transformer 8″ or more from the active circuit area to avoid induced circuit hum. Additional transformer shielding might reduce the required distance.

The super-fast soft-recovery diodes with snubbers greatly reduce diode-switching noise that might get through to the power rails. I also used point-to-point wiring and thick copper wire ground traces on the power board, which was fabricated using a standard

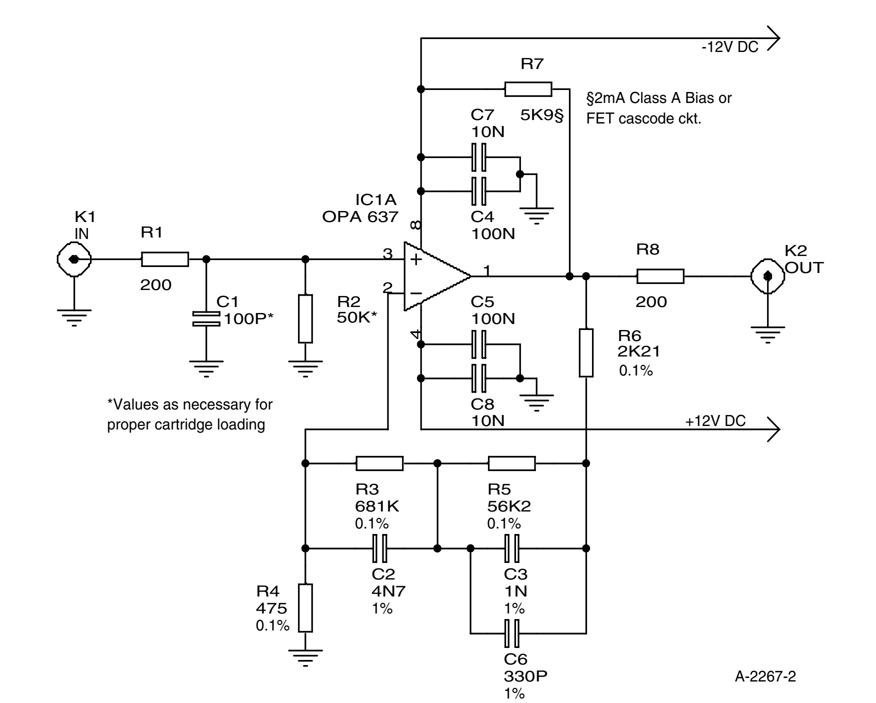


FIGURE 2: Phono preamplifier circuit.

pad-per-hole prototyping board.

The filtered DC from the power board feeds a pair of ±12V DC JRC (NJM) three-terminal regulators mounted on the main circuit board close to the op amps. The units chosen have very good ripple rejection specifications and low noise. This is a spot where, for still higher performance (at a cost), you would do well to check into some of the available higher-performance Linear Technology IC regulators. The outputs of the regulators each have three paralleled Black Gate F 100µF electrolytic capacitors. Note that there are also power resistors on the regulator outputs to produce a total current pre–load of approximately 42mA to bias the regulators into a region of better performance.

This supply design is much better than those of typical inexpensive commercial phono preamps. It provides a cost-effective level of sonic performance that is well suited to the goals of this project.

THE PREAMPLIFIER CIRCUIT BOARD

In an attempt to wring out maximum performance at cost, I employed several design tactics on the phono circuit board. I pre-mapped on graph paper the complete layout of all components, connections, voltage rails, and ground traces to ensure a compact, symmetric layout with minimum connections and solder joints. All of the wiring on the board is point-to-point, with no dielectric insulation involved, except at the solder pads on the board. In most cases, the leads of components serve as the traces between circuit points, and are positioned several thousandths of an inch away from the plane of the board. Note that the legs of the input and output resistors are directly soldered to the RCA jacks on the back panel to avoid the need for internal hook-up wires.

I also tried to adhere, where possible, to individual low resistance ground return paths on the phono board. There is a completely separate path from the RCA jack ground lugs to the power supply ground using bare Wonder Wire. The board, which is made of high-grade glass-epoxy material, is a prototyping style with plated copper pads per hole on 0.1″ centers. Both of the circuit boards of the preamplifier were very solidly mounted using 0.5″ threaded hex aluminum spacers.

THE PREAMPLIFIER CIRCUIT

As I previously stated, I was attracted to the simplicity of a single, active feedback op-amp gain-stage design, even though I am aware of the arguments for a buffered passive RIAA equalization scheme. Both approaches have their advantages, but in the interest of cost and reduced circuit complexity, I went with the single-stage concept. This particular RIAA feedback network configuration, which was produced by audio enthusiast Thorsten Loesch[1], has 44dB of gain at 1kHz, and is well suited to moving magnet cartridges with 3 to 5mV output levels. There is also enough gain for some of the high-output MC cartridges. The circuit is illustrated in *Fig. 2*.

The op amp used is the Burr-Brown OPA 637, which is a premium-grade FET input design with a very high slew rate, fast settling, low noise, and mini-

PREAMPLIFIER PARTS LIST

DESCRIPTION	QUANTITY	REMARKS
RESISTORS: (¼W UNLESS NOTED)		
100Ω, 1%, metal film	2	2mA current source (if used)
200Ω, 1%, metal film	4	Input/output
430Ω, 5% metal oxide, 2W	2	Regulator output current bias
475Ω, 0.1% metal film	2	RIAA, hand-matched
2.21kΩ, 0.1% metal film	2	RIAA, hand-matched
5.90kΩ, 1% metal film	2	Op amp class A 2.0mA bias
681kΩ, 0.1% metal film	2	RIAA, hand-matched
50.0kΩ, 1% metal film	2	Cartridge loading
56.2kΩ, 0.1% metal film	2	RIAA, hand-matched
CAPACITORS		
100pF, polypropylene, 50V DC, 2%	2	Panasonic P series cartridge load
330pF, polypropylene, 50V DC, 2%	2	Panasonic P series, RIAA, matched
0.001µF, polypropylene, 50V DC, 2%	2	Panasonic P series, RIAA, matched
0.0047µF, polypropylene, 50V DC, 2%	2	Panasonic P series, RIAA, matched
0.01µF, polypropylene, 50V DC, 2%	8	Panasonic P series bypassing
0.1µF, polypropylene, 50V DC, 2%	2	Panasonic P series bypassing
0.1µF, polyester, 630V DC, 10%	1	Panasonic ECQ-E(F) series T1 sec.
2.2µF, polyester, 100V DC, 10%	2	Panasonic ECQ-E(F) series bypassing
100µF, electrolytic, 16V DC	6	Black Gate standard series
4700µF, electrolytic, 25V DC	2	Panasonic FC series
SEMICONDUCTORS		
Diode, 1N4001	2	Regulator protection
Diode, 4A, 600 PIV, TO251	4	Harris hyperfast FRED RHR 460
2N4304 FET transistor	4	2mA current source (if used)
OPA637AP DiFET op amp	2	Burr-Brown (TI), 8 pin DIP
OPA604AP FET op amp	2	Burr-Brown (low-cost alternate)
VOLTAGE REGULATORS		
NJM 7812FA, +12V DC, 1.5A	1	JRC, 3 terminal, TO220
NJM 7912FA, –12V DC, 1.5A	1	JRC, 3 terminal, TO220
CONNECTORS		
RCA jack, gold over brass	4	Standard type, inside nut
IEC power input socket	1	Standard type, 120V AC
TRANSFORMER		
4.166A, 12V RMS × 2, 50VA	1	AMVECO toroidal #62082 Digikey #TE62082-ND
WIRE		
Bare Wonder Wire		Ground returns from I/O jacks
16–18 gauge hook-up wire		Standard type
MISCELLANEOUS		
Fiberglass prototype board		0.1″ pad per hole, tinned
Preamplifier enclosure		Aluminum preferred
Rubber isolation feet		Highly damped preferred
Line cored with IEC connector		Standard 120V AC
Terminal strip, 3 lug with base		Standard type
Aluminum hex standoff posts, 0.5″		With 6–32 threaded ends
Regulator heatsink(s)		Standard type for TO220
Fastening hardware and wire lugs		As needed
Fuse Holder		Std. 1¼″ × ¼″ chassis type
Fuse		Std. 1¼″ 250mA SB type

mum input offset voltage. This unit also allows for the omission of a signal input coupling capacitor, which further simplifies the circuit path. If you wish to keep costs down to a minimum, you could use the Burr-Brown OPA 604, although noise and distortion will be a bit higher.

Note that there are film de-coupling capacitors at each DC power feed pin on the op amps, which is important to reduce the possibility of noise and oscillation. There is also a "pull down" resistor between the op amp output pin and the –12V DC power rail to produce a class A output bias of 2mA. You could implement a more advanced bias system using an FET current source (*Fig. 3*). I suggest that all of the film capacitors and metal resistors used in the amplifier's RIAA/gain network be value matched with suitable meters, and that they have very good sonic characteristics.

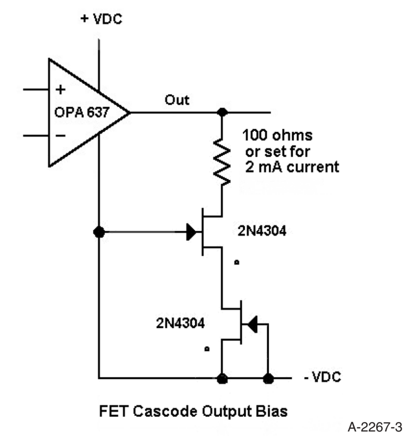


FIGURE 3: FET Cascode output bias.

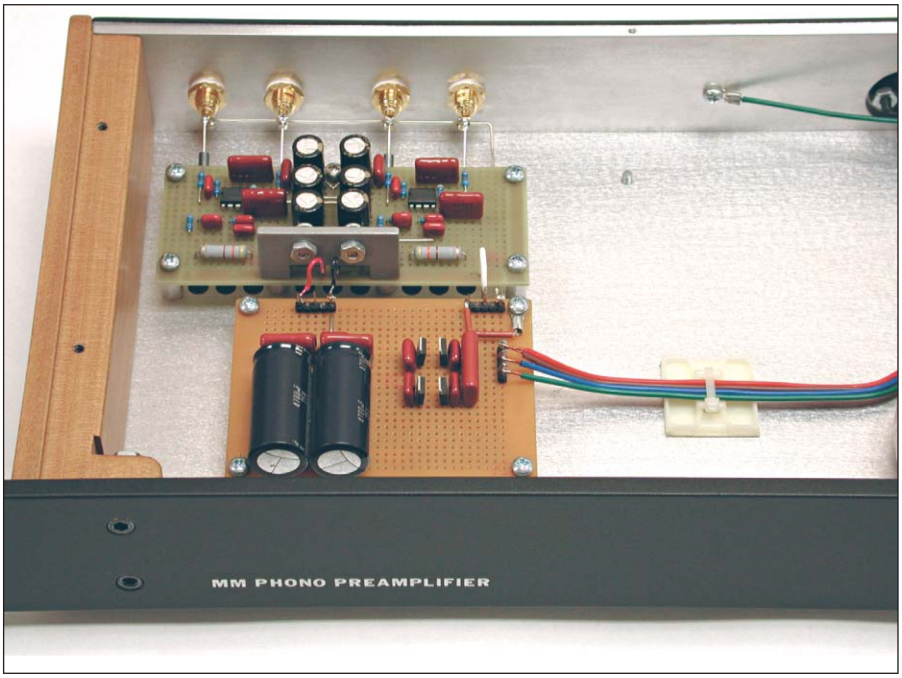


PHOTO 2: Front detail view of preamplifier circuitry.

The input section has a 200Ω series resistor that forms a low-pass filter via the capacitor that sets the desired input capacitive loading for the phono cartridge. This configuration filters out frequencies in the MHz range that might contaminate the input signal to the op amp. If you live in a troublesome RFI area, the addition of a small ferrite bead on the input resistor leg might also be of help. The resistive loading is set to the specifications of the particular cartridge being used.

I did not install any sort of adjustable loading switches to avoid circuit complexities that might compromise the sonic performance. The outputs of the OPA 637 devices are directly fed to the output RCA jacks through 200Ω metal film resistors. I chose not to use an output coupling capacitor to keep costs down and reduce sonic degradation, although it would be easy to add one as part of a first-order high-pass filter to block infrasonic frequencies. If you choose this path, it is wise to keep the filter turnover point well below 20Hz to avoid phase shifts that might color the sound, and select a very high performance capacitor.

In most cases, the components following the phono preamplifier have their own DC blocking input capacitors, and my op amps had only −50mV and +100mV DC offsets for the left and right channel outputs. Note that the output offsets will be significantly higher if you use the OPA 604 op amps.

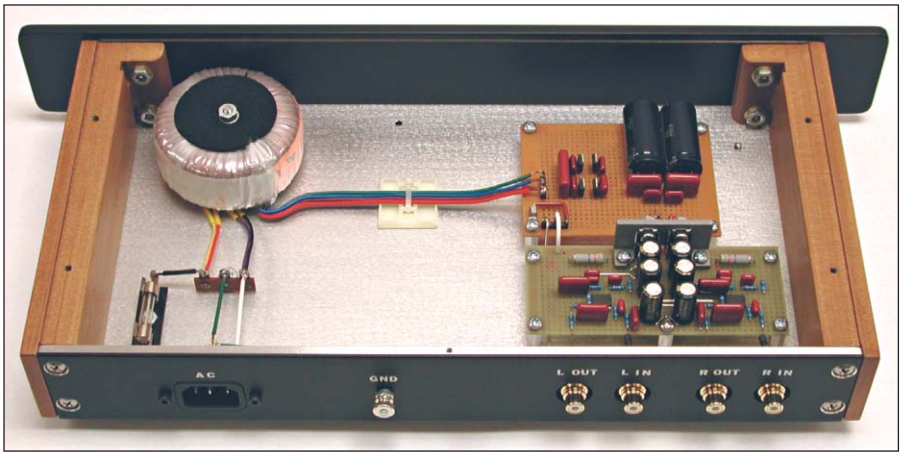


PHOTO 3: Rear overall view of preamplifier.

PHONO BOARD CONSTRUCTION HINTS

With many components closely situated on the phono board, it is a good idea to follow a clean, symmetric layout that has been pre-planned. It is also important to try to produce low resistance/inductance ground return paths that have been separated out as much as possible. When fitting and soldering the various components and trace wires to the perfboard, it is very helpful to use an illuminated magnifier to ensure good solder joints and non-contact of adjacent solder pads. I used a fine gauge version of Wonder Solder that was very easy to work with.

PROJECT RESOURCES

PARTS:

Digikey.com: 1-800-344-4539 (sales)
Voltage regulators, transformers, op amps, capacitors, resistors, hardware.

Graniteaudio.com
CD with RIAA-equalized break-in noise track.

Jameco.com: 1-800-831-4242 (sales)
Pad per hole circuit boards (fine quality, good pricing), also cases, Hardware.

Mcmaster.com: 1-630-833-0300 (sales & customer service)
Mu-metal shielding materials for transformer hum radiation.

Percyaudio.com 1-530-470-8650
Harris super-fast soft recovery power supply diodes, capacitors, resistors.

Mouser.com: 1-800-346-6873 (sales)
Voltage regulators, resistors, capacitors, transformers, hardware.

Radioshack.com: 1-800-472-7221 (commercial sales division)
RCA jacks (quality gold-plated Tiffany style units at an excellent price)
910-0862 (panel mount with red band)
910-0863 (panel mount with white band)

Sescom.com: 785-883-3009 (sales / info)
Very nice aluminum rack chassis kits suitable for this preamplifier design.

PREAMPLIFIER HOUSING

The biggest surprise about the preamplifier housing was the amount of time and effort that it required. I spent significantly more time with this part of the project than the circuitry and components that went inside! Originally I was going to find a commercial case to hold the components, but I decided against it due to the cost of obtaining one with enough space inside to allow keeping the toroidal transformer at a reasonable distance from the circuitry. If you want a good commercial aluminum enclosure, check out reference 2.

I constructed the case entirely out of aluminum, except for the side panels. The front and rear panels are made of 0.25″ milled extrusions, while the bottom panel is of 0.125″ sheet stock. The top panel is made of 0.030″ sheet stock. I made the side panels with milled hardwood.

Almost everything is held together with stainless-steel sheet-metal screws that attach to the wood end panels, except for the front panel, which is attached with black oxide cap screws and nuts. All of the outer surfaces of the aluminum panels were fine-sandblasted as a preparation for spray painting, and the end panels received several hand-rubbed coats of polyurethane. I used some white press type that I had on hand for the lettering, and over-coated the finished front and rear panels with Krylon clear matte spray as a protective measure. I added three vibration-isolating feet to the bottom of the case to finish up.

BREAK IN, TESTING, AND LISTENING

With a little help from my scrap parts box, I spent just $115 to complete the preamplifier, although I believe it would be possible to buy all materials for $150 or less. A higher-performance version (even better RIAA components, regulators, and so on) might run $200. Now the task was to break in the unit, test a few parameters, and maybe even listen to it!

I had no desire to play records for hundreds of hours to properly break in the piece, so I ran a series of very low level signals into the preamplifier while loading the outputs with 10kΩ resistors. I later located a specially produced CD with RIAA-equalized broadband noise that was ideal for such a break-in procedure.

When the break-in was completed, I decided to run a few tests on the preamp to check aspects of its performance. One test involved measuring the 60Hz hum level in each channel relative to a 4mV input sine wave at 1kHz, which matched the output of my car-

to p. 67

REFERENCES

1. www.users.nac.net/markowitzgd/phonopre.html
2. http://tangentsoft.net/audio/opamp-bias.html

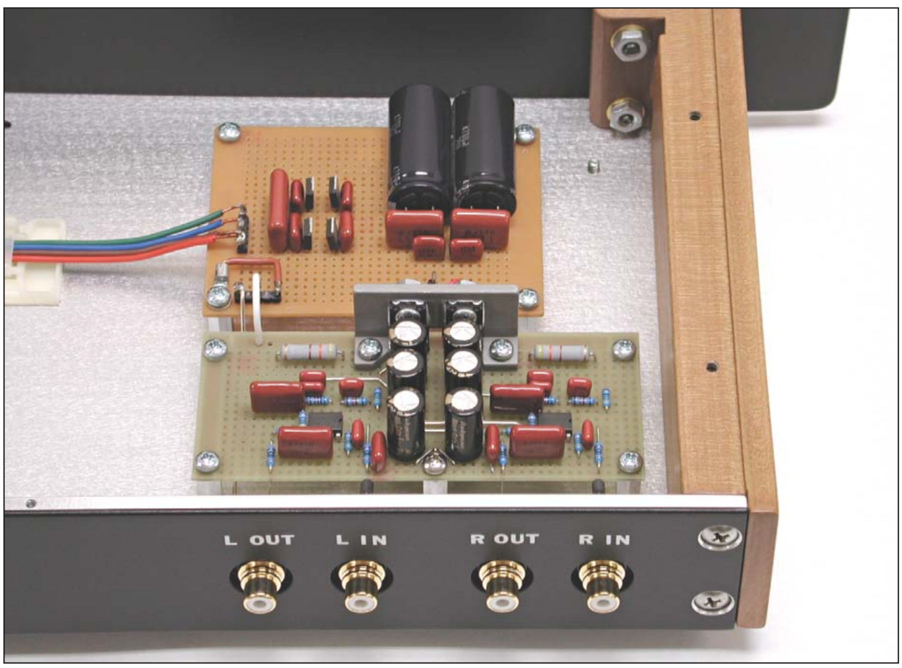


PHOTO 4: Rear detail view of preamplifier circuitry.

A Battery-Powered Class A Headphone Amp

This recording specialist shares his experiences with a proven design that adds to your music listening pleasure. **By Rick MacDonald**

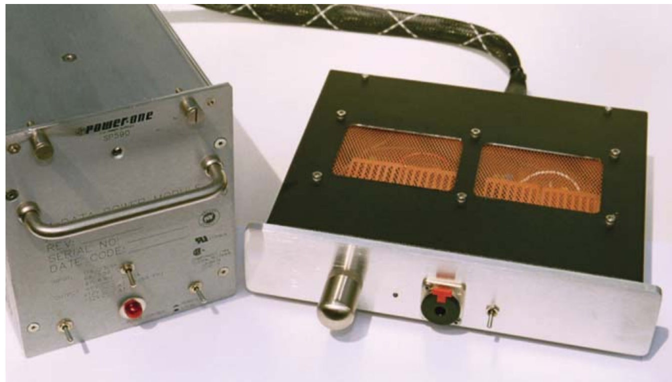


PHOTO 1: Completed amp with battery PSU/charger.

One of the most rewarding ways to listen to music—without spending astronomical amounts of money—is with high-quality headphones fed by a well-designed HP amp. Having to work with headphones while making music and sound effects has exposed me to a lot of HP amps. I've also encountered mixing consoles, DAT, CD, DAW (digital audio workstations)—all op-amp based, and all low-cost designs.

It was time for something better (*Photo 1*). Rather than going with a new design and engineering model, I wanted a tested, proven, discrete Class A circuit. Several friends suggested Erno Borbely's MOSFET amp, and with the variety of ways you can buy the PCB and components, it was the best option. With the EB-602/210T design, you can buy just the PCB, the PCB with all components, or the PCB with semis only (*Photo 2*). I recommend the latter because some of the FETs may be hard to source, and the matching (if needed) is done for you.

The rest was provided from surplus and recycled gear, but there will be more on that later. I will admit I had never heard a pure Class A HP amp. The sound for such a simple and elegant design is better than much more expensive "high-end" systems I have auditioned. This circuit has good detail, dynamics, and low-level resolution. However, I did hear differences depending on the power supply used.

PURE CLASS

I want to thank Erno Borbely for his assistance, permission to reprint his designs, and for his contributions. The EB-602/210 is a single-ended pure Class A amp capable of driving headphones between 32 and 600Ω. It consists of two identical but separate amps laid out on the same PCB.

The EB-602/210 requires regulated ±15V to 24V power, and separate supplies are recommended. The schematic (*Fig. 1*) is the same as Erno's hybrid tube/MOSFET line amp, published in *Glass Audio* 1/98. However, the dual triode at the input is replaced by a dual JFET (SK389). The JFET offers less noise, more gain, and better linearity than the ECC86 tube.

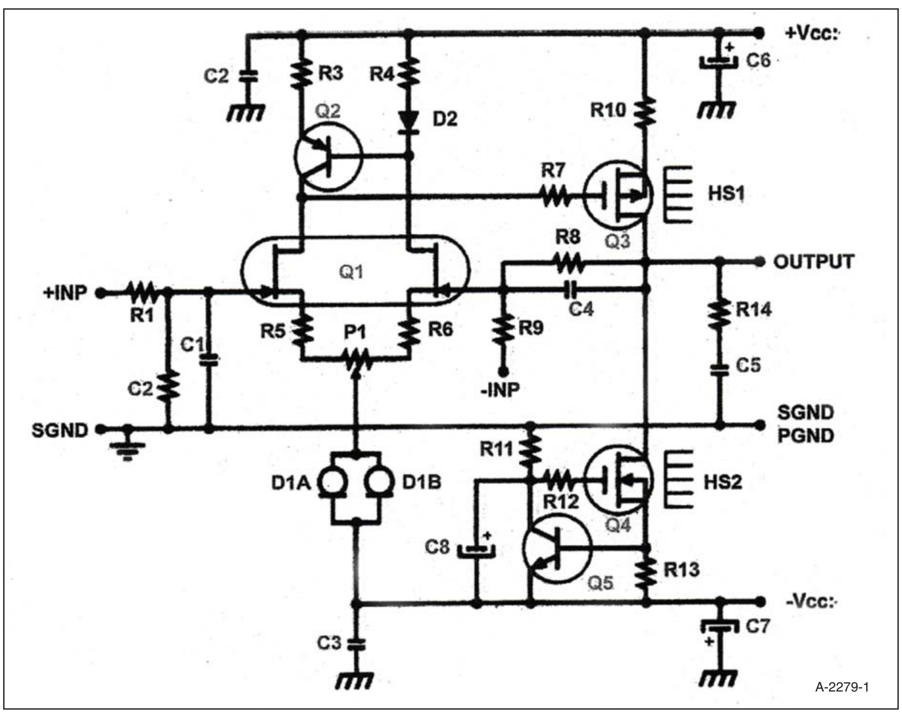


FIGURE 1: SE Class-A Line Headphone Amplifier EB-602/210.

ABOUT THE AUTHOR

Rick MacDonald, after 15 years as an engineering tech designing and prototyping broadband RF antennae and GPS satellite subsystems, moved into software engineering with E-Systems/Raytheon. Rick's most recent work has been with digital audio, composing music with MIDI and editing audio for games, software, kiosks, and websites for Gateway, Tandy, and Voyetra, among others. Rick has also written articles for *Recording* magazine, and has published several books on interactive audio and computer music. He continues to pursue the perfect power supply for all his audio projects.

It operates as a differential amplifier with about 2mA in each JFET and is supplied by the constant-current diode D1, which is made up of two J508 2.4mA diodes in parallel (or a single J511-4.7mA diode). Q2, D2 comprise a current mirror which converts the out-of-phase signals from the Q1 drains to a single-ended signal. Q3 is a 2SJ79 P-channel MOSFET (TO-220) used in common-source mode as a Class A output stage.

Its drain resistor, replaced by a second constant-current source, supplies 160mA, thus increasing gain and linearity of the second stage, which is made up of Q4 and associated parts. Q4 is an N-channel MOSFET (2SK216) in a TO-220 package.

Open loop gain of the amp is 67dB. As used in this project, with ±12.5V, Erno suggested I start with the same resistor values as the 15V version, so that I get 160mA for the constant-current source. The resultant maximum power into load is:

300Ω: 200mW
100Ω: 462mW
50Ω: 300mW
32Ω: 219mW

I have tried many types and various impedance headphones from 30Ω to over 300Ω. These include SONY MDR-777 PRO (70Ω), Sennheiser HD570 (62Ω), and some inexpensive 30Ω and 300Ω cans with no sign of clipping and plenty of volume on the amp's attenuator.

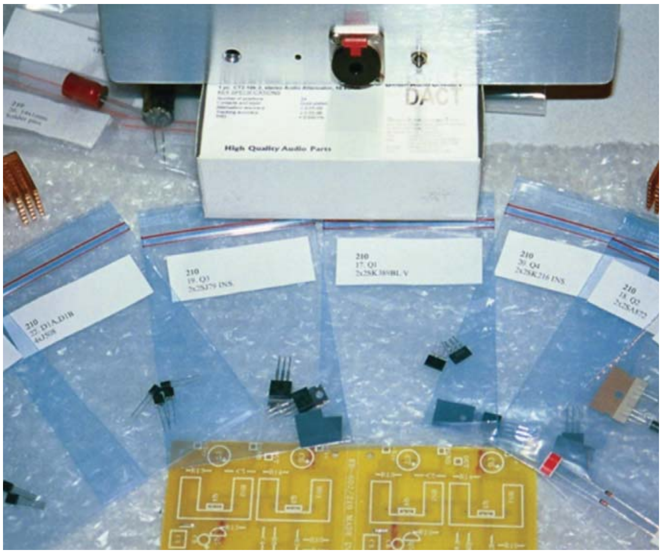


PHOTO 2: Borbely PCB, FETs, and author's parts.

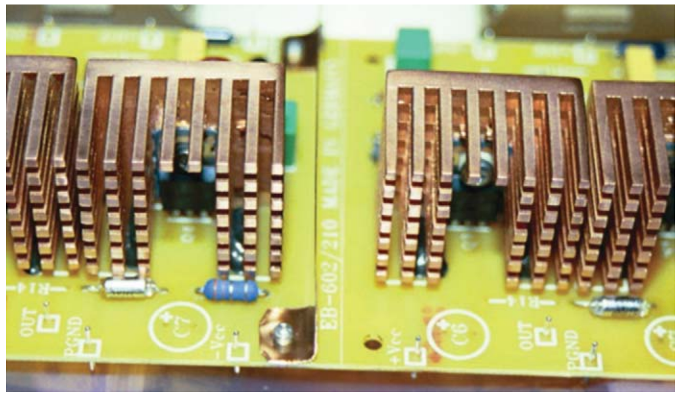


PHOTO 3: Partially stuffed PCB with copper heatsinks.

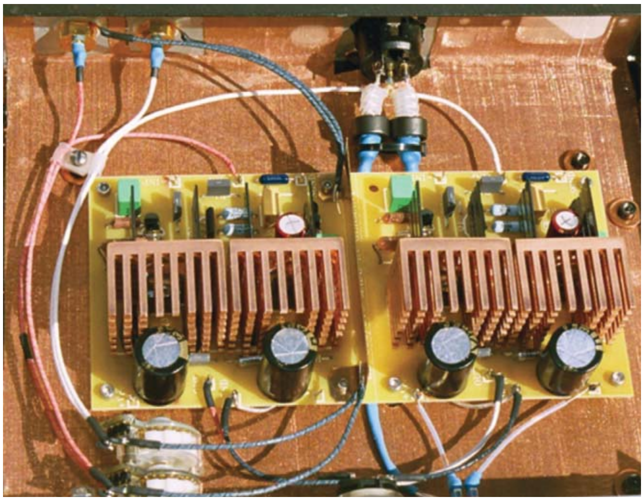


PHOTO 4: Internal view of completed amp.

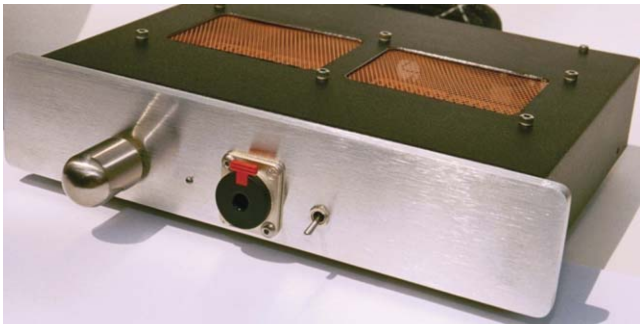


PHOTO 5: Completed amp with power umbilical.

BUILDING THE AMP

Construction of the HP amp itself was simple because of the clear documentation sent with it, so I won't spend a lot of time on PCB stuffing. Specific problems I ran into had more to do with my own additions and modifications. Heatsinks were included but were ¼″ too tall for my case.

In *Photos 3* and *4* you can see the copper heatsinks originally made for a microprocessor. These had to be cut and machined to fit a TO-220 device perfectly flat, then supported on the PCB with high temp RTV. These keep the temperature close to 48°C on the device. At 12V, R13 = 3R99Ω and R10 = 7R52Ω.

I determined R13 by measuring the voltage (approximately 650mV) across it, and selecting a value slightly above or below 15V values until the current source for Q3 came up to 162mA. Be careful when mounting the MOSFETs to the heatsinks. Use the insulating washers and silpads provided.

Mount MOSFETs on the heatsink first, then solder them to the PCB only after installing all other parts (except C3, C4). Then you can mount C3 and C4.

Use proper static handling methods for the semis and check polarity on all electrolytics. I used polypropylene caps, but Dale RN60 resistors work well except where I installed Caddock MK-132 on the input (R1) and output (R14) where some differences could be heard.

I also used a small amount of Teflon solid core silver wire for all audio connections (*Photo 4*). Short runs and spacing between the wires negated any need for shielded cable inside. I used a copper divider between the channels for all audio grounds (star Gnd), and to keep crosstalk down.

A CASE FOR RECYCLING

Having made a pattern from the PCB stuffing guide, I used it to mark the amp case for drilling standoff holes. While you can get cases from Audiokits.com for both the HP amp and PSU, I recycled an old 16-bit DAC case after no success selling it. I replaced it with a 24-bit TPC DAC-3.

The power supply case was an old server switch-mode supply with a bad HV transformer purchased from a computer recycler. The DAC case had a black anodized finish on the front plate and was fairly beat up, so I removed all of it down to aluminum with

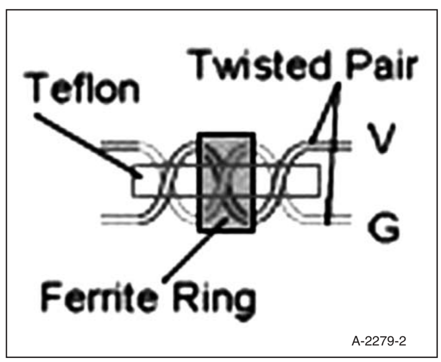


FIGURE 2: Battery PSU with charger.

a belt sander with a medium grit belt. By keeping the sander in line (horizontal), I was able to produce a nice brushed look in that direction (edges too). I then used a finer grit sanding sponge (keeping the same direction) until I achieved a smooth-to-the-touch brushed finish (*Photo 5*).

I first drilled the vent holes in the corners, then cut them with a sawzall. Smoothing the corners with a Dremel and touching up on the edges with black model paint completed the vent openings. I drilled out all holes in panels with a Greenlee stepper bit or Unibit. This is the only way to do holes up to ¾″ diameter for up to ¼″ thicknesses. I mounted the DACT 10K attenuator, Neutrik jack, RCAs, and XLR for power with the unibit.

CHOKES AND COPPER SHIELDING

In a modern environment where so many sources exist for producing and emitting RF and EMI, you must take them into account and control them.

Using batteries is a good place to start, but eliminating the AC connection is not going to stop EMI caused by wireless telecom, HV lines, home wiring, and switching supplies in your computer, to name only a few. RF guys have known this for decades, which is why companies such as Raytheon and Toyota build huge RF anechoic chambers for controlled tests. I was impressed with how well RF screen rooms (copper mesh on a frame around an RF design lab) keep out much of this radiation, so I decided to apply the same priciples to this project.

Since you can't build an anechoic chamber in your homes, use shielding. Pure copper mesh and screen is one way to shield the amp, and can be found in arts and crafts stores. I used Scotch spray adhesive #77 to attach the mesh to the chassis (*Photo 4*), and several screws with washers to ensure good electrical contact.

Over the vents I used copper screen (*Photo 5*) doubled over for strength, and attached with socket cap screws in six locations. Be careful with both types of shielding, making sure no small threads or parts of the mesh touch the PCB, connectors, or short any part of the amp.

The two small chokes on the power coming into the amp are simple common-mode chokes (*Fig. 2*). You can make a simple version by twisting +V and Gnd, –V and Gnd, and slipping a ferrite ring over each twisted pair. These, with the shielding, will help keep RF and EMI at bay. This may all be overkill, but a battery alone is not the end-all solution to power-supply ills.

POWER SUPPLY CHARGER

I chose to go with a battery PSU, but Erno makes a good discrete regulated AC supply, and Audiokits.com has a nice case that it will fit in. For the battery PS, I purchased some new surplus Panasonic 12V 4.5ah lead acid cells, and the intelligent 12V charger (Optimate-*Photo 6*).

If you decide to build your own charger, I recommend one based on an hysteretic algorithm. These work well, won't overcharge, and will extend the

PHOTO 6: Internal view of PSU with charger.

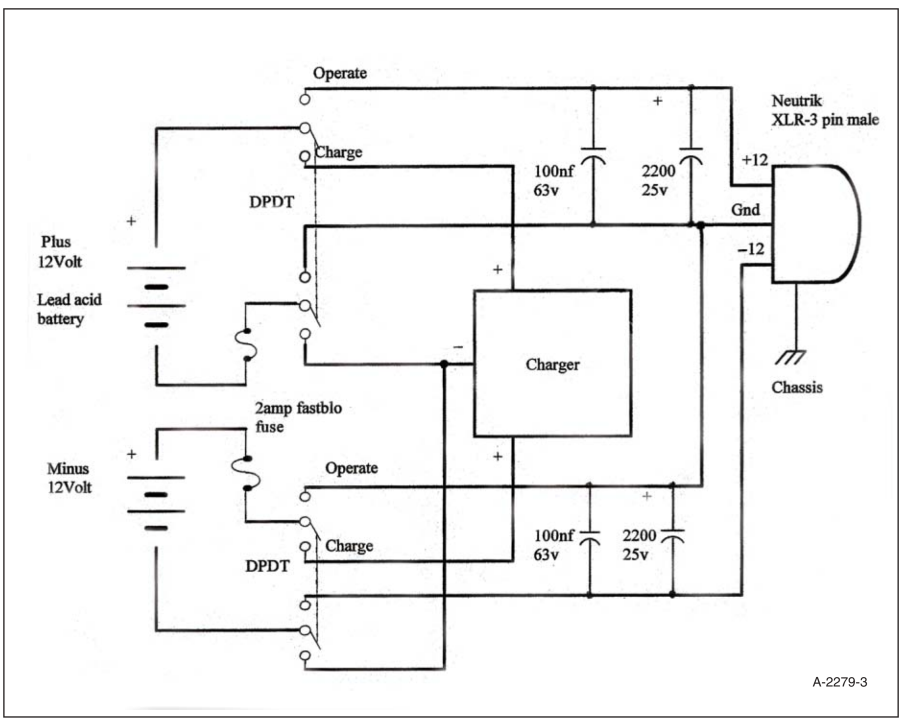


FIGURE 3: Common-mode choke diagram.

life of your batteries. I don't have room to go into chargers here, but many sites exist, and an article in *Nuts & Volts* April '03 has a good charger design. Altex.com carries Optimate chargers and batteries. I considered building a charger, but cost was more than the Optimate.

PARTS LIST

Headphone Amps	
Resistors, Trimpots	
R1, R7, R12	100R
R2	1MEG
R3, R4	499R
R5, R6	75R
R8, R11	10k
R9	1k1
R10	33R
	7R5
R13	6R8
	3R9
R14	47R5
P1	50R
All resistors Vishay-Dale RN60, unless otherwise noted.	
Capacitors	
C1	100pF, 160V
C2, C3	0.01µF, 100V
C4	10pF, 160V
C5	1000pF, 160V
C6, C7	220µF, 35V
C8	100µF, 25V
Semiconductors	
Q1	2SK389BL/V
Q2	2SA872
Q3	2SJ79 INS
Q4	2SK216 INS
Q5	2SC1775
D1	J508
D2	1N4148
Miscellaneous	
PCB (80 × 80mm)	EB-602/210
HS1, HS2	SK76-37.5
12 × 1mm solder pins	
4 × M3 screws, nuts	
1 × 10k attenuator-stereo	
1 × Neutrik headphone jack	Mouser#568-NJ3FP6C
2 × RCA female chassis mount	
1 × DPDT switch panel mount	
2 × Ferrite toroids ¼″ diameter	
2ft^2 copper mesh/screen	
Battery PSU	
Batteries	
2 × 12V 2.2ah (min) lead acid	Altex.com # GP1222
Charger	
Optimate 12V charger or equivalent	Altex.com # LS12/1.2
Capacitors	
2 × 2200µF, 25V electrolytic	Digikey.com
2 × 100nF, 63V polypropylene film	
Switches	
2 × DPDT (ON-OFF-ON)	Digikey.com
1 × SPDT (120V AC front panel charger ON/OFF)	
Connectors	
1 × male Neutrik XLR 3-pin chassis mount	Mouser.com
1 × female Neutrik XLR 3-pin chassis mount	Mouser.com
Power Umbilical	
1 × male Neutrik XLR 3-pin cable mount	Mouser.com
1 × female Neutrik XLR 3-pin cable mount	Mouser.com
15′ 16GA copper cable (PVC insulated)	
5′ copper braid ¼″ sleeve (shield)	Michael Percy Audio
6′ expandable ½″ plastic or cloth sleeve	Michael Percy Audio
Miscellaneous	
2 × automotive fuses with holders (12V-2A)	
4 × ¼″ standoffs	
1 × 3″ × 4″ proto-board	
1 × case (chassis for PSU batteries/charger)	
Double-sided foam tape to mount batteries/charger	
Heavy-duty cable ties to secure batteries/charger	

BATTERY POWER SUPPLY

CAUTION: Lead acid batteries can explode if connected in parallel, or shorted, so they must be fused (*Fig. 3*). I derived the power-supply requirements by taking the total current drawn by the load (each channel draws about 400mA, $2 \times 400 = 800$mA). Maximum current draw is less than 1A. Realistically, 1.2ah batteries should work, but you should charge them every (approximately) eight hours if you want the full voltage driving your amp (ideal for audio). Alternately, a 2.2ah would be the best choice, needing a charge every 16 hours (worst case).

Since I already had the 4.5ah cells, I figured they would do fine. While a battery's DC resistance is low, its impedance is relatively high, and more so at high frequencies. For a battery to deliver full energy for short high-frequency energy bursts, you need to parallel it with an electrolytic bypassed with a polyproylene, or polystyrene film cap. A rule of thumb for this is 1000µF per amp of load current. I double that for audio, which gives me about 2200µF (*Fig. 3*).

In this PSU (*Photo 6*), I actually used several WIMA 1µF caps, and one 100nF polystyrene across the 2200µF mounted on a proto-board. PSU (*Fig. 3*) shows only one 100nF, which is adequate. By removing the battery's burden, heating is also reduced, thus extending battery life.

Automotive fuses (2A) are used to protect the batteries from being shorted together. One battery is charged at a time, but you could build a dual charger to simultaneously charge both. Take note of the polarity of the batteries; −12V polarity is reversed in the Operate switch position (+ to Gnd), but is the same as +12V in the Charge position.

As an option, a connector is mounted on the back of the PSU, allowing external cables to be connected to the charger (*Photo 7*). Other batteries including car and mower batteries can be charged with a flip of the power switch on the PSU front panel. The DPDT switches are of the ON-OFF-ON type, so left in the center position (OFF) the batteries disconnect (draw no current), and the external charger cables can be used.

PULLING IT ALL TOGETHER

After verifying voltages out of the PSU and HP amp bias current (162mA) through R13, I adjusted out any DC offset (P1) using a bench PS and a millivolt meter. Per the instructions, I checked for oscillations, and verified output gain using an HP 206A sine wave generator and a TEK 465B scope.

At 1V RMS the THD measured 0.005% with a 50Ω load, which agrees with published specs. Because XLR connectors were used, I was able to use a microphone cable initially as a power umbilical. With no input, and the volume at full, I noticed a slight hum—most likely AC from house wires picked up by the microphone cable, or a ground loop.

To correct this, I made a 5′ 3″ conductor cable and used a separate outer shield (¼″ copper braid sleeve) and connected this to chassis GND on both ends. The Neutrik XLR has a separate connection for chassis GND, and needs to make good contact to the mounting hole.

The three wires are connected to +12V, −12V, and PS Gnd. This completely eliminated any hum, and was now so quiet, with volume at full, I needed to check several times to verify it was on. It was on, and now with the DAC 3 as an input fed by a Mitsubishi DD-5000 DVD player and a GW labs DSP upsampler, I could finally hear the result.

After a two-day break-in, the sound was very involving with jazz such as Weather Report, or classical (Debussy's Clare de'Lune) in which the piano sounded alive without harsh overtones (as it does on other amps), and DVD movies ("Stargate," "Jurassic Park"). It was a revelation. I was hearing such details as the plucking of the string bass and the drummer quietly setting down a pair of drumsticks midpassage. I had never noticed these with any other HP amp or speaker. Using it now for mixing, composing, and editing, my ears don't fatigue as quickly, and I get more done.

When I used an AC regulated supply (Linear Tech. LT317), the sound became harder, with more edge, and lost some of the smoothness in the high frequencies. Low end was always very tight and extended with both supplies. I must conclude that the battery PSU makes an audible difference and was well worth the time and cost. Final touches on the amp included a Neutrik HP jack with silver-plated contacts, and the locking mechanism that keeps me from pulling out the plug. I prefer a stepped attenuator over a pot, and the DACT 10k is reasonably priced.

I also wanted a knob that wasn't too large, and found a solid stainless-steel one made for cabinets at Home Depot Expo for a few dollars. I had to drill and tap it, but it feels great and is just the right size for this amp. Thanks to Ed Dell and his inspiring editorial on making aesthetics a crucial part of any project, this old 16-bit DAC never looked, or sounded, better. ❖

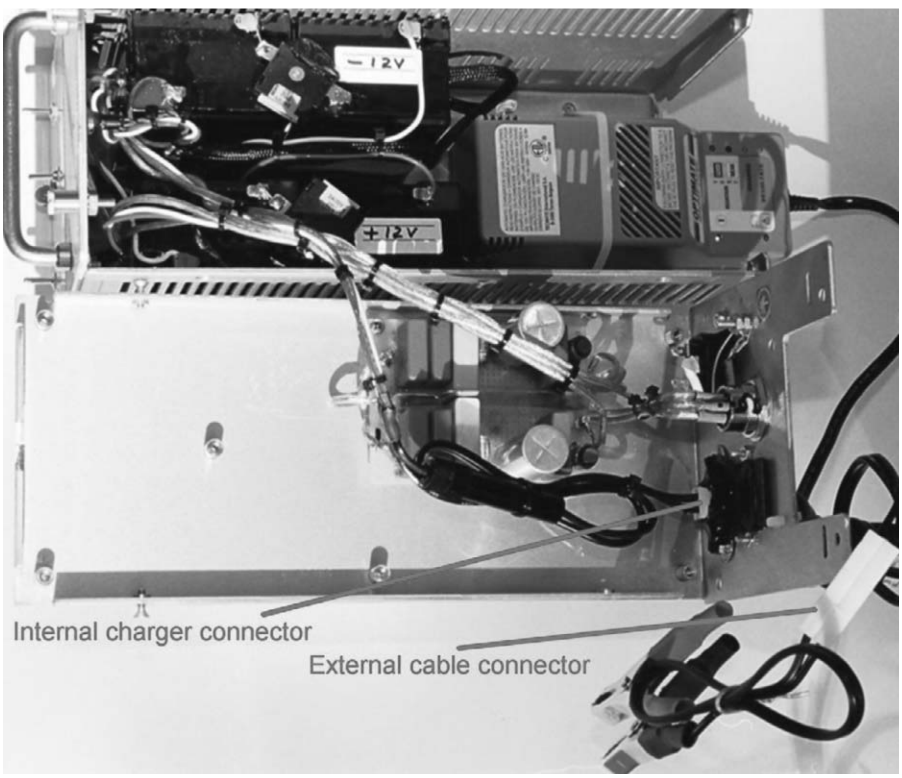


PHOTO 7: PSU with optional external (charger) cable.

SOURCES

Borbelyaudio.com—HPamp PCBs, kits, parts, cases, Cerafine caps, tantalum resistors, silver mica caps, FETS

Audiokits.com—HPamp PCBs, Borbely kits, audio parts, cases for HPamp, FETS, MOSFETS, DACT attenuators

Percyaudio.com—Components, braided sleeve, Caddock resistors, MUSE caps, silver wire, extensive audio parts

Altex.com—Batteries, chargers, electronics, computer parts

Mouser.com—Components, Neutrik jacks and XLR connectors, RCA jacks, more

bgmicro.com—Surplus heatsinks, batteries, surplus parts, much more

aloha-audio.com—DACT step attenuators, audio components, kits

DIYHIFISupply.com—DACT attenuators, audio components, Kiwame resistors, kits

High Power SE 6C33C Amp

This article deals with a single-ended amplifier with enough power to drive even the most difficult speaker systems. **By Ari Polisois**

PHOTO 1: A view of the 50W-SE-6C33C-B Amp with the left and right driver sections in the front row, and their input sockets next to the 6SN7GTs.

In my opinion, single-ended amplifiers are simpler, do not require frequent adjustments due to valve characteristics' drifts, and cost less than push-pull versions. Also, I am convinced that they are more sensitive, carefully amplifying the quietest sounds. There is another good reason: the zero level of the amplification is normally set halfway between the straight portion of the plate curves, letting the anode current changes take place smoothly, up and down. The electrical consequences are that if any harmonic is generated, it will be of the even family, meaning that the added "la" will still be a "la," the "sol" will still be a "sol," and so on.

In other words, these harmonics are "harmonious," unlike the odd harmonics generated by the push-pull. Do not underestimate your ears' judgment. This sense is one of the most advanced that Mother Nature has granted us, detecting micrometric details that you would not believe existed. Odd harmonics are disturbing, even if in almost unmeasurable quantities. As a result, you become tired of listening very soon.

I received several e-mails asking for a powerful single-ended amplifier. One of these correspondents owned some glorious speaker enclosures that are hard to drive (such as the AR-pi), with sensitivity around 80–85dB. Thus far he had been using a push-pull amplifier, but I think I convinced him that the single-ended is "something different." I decided to use three 6C33CB in parallel, per channel, giving approximately 50W RMS output.

THE DESIGN

It took me some time to discover a suitable output transformer capable of handling the resulting quiescent current of 750mA (ten times the current required by a 300B in a single-ended configuration!) and, once this big obstacle was removed, I started designing the amp around this special OPT[1].

Figure 1 represents the schematics of the amplifier sections and their power supplies. From an electrical point of view, these power supplies are "stacked," that is, the final stage's is "on the top" of the other[2]. These stacked power supplies meet at one point: the negative of the upper one (power tubes) touches the positive of the lower one (driver section).

This is a must and strictly means that you must not connect the negative of the upper one to the chassis ground; otherwise, a short circuit will result, blowing the fuse(s). Keep this in mind, and everything else is simple.

In order not to have a tremendously heavy single cabinet, I placed the driver unit with its dedicated power supply and the output stage on the main chassis, and connected the power supply for the 6C33C-B—in a separate box—with a twin wire to the main unit.

The first triode (half of the 6SN7GT) amplifies the signal at the input and feeds it to the second half without any blocking capacitor. This system is widely used, most notably in the famous Williamson amplifier.

I call your attention to the heart of the system, the anode load of the driver section of the 6SN7GT, namely, the resistor R6 (*Fig. 1*). This resistor feeds the 6C33 triplet in two ways: it supplies them the bias required (about 85V—negative to grid, positive to cathode) as well as an AC signal of adequate swing (80 to 85V peak). To get this amplitude, you need about 1V RMS at the input, the amplification being 60–65 times.

POWER STAGE

While all anodes are directly connected together and to the primary of the OPT, the grids get the signal through small resistors (anything between 220Ω to 1k ¼W would do), in order to be somewhat separated (*Fig. 1*). As far as the cathodes of the 6C33s are concerned, I needed to fit a variable resistance network to "align" them, due to unavoidable differences in the valves' characteristics. The six 50Ω potentiometers, shunting the 100Ω fixed resistors, adjust the resulting resistance from 33Ω to zero.

At its maximum level (33Ω) there is a drop of about 8V that represents an individual bias for each 6C33, in addition to the common negative voltage supplied by R6. It may be necessary to have, for instance, one pot set halfway, the second fully turned clockwise, and the third in between, in order to adjust the quiescent currents of the three 6C33s to almost the same level. (I would like to add, however, that, in practice, I did not notice a big change in the sound with as much as 20–30mA difference between the valves)[3].

The 6C33C-B is a wonderful valve: Sturdy (designed for the Soviet Migs and tanks), it has such an unbelievably beautiful low internal resistance (80–100Ω); which makes a difference during bass amplification. As you know, the reactance of the primary coil tends to drop at low frequencies, and, if the internal resistance of the valve is high, most of the power is kept inside the valve and not transferred (as in a voltage divider attenuator)[4]. This is all you need to understand about the final stage layout, which is otherwise quite conventional.

There is nothing really special in the power supplies. *Figure 1* shows this portion of the amp also. I used solid-state diodes because they are handier, but managed to shunt them with small capacity, high-voltage capacitors to reduce the switching peaks. The transformer is a normal line insulation step-up type. Its rating should not be less than 450VA.

Figure 2 shows the frequency range (which is in excess of most available speaker systems); it is measured at 32W output. *Figure 3* shows the distortion/power relationship. This is a very satisfactory result, considering that there is no overall feedback. Because my usual listening level does not exceed 10W (due to a relatively modest room), I enjoy a rate of about 1% distortion restricted to just the second harmonic (I found no trace of odd harmonics with the HP spectrum analyzer).

In *Photo 1*, the protruding knobs are the bias controls for the first section of the drivers, which also determine the common bias to the power tubes, and therefore set their average idle plate current. The trimmers that take care of a finer individual alignment of the 6C33s are next to these valves, in the back.

The white octal socket, in the middle, contains six test points that measure

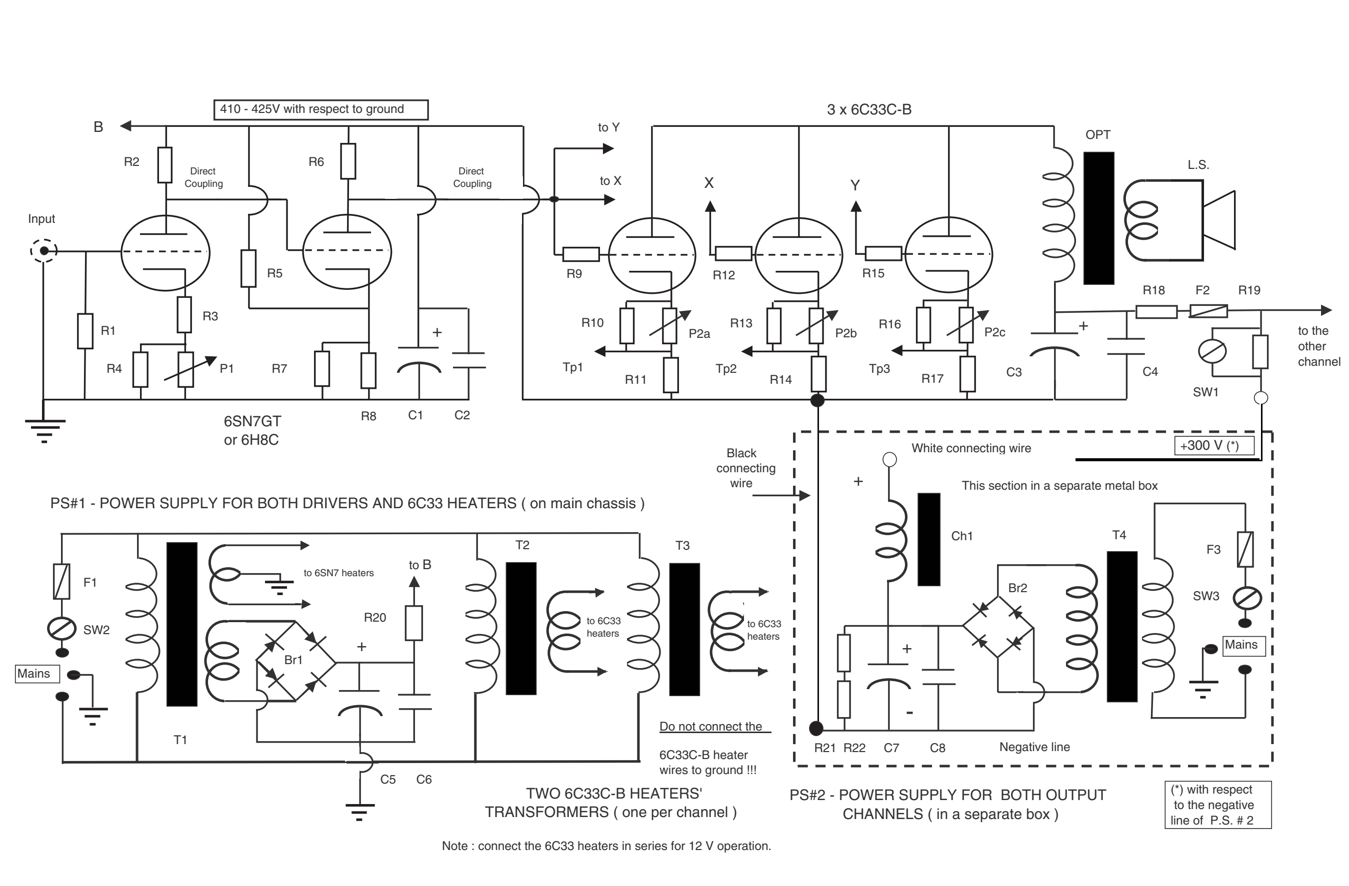


FIGURE 1: The 50W-SE 6C33C-B amplifier.

the 6C33s plate currents, using an external portable multimeter; however, a removable box comprising a selector and a meter can be plugged into this socket, to ensure easy monitoring of the anode currents.

The huge boxes in the back enclose the powerful output transformers. The connections for the loudspeakers and to the power stage are on the hidden side of the boxes. I can remove or replace each output transformer in less than a minute, including the connection, by means of safety plugs and sockets.

HINTS

1. Use generous-size chassis to accommodate this design.
2. The 6C33s generate a lot of heat, so do not place the capacitors close to them, but possibly underneath.
3. The two OPTs weigh 30 lb each! My solution was to make them easily removable with safety connectors (the same used with some portable multimeters) of different colors (*Photo 3*), so that taking them away or putting them back and connecting the wires takes less than one minute. Do the same and you won't regret it.
4. The power supply for the output valves is split into two parts:

   a)The first comprises the mains (115 or 230V) to 210–215V transformer, plus the rectifier bridge, the first filtering capacitor, and the 1200mA/5H choke—all enclosed in a separate, well-ventilated box that I put on the floor. The connection to the main chassis is ensured by a twin wire cable with safety connectors (one white—positive—and the other black—negative). Approximate weight is 40 lb (18kgs).

   b)The second part of the power supply is settled in the back of the main chassis and includes four big (1000μF/385V) capacitors, each pair being fed from the B+ of about 300V through a 10Ω/5W resistor, used as a line separator (*Fig. 1*).

   This part also includes two filament transformers (12.6V–10A each) or just one (12.6V–20A). Considering that these transformers might generate heat, the best location is on the top of the chassis. I discarded the idea of putting them with the 450VA tranny in the same separate box, because of the high current they must convey to the valve heaters.

   The removable OPTs solution helps keep the main cabinet's weight at a normal level for such an amplifier (36 lb.).
5. For the wiring, always use the same colors for the same functions. I suggest:
   - Red for the 300V B+ line
   - Brown for the negative of the 300V line (power tubes)
   - Brown, also, for the B+ of the 400–425V positive line of the driver's power supply
   - Blue for the connections to the plates
   - Green for the connections to the grids
   - Combined yellow/green colored wire for the connections to the main ground (metal chassis)
   - Twisted bifilar yellow wire for the heaters
   - White for any other unmentioned use.

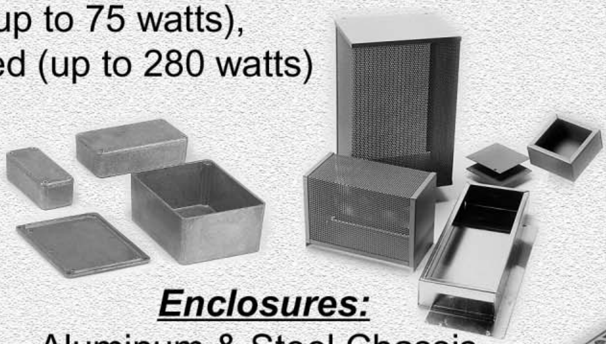
   Believe me, this color code will save you a lot of headaches and expensive errors. Once you memorize it, always

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™

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

use the same colors (no exceptions, please).

6. I strongly advise you to follow the recommended layout (including the values of the components, their rating, and their suggested location, particularly with regard to those with high heat dissipation) if you want to build this unit successfully. I have built three of these amplifiers so far (photos show my latest unit), and this is what my experience suggests.

OPERATING TIPS

Most of the heating elements will find

PARTS LIST OF THE 50W SE 6C33 AMP (ESTIMATED COST $2,000)

REFERENCE	VALUE	DESCRIPTION	QUANTITY	FUNCTION	SOURCE(#)
R1	220kΩ	¼W carbon	2	Grid resistor	A-B-C
R2	180kΩ	2W carbon	2	Anode load resistor	A-B-C
R3	470Ω	¼W metal oxide	2	Cathode bias resistor	A-B-C
R4	1kΩ	¼W metal oxide	2	Auxiliary cathode resistor	A-B-C
R5	10kΩ	50W w/heatsink	2	Voltage divider resistor—upper	A-B-C
R6	10kΩ	3W carbon	2	Anode load resistor and bias supply	A-B-C
R7	2,700Ω	3W carbon	2	Cathode bias resistor	A-B-C
R8	2,700Ω	3W carbon	2	Cathode bias resistor	A-B-C
R9	670Ω	¼W carbon	2	Anti-parassitics grid resistor	A-B-C
R10	100Ω	5W carbon or wire wound	2	Cathode resistor	A-B-C
R11	1Ω	2W carbon or wire wound	2	Test point for 6C33 plate current	A-B-C
R12	670Ω	¼W carbon	2	Anti-parasitics grid resistor	A-B-C
R13	100Ω	5W carbon or wire wound	2	Cathode resistor	A-B-C
R14	1Ω	2W carbon or wire wound	2	Test point for 6C33 plate current	A-B-C
R15	670Ω	¼W carbon	2	Anti-parasitics grid resistor	A-B-C
R16	100Ω	5W carbon or wire wound	2	Cathode resistor	A-B-C
R17	1Ω	2W carbon or wire wound	2	Test point for 6C33 plate current	A-B-C
R18	10Ω	5W carbon	2	Filtering and separating resistor	A-B-C
R19	4,700Ω	50W w/heatsink	1	Standby plate current resistor	A-B-C
R20	220-470Ω	25W w/heatsink	1	Filtering resistor (choose value for 415V)	A-B-C
R21	33kΩ	7W wire wound	1	HI-Voltage bleeder resistor	A-B-C
R22	33kΩ	7W wire wound	1	HI-Voltage bleeder resistor	A-B-C
P1	470Ω	2W linear potentiometer	2	To adjust bias of the 6SN7 and 6C33s	A-B-C
P2a-b-c	50Ω	5W wire wound	6	To adjust bias of the 6C33s	A-B-C
C1	330-470µF	450-500V electrolytic cap	2	Driver PS filtering cap	A-B-C
C2	470nF	1000V-MKT-capacitor	2	Additional cap to above	A-B-C
C3	2000µF	385V-electrolytic cap	2	Output stage PS filtering cap	A-B-C
C4	470nF	1000V-MKT-capacitor	2	Additional cap to above	A-B-C
C5	100µF	500V electrolytic capacitor	1	Driver's PS reservoir capacitor	A-B-C
C6	470nF	1000V-MKT-capacitor	1	Additional cap to above	A-B-C
C7	220µF	385V-electrolytic cap	1	Power output stage PS reservoir cap.	A-B-C
C8	470nF	1000V-MKT-capacitor	1	Additional cap to above	A-B-C
Br1	4×1N4007 diodes	Shunted with four 4700pF 1500V			A-B-C
		snubber capacitors	4	Rectifying bridge set-drivers stage	A-B-C
Br2	4×1N5408 diodes	1200V piv-3A, shunted w/four			A-B-C
	or BY255 diodes	4700pF/1500V caps.	4	Rectifying bridge set-Output stage	A-B-C
6SN7GT	Triode valves	Sovtek or GE or Sylvania	2	Voltage amplification	A-B-C
or 6H8C	Triode valves				A-B-C
Sockets		Octal sockets for 6SN7	2		A-B-C
6C33C-B	Triode valves	Sovtek	6	Power amplification	A-B-C
T1	Power transf.	Mains to 345V-150mA plus			D-E-F
		6V-1.5A with c.t.	1	Driver's PS and heaters power transf.	D-E
T2	Heaters transf.	Mains to 12V-10A	2	6C33s heaters' transformer	D-E
T3	Heaters transf.	Mains to 12V-10A	2	6C33s heaters' transformer	D-E
T4	Power transf.	Mains to 210-215V-1.2A	1	HI Voltage power transformer for Br2	D-E
OPT	Output transf.	Z=200Ω-on 8Ω			F-G
		load-power = 50W min.			
		Idle current = 800mA min.	2	Valves to Loudspeaker interfacing	
Ch1	Filter choke	1200mA DC-minimum	1	Filtering	F-G
SW1	Standby switch	SPST 250V-3A or 125V-6A	1	Standby	A-B-C
SW2	Mains switch	DPST 250V-3A or 125V-6A	1	Warm up switch	A-B-C
F1	Fuse fast	4A for 230V or 8A for 115V	1	Drivers' PS and heaters transf. protection	A-B-C
Bulgin	Integrated socket	Alternative to SW2	(1)		A-B-C
		and F2 fuse holder	1		A-B-C
F2	Fuse slow	3A-250V	1	6C33s valves protection against overload	A-B-C
SW3	Mains switch	DPST 250V-3A or 125V-6A	1	Main switch	A-B-C
F3	Fuse fast	5A for 230V or 10A for 115V	1	6C33s PS transformer protection	A-B-C
Bulgin	Integrated	Alternative to SW3 and F3	(1)		A-B-C
	socket	fuse holder	1		
Miscellaneous	RCA jacks, LS jacks, cabinet, cords, wires,				A-B-C
screws and nuts, and so forth					

(#) A-B-C = Tubesandmore.com, partsconneXion.com, RadioShack.com
D-E = plitron.com, hammondmfg.com F = Bob@klinbergconsulting.com
G = A2Belectronic@AOL.com

their better place outside the cabinet, as shown. As recommended, do not attempt to change the values of the components. Particularly critical are P1 and R3. Depending on the 6SN7GT characteristics, you may need to use a slightly higher or lower value for R3 in order to build—across R6—the correct amount of bias for the 6C33C-B.

Once you have adjusted the plate current levels of the 6C33s (up to and including the warm-up time of about 15 minutes) by operating first on P1 and then on the individual 6C33C-Bs pots, your amp will be quite steady. However, if you choose to constantly witness its behavior, you can connect a 0.5 or a 1V fs meter to a six-position selector to measure the voltage across the 1Ω resistor test points. The reading in volts

Frequency kHz	Output Volts % (1 kHz=100%)
0.015	50.0
0.020	65.6
0.030	78.0
0.040	85.5
0.050	90.6
0.060	93.8
0.080	96.5
0.100	98.3
0.200	99.5
0.500	100.0
1	100.0
2	100.0
3	100.0
5	100.0
10	99.8
15	95.0
20	88.0
25	78.0
30	67.0
35	56.0

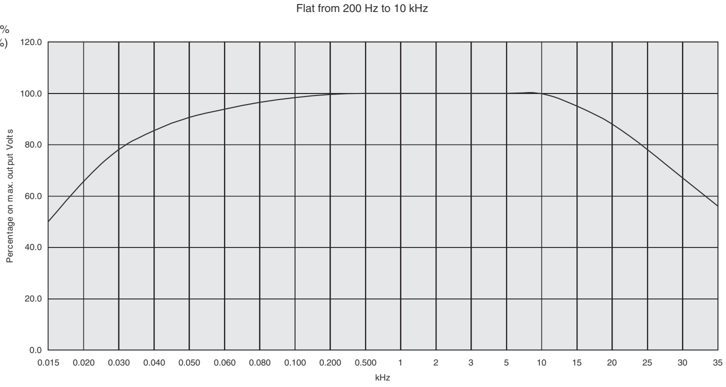


FIGURE 2: Frequency range. GA-2303-2

across these resistors indicates the milliamps (example: 200mV = 200mA) of plate current[5].

Regarding the sound, every person who listened to the amp found it astonishing, but I would like to quote especially a sound engineer who is also a teacher of sound recording techniques in Turin, Italy: "Allow me to express my congratulations for the quality of the sound that your amplifier can reproduce; I have been able to enjoy the works that I know by heart *[we had been listening to some of his recordings on CDs]* and discovered such a living dimension very close to the one I received when I was listening to the musical instruments before placing the microphones."

PHOTO 2: The special OPT used with the three paralleled 6C33s, whose total idle current is 700-750mA. It's a gapless transformer, self-compensating at low flux. *Figure 3* confirms that this OPT can handle the 50W output without excessive distortion, and *Fig. 2* shows that the frequency range is quite extended.

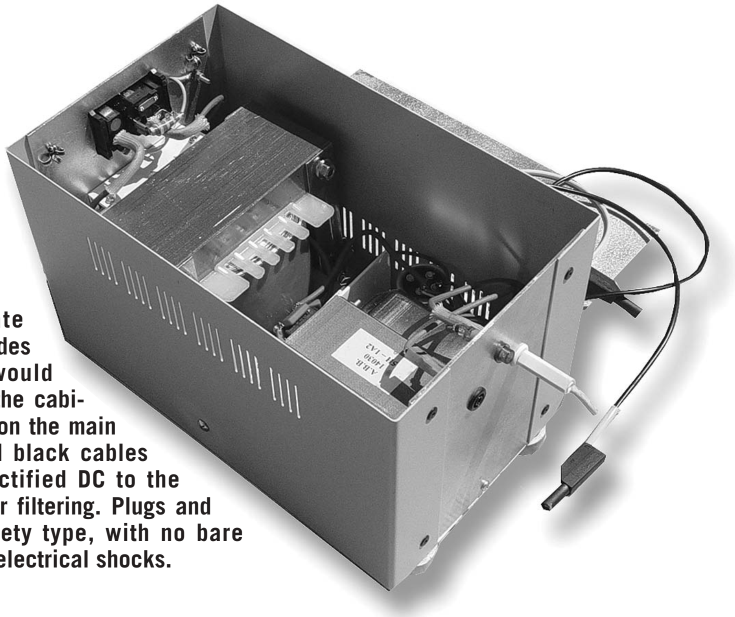


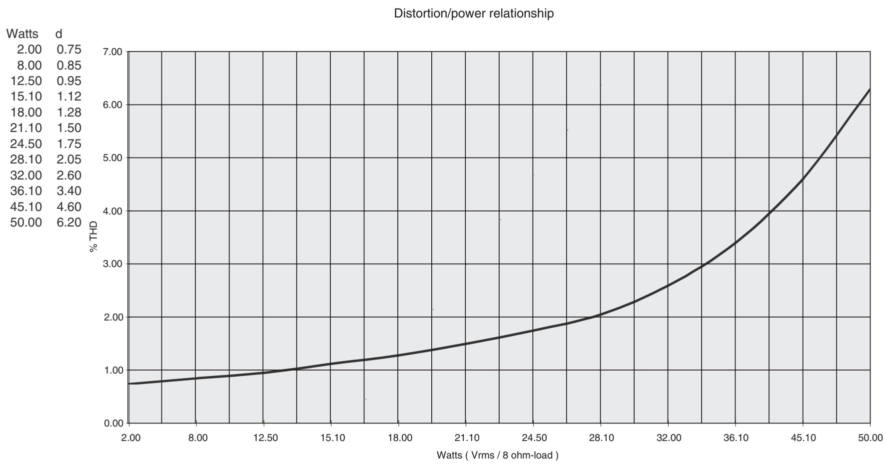
PHOTO 3: The separate power supply box includes the heavy parts that would tremendously increase the cabinet's weight, if located on the main chassis. The white and black cables bring the pulsating rectified DC to the main chassis, for further filtering. Plugs and sockets are of the safety type, with no bare points that could cause electrical shocks.

Bass is strong, damped, and dry. Not surprisingly, the three 6C33s in parallel mean a resulting plate resistance of about 90/3 = 30Ω, which is not too far from a solid-state device but with the advantage of the valve sound.

In sum, please note:

a) This amplifier is not for beginners, unless you take time to understand everything, not to rush, check and recheck what you have assembled, and so forth. Needless to say, you must have some electrical background know-how.

b) As an experienced builder, you know that some of the voltages are lethal. Never forget that. If you are a beginner, this warning is of primary importance. ❖



N.B. - Distortion was measured with a HP 334A at the frequency of 1 kHz. At the same power level of 32 W it shows 3,4 % at 100 Hz and 2,3 % at 10 kHz.

FIGURE 3: Distortion.

GA-2303-3

REFERENCES

1. This OPT is available at the following address: A2BElectronic@aol.com or Bob@klingbergconsulting.com under the reference OPT-OPUS-70-200 or King III.

2. The system is fully described in Plitron's site—quoting an article published by *Glass Audio* (17 novel audio amplifier circuits), available from Antique Electronics.

3. The circuit is based on the 6C33C-Bs working with a plate current between 180 and 240mA. Immediately after switching the high voltage on, the current is 100–120mA, but within 10 minutes it settles to 200–220mA, which is an excellent working condition, and it does not move from this level.

4. As an example, consider a 3H impedance primary under the frequency of 20Hz. According to the formula $2 \times 3.14 \times f \times L = Z$ (Ω), its reactance would be 377Ω. With the plate resistance for one 6C33C-B at approximately 100Ω, the primary would get a share of 377/(100 + 377), or 79% of the signal amplitude. This is better than 70%, which corresponds to the loss of 3dB. With the three paralleled valves, the result has definitely improved!

5. The sequence to operate the amp is:

a. Make sure the standby switch is off (open).

b.Turn the main chassis switch on: this will enable the driver to operate and supply the bias required by the 6C33C-B in advance of the plate voltage. Also, the heaters of the 6C33s will start warming.

c.Turn the separate power supply box switch on.

d.About one minute later, turn the standby switch on (closed).

To shut down the amplifier, reverse these steps.

A Dipole Midbass, Part 2

This month we conclude construction of these dipole speakers, install the electronics, and test the system. **By Tom Perazella**

I connected the three AudioSource amps to the speakers using 12-gauge zip cord. Drive for the amps was provided by a custom analog crossover that fed frequencies from 60Hz to 250Hz to them. The crossover was a Butterworth design with 12dB/octave slopes. Basic level control to the amps was accomplished in the crossover. EQ was provided by a Behringer DSP8024 Ultra-Curve Pro. I used this setup for several months while waiting for two new devices from Behringer, the DEQ2496 Ultra-Curve pro equalizer and the DCX2496 Ultra-Drive Pro speaker control system. More about those later.

The author's completed system.

Initially, it was very apparent that the dynamic range of these new speakers was much greater than the original single 12″ driver that was used for this range. In addition, the frequency balance as I moved around the room seemed more stable than before. However, as usual when starting such a large project from scratch, a major problem raised its ugly head as I ran some sine-wave frequency tests.

As I ran the sweeps, there was an obtrusive, very audible resonance around 190Hz. It was quite narrow in frequency, indicating that it was high Q. I ran my hands across all parts of the speaker to find the resonance and found the source. Believe it or not, the two diagonal braces from the front frame to the legs were vibrating like guitar strings. I could put my hands on the braces and the resonance went away.

My first thought was to use sand in the tubes to damp that resonance. I was not too hopeful of that approach because when I had done that in the past, it had primarily added mass and lowered the frequency and Q of the resonance, rather than eliminating it. However, since I had originally planned on filling the front rails with sand, it was not a big stretch to drill holes in the tops of the braces and fill them with sand. As I expected, that had little effect on the problem.

The best solution in cases like this is to divide the vibrating member into smaller sections whose resonant frequencies would be beyond the exciting frequencies being reproduced by the drivers. To achieve this, I fabricated some additional braces out of 1″ square pieces of wood to be anchored between the legs and diagonal braces. *Photo 29* shows one of those braces before painting.

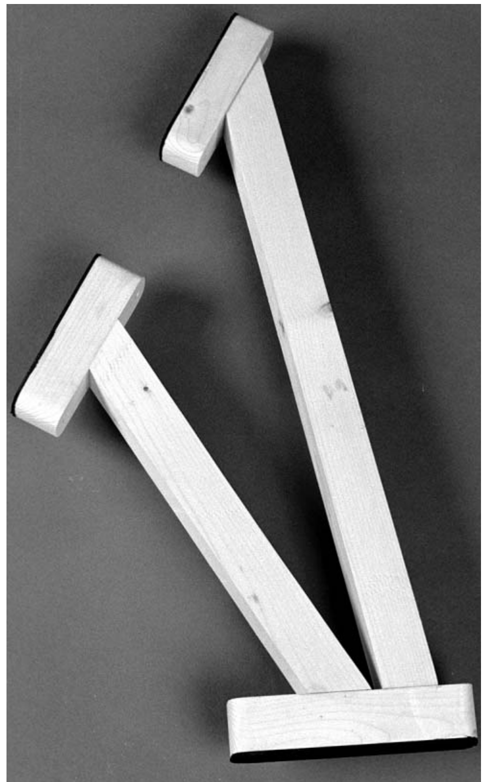


PHOTO 29: Leg brace before painting.

I attached the base of the V to the foot and attached the tops to the diagonal brace, dividing it into three unequal lengths. I produced an additional X shaped cross brace for each speaker to connect between the two new V braces for additional lateral stability. I vacuumed the sand out of the diagonal braces, drilled mounting holes for the new wooden braces in both the feet and diagonal braces, and then installed the braces.

I re-painted the diagonal braces to clean up the scratches that occurred while installing the new braces, and—just for good measure—re-filled the diagonal braces with sand. I repeated the sweeps, and the resonance was gone. My sigh of relief could be heard all over town.

If I were to repeat this project, I would have those braces made of the same box section steel and welded into place during the initial construction of the frame. That would be much easier than having to add the braces later. *Photos 30* and *31* are two views of the new braces.

To make the grilles as simple as possible and not contribute to the overall sound, I decided to make a very thin frame to hold the grille cloth. Structural strength would not be provided by the frame, rather it was a carrier for the fabric until you could place the frame into the space produced by the rabbet in the front plate and the bottom, top, and edge pieces.

I cut four pieces of wood, ⅞″ square and long enough to make a frame to fit in those grooves, and glued them together. To make it easier to slip the frames over the rabbet, I slightly beveled the leading edges of the inside of the frame, then primed and painted them black. I cut the fabric to size so that it would fit over the grille frames and extend three-quarters of the way around. No fabric would extend into the inside of the frames.

To hold the grilles in place, I installed Velcro® pieces on the back of the rabbet on the front plates and the

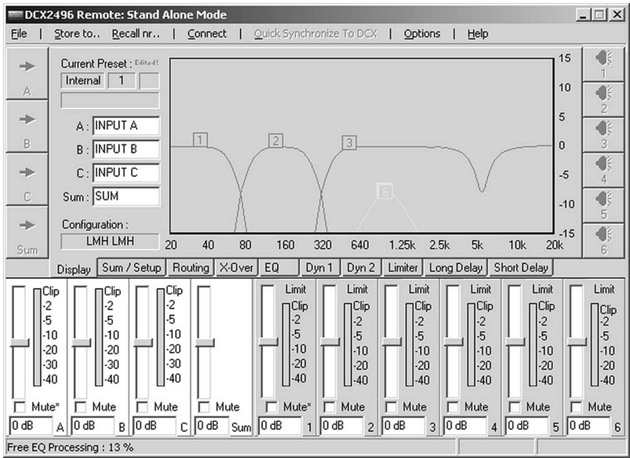


FIGURE 10: DCX2496 display screen.

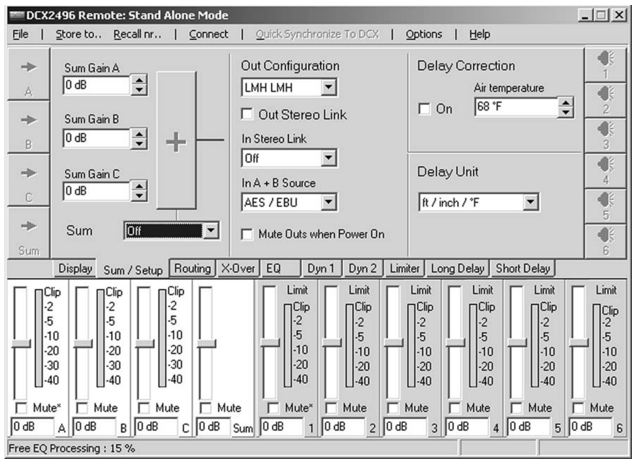


FIGURE 11: DCX2496 setup screen.

backsides of the frames. To prevent rattling of the grille frame against the front plate, I glued thin strips of felt all along the inside of the rabbet in the front plate.

Before fastening the fabric to the frames, I measured the exact dimension of the front plate between, then cut pieces of wood to those lengths. I placed those pieces at the appropriate points inside the frame to prevent it from bending inward as the grille cloth was stretched taut while being fastened. After stapling the cloth to the frame, I removed the braces and pressed the grilles into place. It worked like a charm.

During the initial testing, I decided that the speakers really didn't need a baffle on one side, as the lower frequency being reproduced was only 60Hz. I therefore made another end piece as before, but with a different support as I used the angle iron bracket to fasten the support piece directly. This completed the construction phase of the project.

A BIT OF PHILOSOPHY

Before going any further on this project, I would like to express my view of where speaker design is headed. In the past, many decisions needed to be made about "voicing" the speaker that had to do with resonant frequencies in the box, driver positioning, crossover design, and so on. Most, if not all, of these decisions were implemented through the use of passive methods such as box and vent dimensions, offsets in driver mounting plates, convoluted crossover designs including pads and Zobel networks, and so on. At best they were good approximations but could be quite expensive and large.

Now, however, I see a different view of speaker design enabled by the huge amounts of digital processing available at extremely low prices. In my mind, it is now most important to have several key physical parameters and use digital processing to determine the final response of the speaker. The key parameters are:

1. Sufficient linear volume displacement from the driver or drivers to cover any desired range of sound pressure levels at any frequencies to be reproduced. Many people equate volume displacement requirements only with low bass, but insufficient linear volume displacement at high frequencies can be just as problematic. Although the volume requirements for a given SPL decrease with frequency, the driver area and linear stroke also decrease in midrange and high-frequency drivers. If you do not have enough linear volume displacement at any frequency to generate the SPL you need, you will cause high levels of distortion.
2. The ability to handle high levels of power without compression or damage.
3. A radiation pattern that minimizes room effects, which are very difficult to control once excited.
4. A consistent radiation pattern at all frequencies so that off-axis timbre changes are minimal.
5. Freedom from narrowband resonances within the desired operating range.

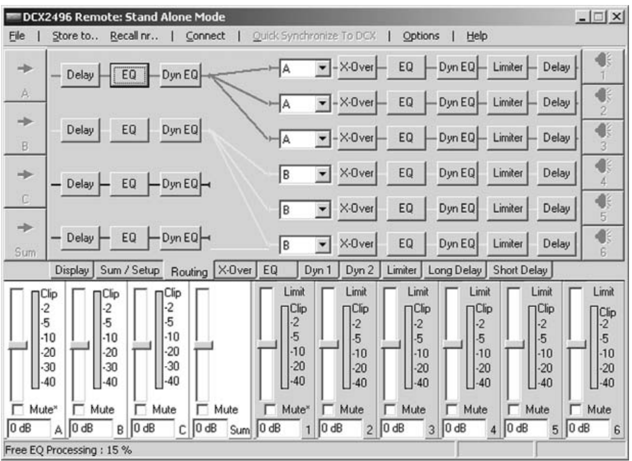
The rest of the speaker building process can be very effectively handled by some of the new digital control units now on the market. Other than the most basic, lowest priced speakers, I cannot see using passive components in a design. The basic advantages of bi- and tri-amping have long been known. The elimination of costly passive crossovers that are hard to tailor for the individual characteristics of production drivers, the elimination of wicked impedances produced by some passive crossovers, the lower IM distortion in amplifiers needing to cover only limited frequency bands, and the ability to dial in crossover points and levels are just a few of those advantages.

The new digital speaker control devices extend control capabilities to new levels. As I mentioned earlier, I was waiting for some new pieces from Behringer, just one of many companies now producing very powerful and cost-effective control devices. I chose them because I previously had good results with their products.

The DCX2496 is a good example of what can be achieved with this family of products. It can take balanced analog inputs and convert them to digital with sampling at 24 bits and rates up to 96kHz. Digital inputs can be used directly. After processing, signals can be output in balanced analog or digital formats. I won't go into great detail because there are too many features to list in this article. You can get the details at their website, www.behringer.com.

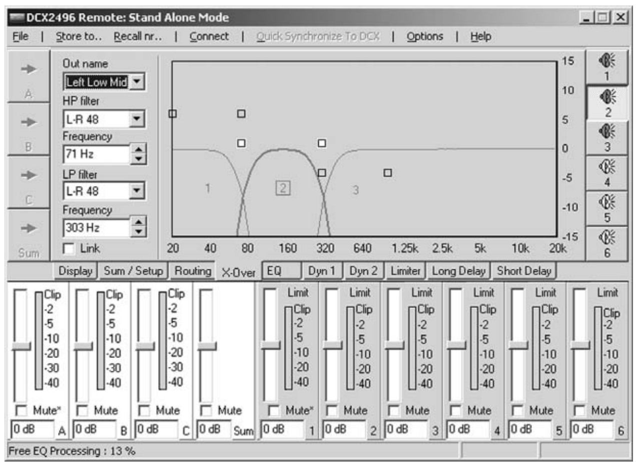
Briefly, the DCX2496 gives you three separate channels of input plus a summed input and six channels of output. You can assign any output to any input channels, providing huge flexibility as to how to use the device. Each input channel has the ability to use delay, EQ, or a dynamic EQ. Each output channel can provide crossover functions, EQ, dynamic EQ, limiting, and delays. You can implement the crossovers in Butterworth, Bessel, or Linkwitz-Riley configurations with slopes from 6dB/octave to 48dB/octave, depending on the configuration. You can accomplish any of these changes from a front-panel-operated menu, or from Behringer-supplied software through a serial connection to your PC.

The software allows you to use your mouse to change the crossover points or EQ center frequencies and Q on the fly. You save settings in memory on-board the device or through PCMCIA memory cards that fit into a slot in the front panel. For those of you using Compact Flash cards in your digital cameras, there are also adapters to put them into PCMCIA carriers and use them to store your Behringer settings.



B-2375-12

FIGURE 12: DCX2496 routing screen.



B-2375-13

FIGURE 13: DCX2496 crossover screen.

Using a device such as the DCX2496 allows you to tailor the drive signal to your individual drivers so that almost any output results, assuming that the key physical parameters of the drivers/enclosures previously mentioned are present. The speakers essentially become acoustic chameleons, changing their "colors" with a push of the button or click of the mouse. You can make these changes on the fly, while listening to the speakers or taking measurements. There is no easier or more precise way to control the output of a speaker.

CONTROLLING THE SPEAKER OUTPUT

Consider a few examples of setting up a speaker using the DCX2496. In this case, the screen images were generated by the Behringer software DCX2496 Remote, available free from the Behringer website. In fact, downloading this application along with the user's manual will give you a very good idea of the capabilities of this device. The display has several tabs, the first one showing you an overall view of your output results.

Figure 10 is a demo situation in which you can see the final results, and then the individual controls responsible for that result. This demo was done in the stand-alone mode with the PC not connected to the DCX2496, although direct input to the device is possible. For the sake of simplicity, only the left channel has been set up as desired.

In the center of the screen, you can see a graphic display of the output. It is divided into three bands for the sub, midbass, and high frequencies. The crossover slopes are 8th order L-R, and you can see a notch to control the cavity resonance of the BG RD75s. Note the level controls for all input and output channels plus muting. The configuration window shows that the six output channels are grouped into two sections of low, mid, and high.

The second tab as shown in *Fig. 11* is for setup. Key things to note are that the input and output stereo links are off, allowing individual controls on each channel. With the links on, any changes to one channel are reproduced in the other channel. The input source is digital in the form of an AES/EBU signal. The delay units are set to English distance instead of metric or time.

The third tab (*Fig. 12*) is the signal routing. In this case, the first input channel is routed to the first three output channels. The second input channel is routed to the last three output channels. You can also see the high degree of functionality available by looking at the functional blocks in each channel.

Now we're starting to get into the fun part. *Figure 13* shows the functionality in the next tab, crossover. The graphic display in the center shows the crossover functions for the three left output channels. However, since I've selected the tab on the right for speaker (channel) #2, it shows how the parameters were set up for that channel.

Note that I've chosen the name Left Low Mid for this channel. The highpass section of the crossover has been

PHOTO 30: End view of brace.

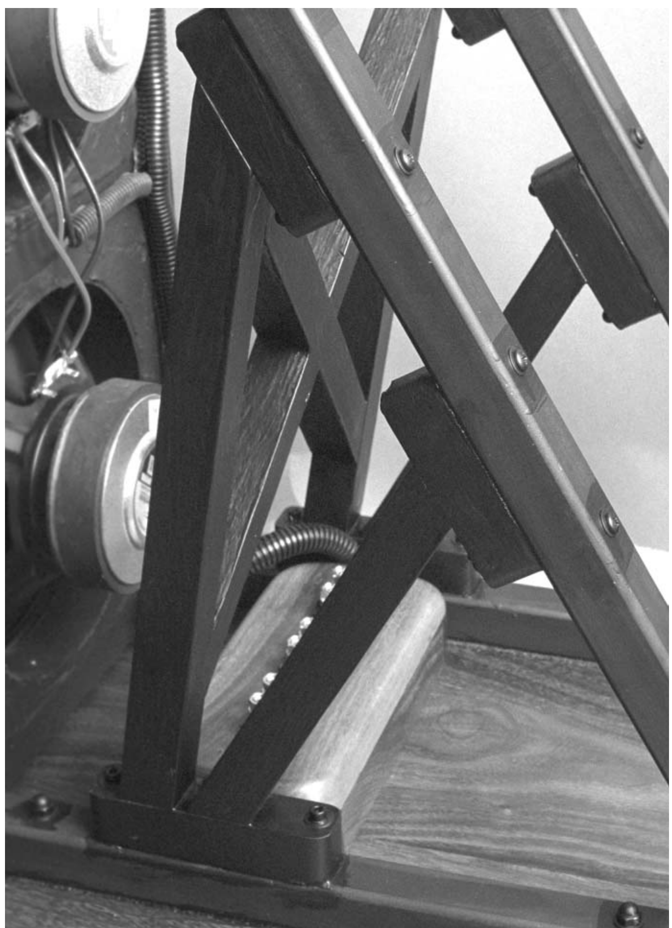


PHOTO 31: Side view of brace.

set as an 8th order L-R with a turnover point of 71Hz. The low-pass section has been set as an 8th order L-R with a turnover point of 303Hz. The left and right channels are not linked, although linking of each band independently is possible. Use the small white squares to drag the turnover points back and forth or enter the values directly.

The RD75s have a cavity resonance around 5.5kHz that must be accounted for. A passive trap is supplied with the drivers, but the DCX2496 can be used to compensate for that resonance without adding impedance in series with the driver. *Figure 14* shows the display of the EQ tab used to achieve this function.

The EQ function is set "on," the type is parametric, the gain is –8dB, the center frequency is 5.54kHz, and the Q is 5. By toggling the "ON" box, you can listen to or measure the effect of this function. Use the white box to drag the center frequency and adjust the amount of boost or cut.

The last function I'll show is time delay (*Fig. 15*). All the drivers are shown on one screen so that you can see their relationship to each other. You can choose to show either time or distance offsets. In this case, distance is shown. The distance offsets are shown graphically.

You pick a base speaker, in this case the subwoofer with no delay. If the mid had an acoustic center that was 10″ in

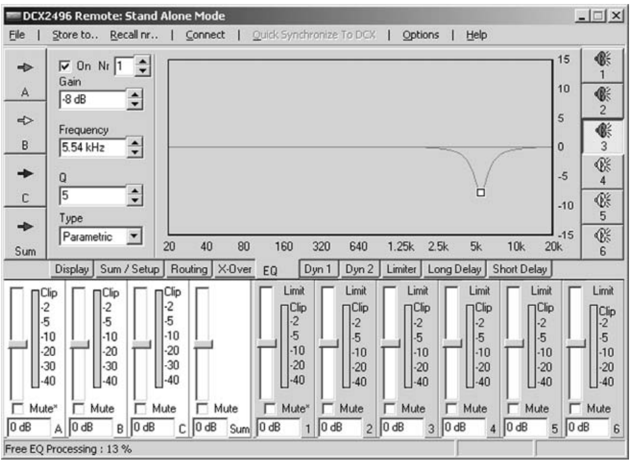


FIGURE 14: DCX2496 EQ screen.

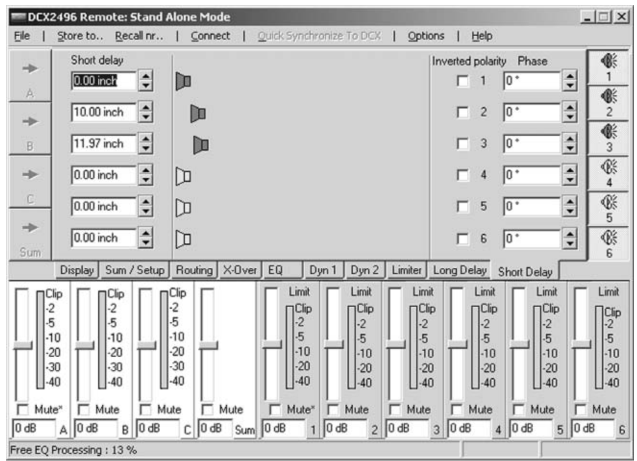


FIGURE 15: DCX2496 time delay screen.

front of the sub, you would add 10″ of "delay setback" to the driver to bring the apparent acoustic centers in line. If the high-frequency driver had its center 12″ in front of the sub, you would add 12″ (actually 11.97″ because the delay is based on time to the hundredths of milliseconds) of "delay setback." In addition to the delay, you could invert the polarity of the signal and add phase compensation.

ADDITIONAL PROCESSING

Once the basic control of the speakers was achieved using the DCX2496, I used the DEQ2496 equalizer for tuning in the room. This device has 31 bands of graphic equalization plus ten bands of true parametric EQ, plus dynamic EQ, and a feedback destroyer circuit that can locate a feedback frequency when microphones are used and automatically produce a narrowband parametric cut to eliminate the feedback.

One of the most interesting functions is the ability to change the bandwidth of the graphic sections from 1/3 octave to 59/3 octave. This is somewhat similar to the Q function of a parametric, but is available at each 1/3 octave frequency. Behringer calls that a Paragraphic function. The DEQ2496 was inserted in the signal chain after the preamp and before the DCX2496.

MEASUREMENTS

Since this speaker is part of a total system, I made measurements on the whole system, which consists of the subwoofers previously mentioned driven by a QSC USA1310 amplifier, the midbass speakers driven by the three AudioSource AMP 3s, and the RD75s driven by a Crown Macro Reference amplifier.

Taking traditional frequency-response measurements of the speakers themselves is not relevant in this case as the two Behringer devices really determine the final response. Because the combined drivers have so much linear volume displacement available, and there is more than adequate amplifier power available, there should be no difficulty adjusting frequency response. To confirm this, my friend, Alvin Foster, the founder of the Boston Audio Society, measured distortion for the whole system–the subs, the midbass, and the high frequency section. The tests were conducted using the software program ETF, available from www.etfacoustic.com.

To measure distortion, ETF feeds 16 discrete frequencies that are not harmonically related to the device under test. The frequencies are:

20.0
30.9
49.0
75.6
113.2
179.7
339.3
538.6
854.9
1161.2
1843.2
2926.0
5335.3
8469.3
13444.2
20000.0

Measurements were taken at a distance of ½m at a level of 96dB. This test is extremely difficult for any speaker and will result in high levels of IM distortion in all but the most capable devices. When tested this way, the total IM distortion was below 2% and the THD was below 0.4%. This was the lowest distortion Alvin had ever measured in any speaker he had ever tested, including some very expensive "high end" models.

THE SOUND

You have heard this many times before I'm sure, but the speakers sound pretty much like whatever you feed them. They are very transparent, they have a huge dynamic range, and the imaging is excellent. You can very easily evaluate the moods of the players, and the quality of the recording. They do not have an "analytical sound," which I equate with an overly bright presentation. As a matter of fact, you can make them sound however you like with the degree of control available from the Behringer units.

Michael Morgan is a friend who does really high quality recordings using DPA (formerly Bruel & Kjaer) microphones. You can find articles on some of his recordings on the website www.dpamicrophones.com under Users, Live PA, Michael Morgan. The July-August 2003 issue of *Surround Sound Pro* also had an article by him detailing the recording he did of Elmer Bernstein conducting the Florida West Coast Symphony in a program of some of his [Bernstein's] best-known film scores. Mike has mentioned on several occasions that he needs to hear his recordings on this system so that he can hear what's really on them. That's a pretty good testimonial from someone who did the recording and knows firsthand what the recording venue originally sounded like.

If you have sufficient clean amplifier power and linear driver volume displacement at all frequencies of interest to ensure low distortion at high sound pressure levels, you can use electronic control of the input signals to achieve any tonal balance you want. It can be flat, or you can set up a group of EQ curves for different types of recordings to compensate for original balance settings that may not be to your liking. Those curves can be stored in the memory banks of the equalizer for instant recall. You are only limited by your imagination and patience.

Before you embark on your next speaker project, do yourself the favor of looking more closely at some of the digital speaker control devices available in the market. By concentrating on achieving the key parameters I mentioned and then using digital control of the speaker feed, you will achieve world-class results and control for a lot less than many of the commercial solutions available. ❖

Vented 8″ Driver Subwoofer, Part 3

The conclusion of this series covers the testing of the subwoofer design using an MCM 8″ driver, along with the development and construction of the enclosures. **By G.R. Koonce and R.O. Wright, Jr.**

In Part 1 we demonstrated that it is unnecessary to use flat alignments for subwoofers used only up to about 100Hz, so we developed a design for the MCM #55-2421 driver using a non-flat alignment (Figs. 1–4). By accepting about 3dB loss in sensitivity, but not power capability, we developed a damped vented box (VB) design that would have a f_3 down near 30Hz. This design uses a net box volume of $0.7ft^3$ tuned to f_B at about 35Hz.

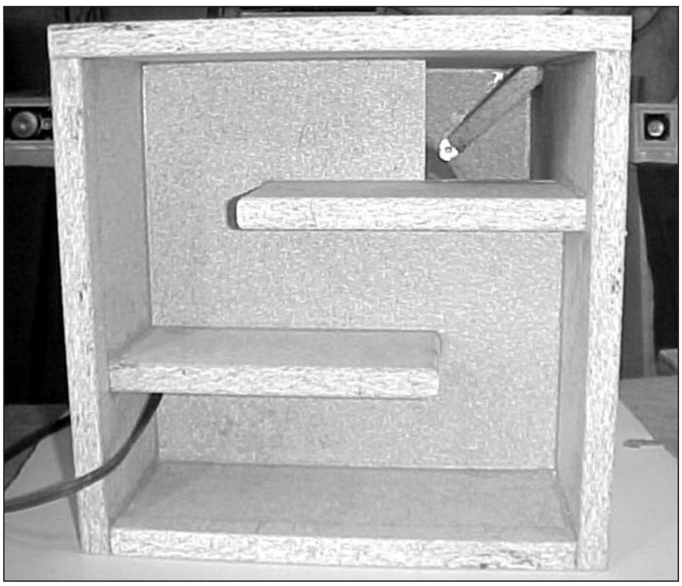


PHOTO 15: Rear view of box for MCM driver showing folded port duct.

Our earlier work also covered the requirements of a port to support such subwoofer designs. We finally decided that a 4″ constant diameter port or a flare-ended port of 3″ diameter would be acceptable. Parts 1 and 2 covered subwoofer development of a design using a 3″ flare-ended duct that worked just fine. The subwoofer covered here uses a non-flared rectangular port equivalent in area to a 4″ diameter pipe.

VERIFYING THE MCM DRIVER VB DESIGN

We put MCM driver unit A into a test box of net volume of $0.697ft^3$ and tuned it with a small diameter (1.97″) duct for low-power testing. *Figure 30* shows the input impedance, and the dip (marker #1) indicates f_{MIN} = 36.3Hz. Thus the volume and tuning are slightly below the design values of V_B = $0.7ft^3$ and f_B = 37.5Hz. *Figure 31* shows the near-field cone and port output magnitudes corrected for area difference. The cone response dip shows f_B is slightly below f_{MIN}, which is typical of the VB. Note that the port output dominates the system response below about 52Hz as was predicted with this design (*Fig. 2*).

Vector summation of the cone and vent responses produces the system response (*Fig. 32*) not far from the predicted response. Above 100Hz, the response is rising toward the rated driver sensitivity. Below 100Hz the response dips down to a shelf of about –3dB. The peak at 40Hz is higher than predicted, possibly not enough damp-

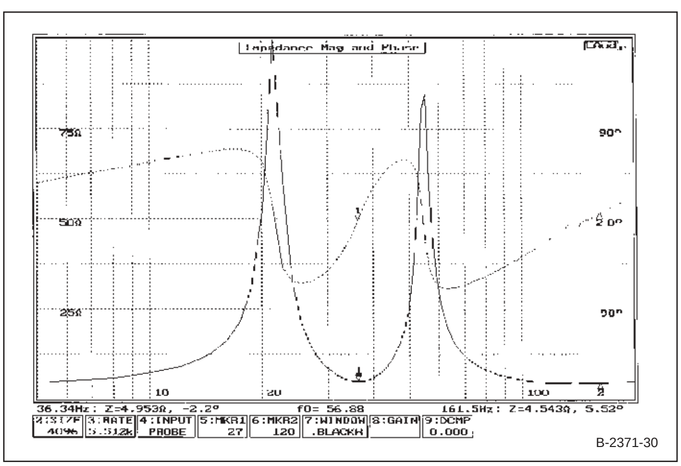


FIGURE 30: Input impedance for test of MCM #55-2421 driver A in a VB.

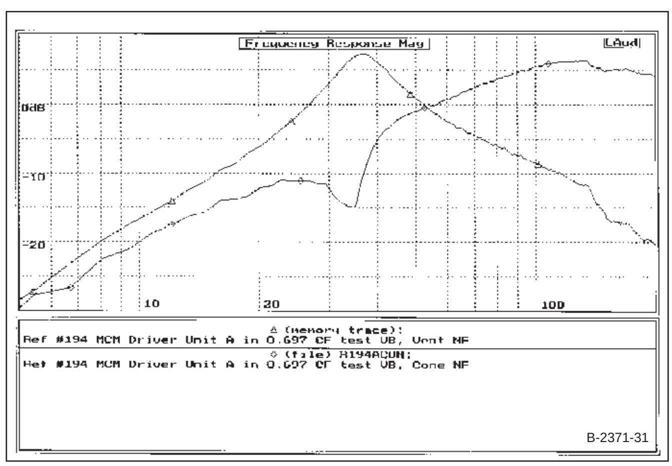


FIGURE 31: Near-field cone and port outputs for MCM #55-2421 driver A VB test.

ing in the test box or peaking built into the driver.

Low-frequency peaking appears in many subwoofer drivers, reported by ROW's sources to be accomplished by tuning the vented magnet structure so common on these drivers. The response looks usable from about 32Hz to over 100Hz. *Figure 33* shows the measured system group delay; ignore anything below 30Hz where noise is messing up the measurement. The group delay is about as predicted in Fig. 2.

ENCLOSURE DESIGN

The design for this driver is a net box volume of $0.7ft^3$ tuned to f_B in the 35 to 40Hz range. We decided to build this system using a rectangular port duct constructed of particleboard on the rear of the enclosure. A study showed that a rectangular duct of 3.33″ by 3.75″ (the area equivalent of a 4″ ID pipe) would fit on the intended box. Again based on equations developed by David Weems, the predicted duct length was 32″ to 23″ for the 35 to 40Hz f_B range.

The design developed corrected the gross box volume for volume loss to the driver, and then we added the duct to the back of the box. The vent duct is three horizontal passes across the back, giving a computed length of about 32″, which, we hoped, would produce an f_B down near 35Hz. The vent is "crude" without inlet or outlet flares and has no corner turning blocks where the duct makes 180° bends. Such a vent duct will likely generate turbulence, which is known to reduce the vent maximum flow capability.

The enclosure feeds the vent at a top, rear corner, and the vent exits at the bottom edge of the back. You can shorten this duct by widening the exit hole

TABLE 8
SIZES OF PANELS AND PIECES FOR ONE BOX

QTY	ITEM AND SIZE IN INCHES		
1	Front panel	11½ by 11½	(Cut from 11½″ strip)
1	Rear panel	11½ by 11½	(Cut from 11½″ strip)
1	Back board	11½ by 11½	(Cut from 11½″ strip)
1	Top	16⅜ by 13	(cut from 13+ strip for final routing)
1	Bottom	16⅜ by 11½	(Cut from 11½″ strip)
2	Sides	16⅜ by 12¼	(Cut from 12¼″ strip)
2	Port dividers	3¾ by 8.17	
4	Side stiffeners	9⅜ by 1.8	(length hand fitted to box)
4	Corner stiffener blocks	4.1 by 4.1 triangle, see *Fig. 39*	

PHOTO 16: Rear view of front panel with driver mounted.

in the back board. Locating the vent entrance right against the box walls is clearly not ideal, but necessary to get the maximum duct length on the smallest box.

Figure 34 shows the basic layout of this box, which contains three vertical panels. The front panel is set back 1″ from the front of the box to form a grille frame that keeps the grille cloth well away from the high displacement driver. The second vertical panel—the rear panel—is actually the rear of the acoustic portion of the box and is where the vent duct starts. The last panel is the back board that finishes the vent duct. *Photo 15* is a rear view of the assembled box with the back board removed showing the folded port structure.

The external dimensions of this box, built of ¾″ particleboard, are reasonable: 13″ high, 13″ wide, and 16-⅜″ deep, including the 1″ front grille frame. While not the 1′ cube we desired, this seemed like a reasonable design to try. We would learn whether a crude vent of this size without flares or corner turning blocks would perform OK for a small subwoofer application.

Part 1 covers the basic construction for small subwoofers, which you should review. Part 1 also details basic front-panel construction. *Photo 16*, a back view of a FP with the MCM driver mounted, shows the T-nuts and relieving.

BUILDING THE SUBWOOFER BOXES

Figure 35 shows a side view of the box and the locations of the three vertical panels and the port dividers. *Figure 36* is a front view indicating the size and locations of the speaker mounting and screw holes and also showing the positioning of the FP stiffeners. *Table 8* lists the sizes of the box panels and pieces all built from ¾″ particleboard. *Figures 37* and *38* are rear views of the rear panel and back board, respectively, showing the port "cutout" location. Note on *Fig. 37* the location of the port dividers and the hole for the wire to the driver. We used #16 zip cord through a 9⁄32″ hole.

The cutout in the back board sets the box tuning. Our plan was to clamp the back board into the box and measure f_B, then make the cut wider until we obtained the proper f_B. We started with the 3¾″ wide cut and retained it after testing, so this is the final dimension (*Fig. 38*).

Table 9 is a list of items you need to build a pair of boxes. *Figure 39* shows the dimensions for the port divider blocks, the side wall stiffeners, and the FP corner stiffener blocks. The rear panel and back board are stiffened via the port divider blocks. We could not quite tolerate the 90° corners on the

PHOTO 17: View inside box before installation of top.

TABLE 9
PARTS REQUIRED TO BUILD TWO SUBWOOFERS

QTY	ITEM
1 sheet	¾″ Industrial grade particleboard (49″ × 97″) cut into strips
2	MCM #55-2421 8″ subwoofer drivers
16	10-24 pan-head machine screws 1″ long
16	10-24 T-nuts
	Silicon rubber (to seal speaker wire)
8′	#16 Zip cord
Box	Elco #V2061 or equivalent 1⅝″ long panel nails
	Yellow glue
	Grille cloth (need two pieces bigger than 13″ × 13″)
	Cover strips for around grille frame
	Owens-Corning #705 (4′ × 8′ sheet) or other damping material
2	Barrier strips to terminate driver wire
	Sheet metal screws to mount barrier strips is used
	Optional feet or spikes for bottom of enclosures

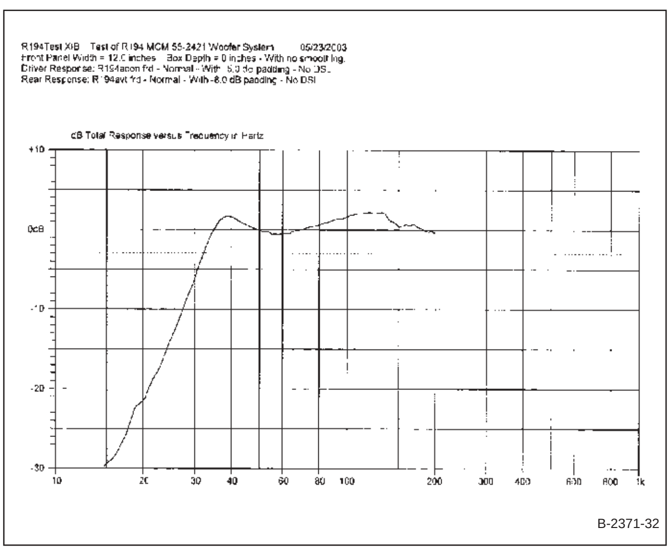


FIGURE 32: System response for MCM #55-2421 driver A VB system.

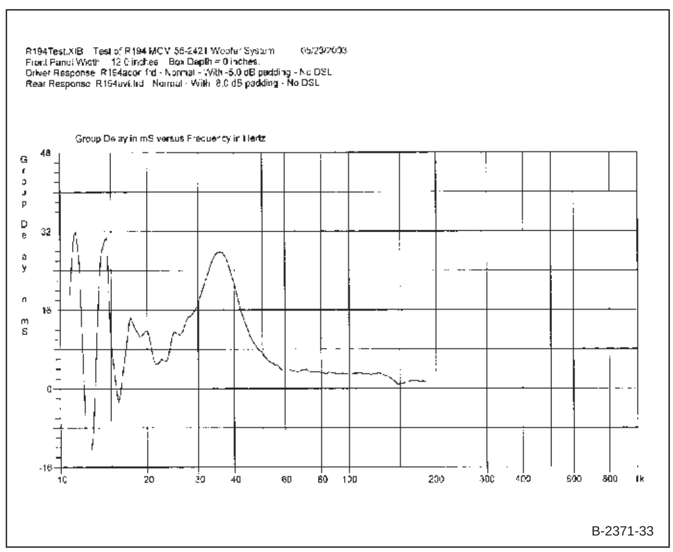


FIGURE 33: System group delay for MCM #55-2421 driver A VB system.

inner ends of the port divider blocks, so we rounded them on both sides with a ¼″ radius rounding router bit (*Photo 15*).

This box can be messy to construct if you don't do things in the proper order. We assembled the boxes as follows using glue and nails on all joints:

1. Attach the lower port divider block to the back of the rear panel, making sure it is flush with the rear panel edge on the box's right side (box right is on your right when you face the box).
2. Clamp the right side to the bottom, making sure the front edges match, and then nail it to the bottom. Remember to glue all joints before clamping.
3. Then carefully position the fabricated FP, clamping it to the right side and bottom, and then nailing it. Don't put it in backwards!
4. Clamp the rear panel with the lower port divider block attached into the proper position and nail it. Be sure to verify the back board will fit properly before you permanently clamp the rear panel.
5. Then clamp the upper port divider into position on the rear panel, keeping it flush with the box's left side, and nail it.

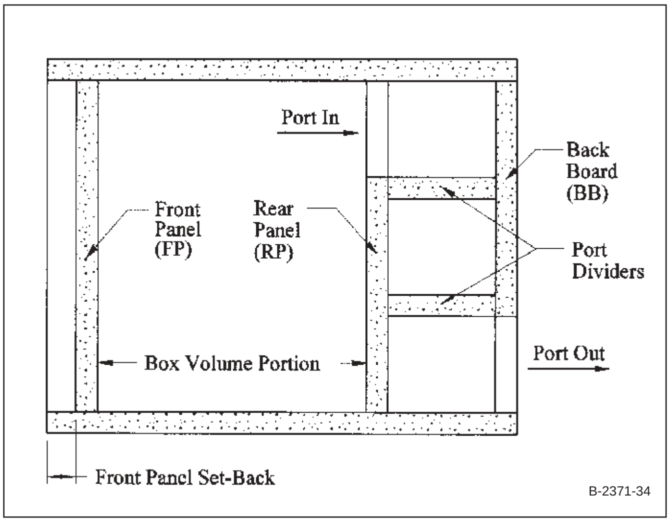


FIGURE 34: Basic layout of enclosure for MCM driver.

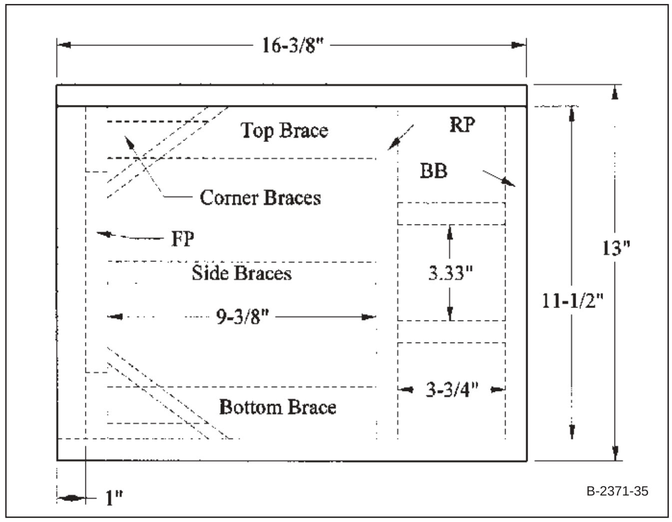


FIGURE 35: Side view of box showing dimensions.

6. Then clamp the left side into position and nail it. Be sure the left side is flush with the bottom at the front edge and the back board will fit properly.

INSTALLING THE STIFFENERS

You perform this work through the open top of the boxes. Fabricate and hand-fit the four side stiffeners and FP corner stiffeners. You should file a notch in the front end of all these stiffeners to just clear the T-nuts with enough gap to prevent rattles. Depending on your construction accuracy, you may need to file some of the stiffeners to just clear the machine screws sticking through the T-nuts.

At this time we installed all the stiffeners not associated with the top and glued and nailed the side stiffeners. We simply glued and clamped the corner stiffeners into position until the glue dried and installed the two top corner stiffeners and top panel stiffener after we fit the top. We gave the internal joints glue fillets to ensure an airtight box with no rattles. The yellow glue tends to suck into any air gap, so keep applying glue until a visible fillet remains after drying. Be especially careful where three panels join. Install the driver wire at this time, sealing it at the rear panel with silicon rubber.

INSTALLING THE DAMPING MATERIAL

We installed the 1″ thick Owens-Corning (OC) #705 damping material on the sides, bottom, and rear panel, cutting the pieces to fit between the stiffeners and gluing them in (yellow glue). You can easily mark the OC #705 material with a dull #2 pencil and cut it with a serrated bread knife.

We gave the damping material around the port entrance special treatment, cutting the pieces for the top and side back about 1″ from the rear panel and then trimming them at about a 45° angle to form a crude flare. We cut the piece for the rear panel to a matching 45°-angle flare.

Cut the pieces of damping material on or touching the top at this time, but don't install them until you've fitted the top. The inside top seams must be accessible for you to apply the glue fillets. *Photo 17* is an inside view of the box through the open top showing the stiffeners and damping material installed at this time. We simply set the top stiff-

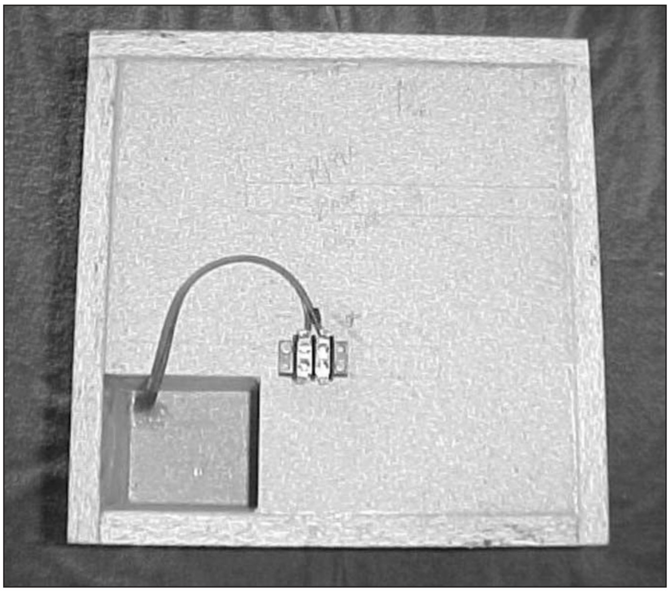


PHOTO 18: Rear view of the completed subwoofer.

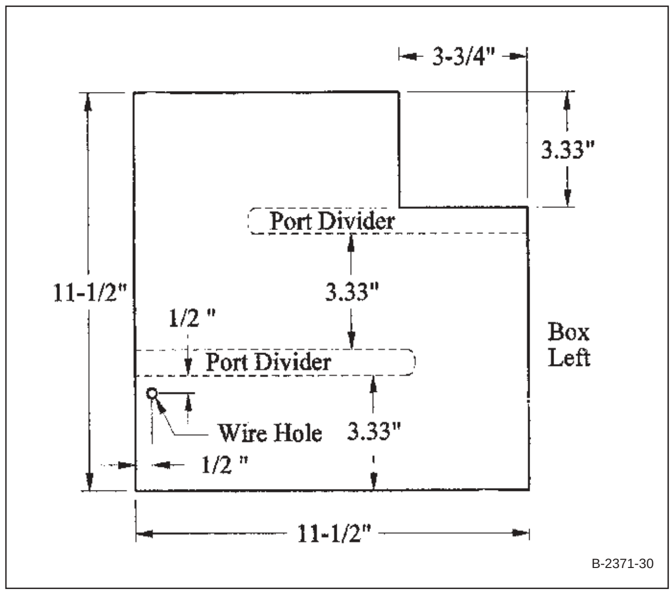


FIGURE 37: Rear view of rear panel showing start of port.

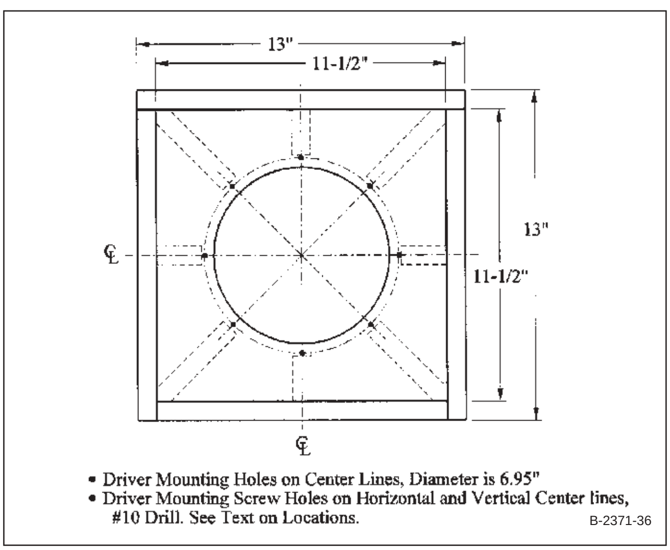


FIGURE 36: Front view of box.

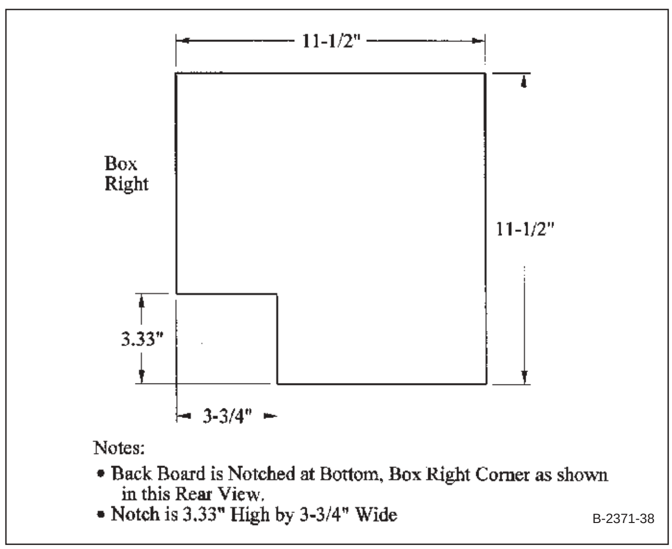


FIGURE 38: Rear view of back board showing end of port.

ener blocks in place for this photo and installed them later. Note the driver wire has been installed.

FINAL ASSEMBLY OF BOXES

Now clamp the over-wide top into position and mark the location of all mating panels on it. Make sure the top is flush with the front on both sides and that the top overlaps the full depth of both sides. Mark the position for the top stiffener brace, remove the top, and drill nail holes in the appropriate places. You can now install the top and rout the side edges flush with the box sides once the glue has dried. Then install the corner stiffeners and brace associated with the top and give all seams a glue fillet. Then install the pre-cut top damping material pieces—top pieces first, then side pieces.

We kept the back board free so we could experiment with the box tuning (discussed later). Unless you plan to adjust the tuning, you should cut the port exit hole shown in *Fig. 38* and install the back board at this time. Use lots of glue to ensure no leaks between the back board and the port dividers and also seal around the outside edge of the back board. Our testing demonstrated that port leaks are fatal. You can now fill external nail heads, sand the boxes, and apply your favorite finishing technique.

Solder the driver wire to the driver and install it. We installed driver unit A

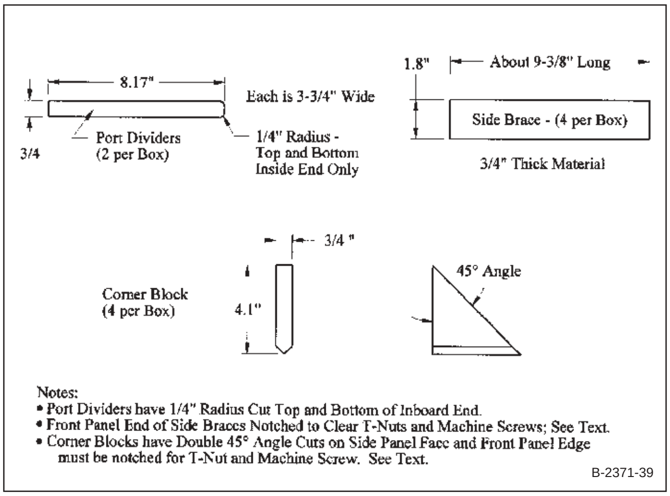


FIGURE 39: Sizes of other pieces for boxes.

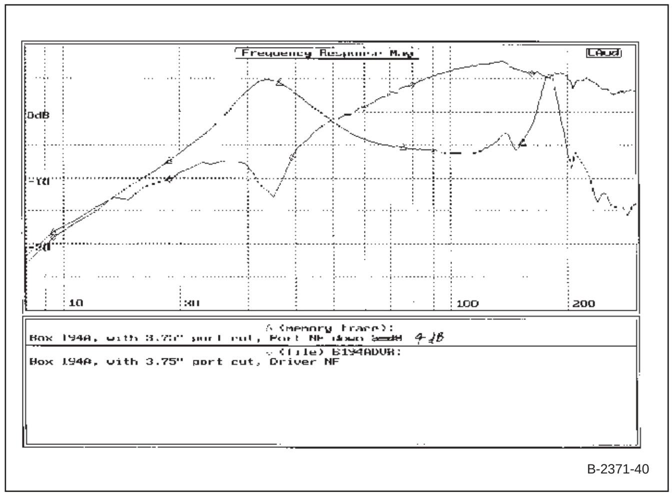


FIGURE 40: Driver and port response for enclosure A.

in box A, and so on for convenient record keeping. Orient the driver so that the four ribs from front rim to magnet structure are aligned with the four side stiffeners.

The MCM drivers come with a thin foam gasket covering only a portion of the inside of the rim. We retained this gasket and tightened the machine screws till the gasket was squeezed to almost nothing. No other sealing was necessary.

After driver installation, you may staple a very transparent grille cloth to the front edge of the box and then miter half-round or other shaped trim strips to fit around the front of the box. But for now, we left the grille cloth off.

After our tuning work, we installed the back board, filled the nail heads, and sanded the area. We added barrier strips to terminate the driver wires. *Photo 18* shows a rear view of the finished enclosure, while *Photo 19* is a front view without the grille cloth.

TUNING AND TESTING OF THE SUBWOOFERS

After driver installation, we tuned the enclosures. With the back board port opening cut to the 3¾″ design width, the computed vent length is about 32.7″. This length is possibly extended slightly by the OC #705 damping material cut to form a crude flare on the box walls. The port length equation indicated this should give a f_B of about 35Hz, yielding the best f_3 extension.

With subwoofer enclosure A, we installed the back board with a pair of carpenter clamps and tested the system. *Figure 40* shows the cone and port magnitude responses via near-field technique and adjusted for area difference. The cone response shows f_B is in fact at 35Hz, so David Weems' equation for a rectangular port predicted the tuning very accurately. We decided to leave the port full length and evaluate the system with this tuning.

Also note in *Fig. 40* that the driver response shows a nice dip at f_B, indicating relatively low loss in the port, which has a nasty resonance just below 200Hz. With this resonant peak, the port would be unacceptable with a woofer design but is fine for a subwoofer where a fourth-order crossover will roll off the input signal above 100Hz.

Reference 3 develops the concept that looking at the input impedance of your VB is a good way to establish just how "good" your port is at handling power. At low power you get the proper double-peaked curve with the dip at f_{MIN} very close to the box tuned frequency f_B. As the power level to the driver is increased, the first sign of port trouble will be a drop in magnitude of the

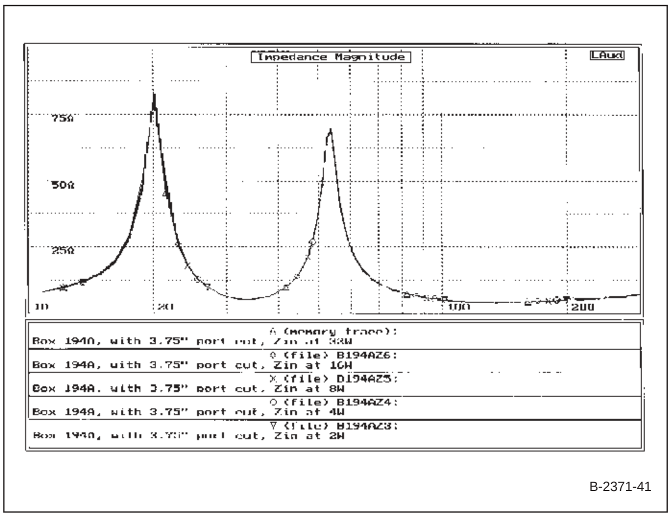


FIGURE 41: Input impedance of enclosure A versus power.

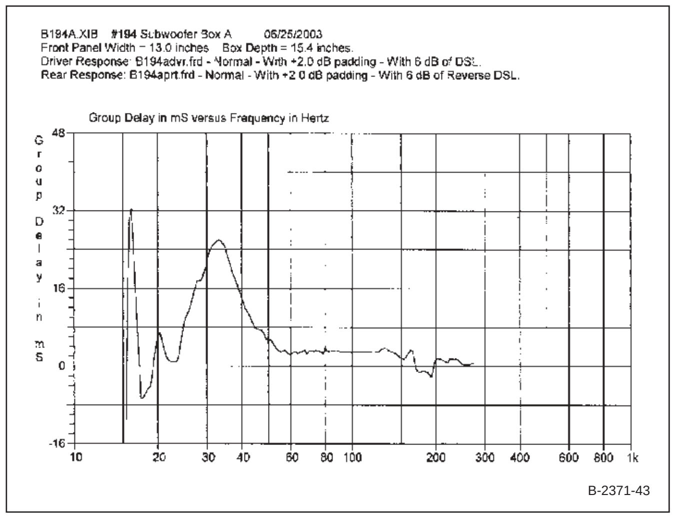


FIGURE 43: Measured group delay for enclosure A.

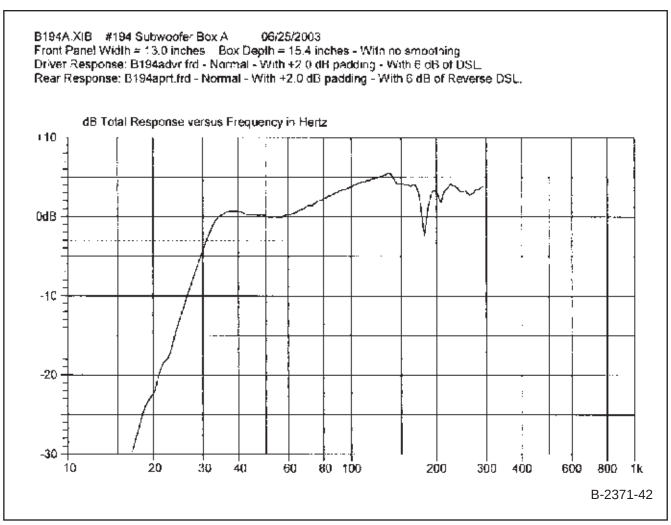


FIGURE 42: Measured system response for enclosure A.

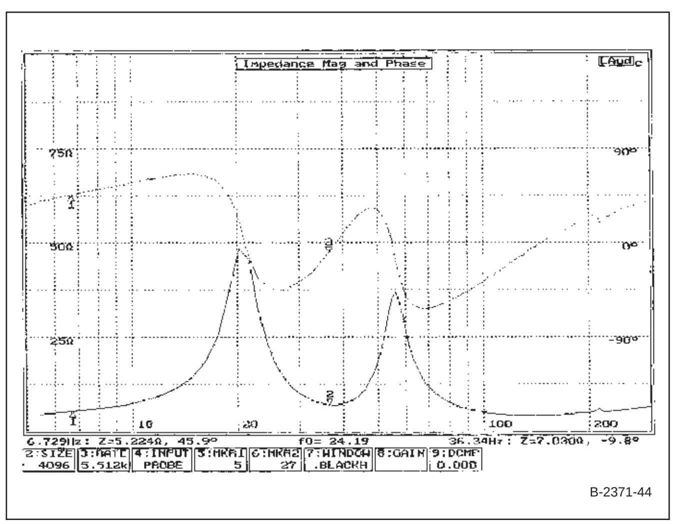


FIGURE 44: Input impedance for first test of enclosure B.

lower-frequency peak. When your port fails completely, the lower peak will disappear, leaving, in effect, a "leaky closed box."

While our test setup is not intended for high power testing, we did manage to measure subwoofer enclosure A's input impedance from low power up to 32W in 3dB steps. *Figure 41* shows that from 2 to 32W there is no change in the height of the lower-frequency peak. We do not know how far above this power level the port can sustain, but at 32W the subwoofer was becoming rather loud.

We fed the driver and port responses into software that does a vector summation, including the effects of diffraction spreading loss on both signals and takes into account the port location and the width and depth of the box. *Figure 42* shows the system response for this subwoofer. This is close to the predicted response for f_B at 35Hz. The f_3 frequency is about 32Hz, and the response is rising at the top end of the band.

This rise is more than predicted in the computer design due to the diffraction spreading loss effect starting to appear with rising frequency. The crossover low-pass filter will moderate this peaking and remove the trash above 100Hz. Thus enclosure A is a subwoofer covering 32 to 100Hz as desired. *Figure 43* shows the measured group delay for the system. Ignoring anything

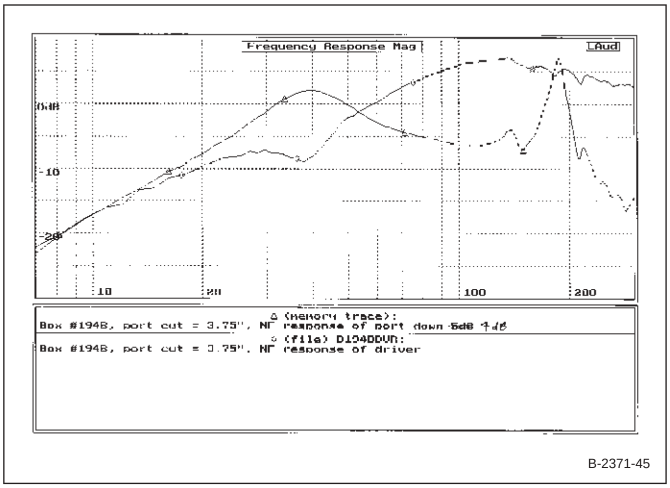


FIGURE 45: Driver and port responses for first test of enclosure B.

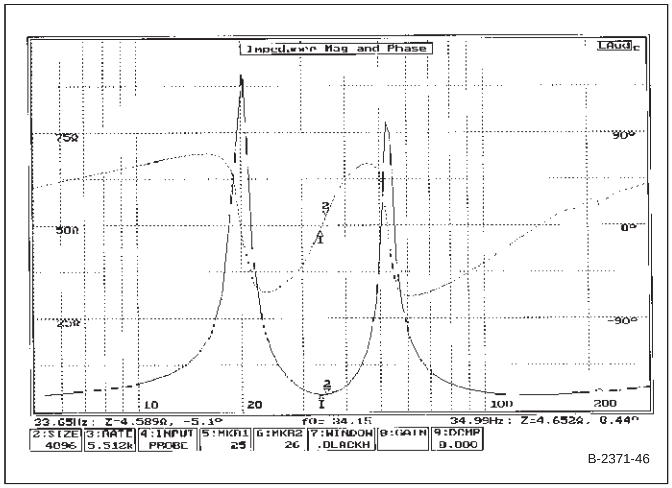


FIGURE 46: Input impedance for retest of enclosure B.

below 30Hz, where noise interferes, the group delay is about what we expected with the low f_B tuning.

We decided to leave subwoofer enclosure A with the full-length duct and tune subwoofer enclosure B starting with this same duct length. Similar drivers in a box of the same construction should produce the same results, but that is not what we initially measured. *Figure 44* shows that the magnitude peaks in the input impedance were greatly reduced in amplitude relative to enclosure A. The driver and port magnitude responses (*Fig. 45*) show f_B is higher and the dip in driver response at f_B is not as deep as for enclosure A.

This all smacks of additional system losses, most likely a port leak. The back of enclosure B did not fit as tightly as for enclosure A and looked as though it might be clamped crooked. GRK verified the driver mounting screws were tight, carefully re-clamped the back board, and taped around the edges of the back board.

Figure 46 displays the input impedance measured during retest, which now resembles enclosure A. *Figure 47* includes the driver and port magnitude responses, which now show the deep driver response dip restored to f_B = 35Hz. This illustrates just how critical it is to make the port structure airtight. *Figure 48* is the measured system response for enclosure B, now a reasonable match for the response of enclosure A. *Figure 49* shows measured group delay results for enclosure B.

Both completed boxes featured full-length port ducts, i.e., 3¾″ wide cut in the back board. Final measurement of input impedance showed the ports were properly sealed with low loss. The resultant subwoofers, with the proper crossover, will cover the range 32 to 100Hz with mild peaking in the 70 to 90Hz range.

See Part 2 for details of the listening system and source material used.

LISTENING RESULTS WITH THE SMALL SUBWOOFERS (BY GRK)

These subwoofers produced a good-sounding system. We encountered no problems with rattles or strange sounds. Integration with the satellites was good with drum strikes fast and taut. At a cautious level (as I watched the cone motion), the system did not seem able to play as loudly as I remember the single 12″ IB subwoofer, but this is a questionable perception. After my initial caution, the power level to the satellites limited the playing level for classical orchestral music, just as had occurred with the 12″ subwoofer (see Part 2).

As with the earlier pair of 8″ driver subwoofers, on bass-heavy material

PHOTO 19: Front view of the completed subwoofer without grille cloth.

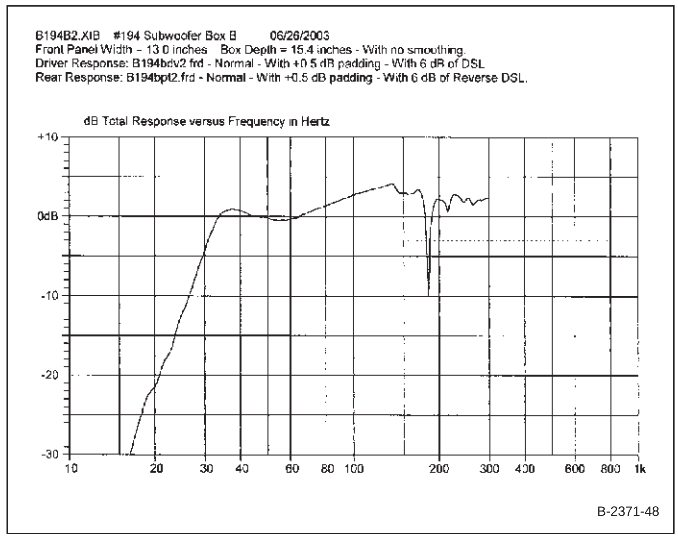


FIGURE 48: Measured system response for enclosure B.

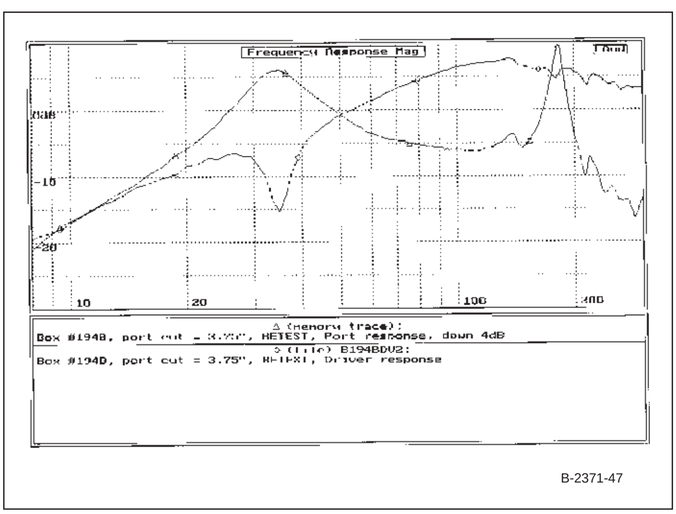


FIGURE 47: Driver and port responses for retest of enclosure B.

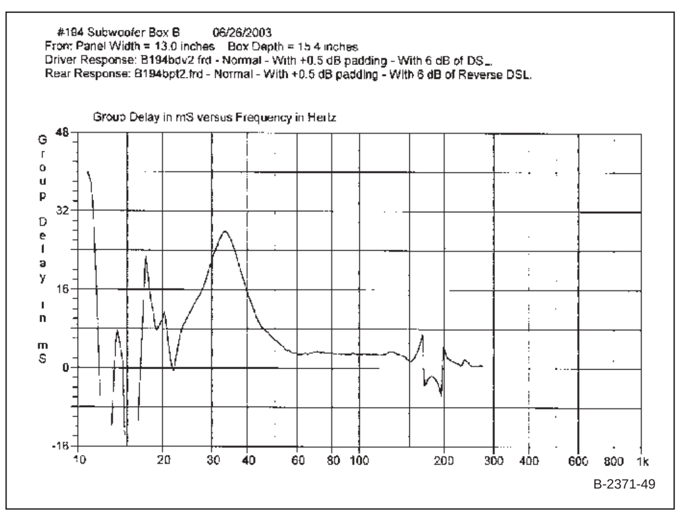


FIGURE 49: Measured group delay for enclosure B.

these units would take drum strikes shown to be 256W peak. The fact that the subwoofers produced drums this loud is not proof the ports "hold up" to this power level. Even if the subwoofers converted fully to a closed box they might still produce the drums.

I thought that blocking the ports and listening would be an interesting test. With the speaker wire passing through the port exit, a solid cover was not possible, so I cut two plugs of 1″ thick OC #705 damping material. After slipping in the plugs, about 30 seconds of listening revealed no change to the sound of the drums.

I decided to pull the plugs for a comparison listen and found the first drum strike must have blown them out of the port! I pushed them in securely and listened again. Now the drums were clearly not as loud, sounded a bit slower, and the music lost overall depth. Clearly the ports are still contributing at these high peak power levels.

I tried the "Earth Drum" on Mickey Hart's Planet Drum CD (Rykodisc RCD 10206) with these subwoofers. Since this drum is basically below 30Hz and sustained a fair amount of time, I limited the peak power to near 128W. At this level the Earth Drum was clearly audible, but not as prominent as I remember on the 12″ IB subwoofer. With the vent plugs installed, I saw the cone moving when the drum was due, but I did not hear it. The ports are clearly extending the subwoofers' LF response to reasonably high power levels.

You can detect one difference between these subwoofers and the earlier set, with the flared port duct at the port output. With the flared port boxes you almost do not hear drum strikes, and 2″ back from the flare you do not feel any air blast. With these boxes you hear the drums clearly at the port output and feel the expected air blast.

On certain musical cuts, I believed these subwoofers were not quite as taut as the earlier units with the TBspeakers W8-704C drivers and flared ports. I thought that perhaps these boxes could benefit from a bit more internal damping material, something that we have not yet tried. Of course, the difference in sound could also be caused by the rather crude port structures used on these subwoofers. ❖

Tube Audio Construction Tips

Part 4: Transformers

This author offers some of his tricks for dealing with mistruths and misinformation about output transformers. **By Graham Dicker**

My first trick relates to the power rating of output transformers. Most manufacturers select sufficient core in designing an output transformer to ensure that it operates without any mechanical noise at the full-rated power. This is achieved by keeping the flux density low for the size of the core. Taking into account the winding resistance of the transformer, you can push the power rating by as much as 100 percent on most transformers. For example, you can use a 100W transformer up to 200W if you do not mind a little heating of the core and a noisy transformer.

Next, you can use that same transformer rated at 100W at 20Hz at higher power levels if the cutoff frequency is higher. For example, a bass guitar amplifier need only have a lower corner frequency response to 40Hz (bottom E). Because this is an octave higher than 20Hz, a 100W hi-fi grade output transformer can now be rated at 200W.

For a lead or rhythm guitar, the lower corner frequency is 80Hz, an octave higher again. You can now raise the rating to 400W for the same transformer. As you progressively increase the rating, the efficiency will fall off due to a number of factors including winding losses, but this has never stopped me from trying it out. I have often used a 10W push-pull hi-fi output transformer in a 50W guitar amp as a low-cost replacement.

LOW-COST OUTPUT TRANSFORMERS

PA line transformers are my best source of output transformers (*Photo 1*). They come in different sizes and impedances for line matching in PA applications. Moreover, they are really low-cost, at around $3 for the 5W units and $12 for the 40W ones. The core material is good and the frequency response from 20Hz upwards. Many even have convenient taps so that you can use them as push-pull output transformers without any changes at all.

For single-ended use, all you need to do is remove all of the laminations and repack them with an air gap, which is used to stop the transformer from saturating with the steady plate current that flows through the transformer primary. The air gap needs to be optimized for one current only, so you may need to play around a bit to get it right. As a guide, start with some 100gsm card for around 80mA plate current.

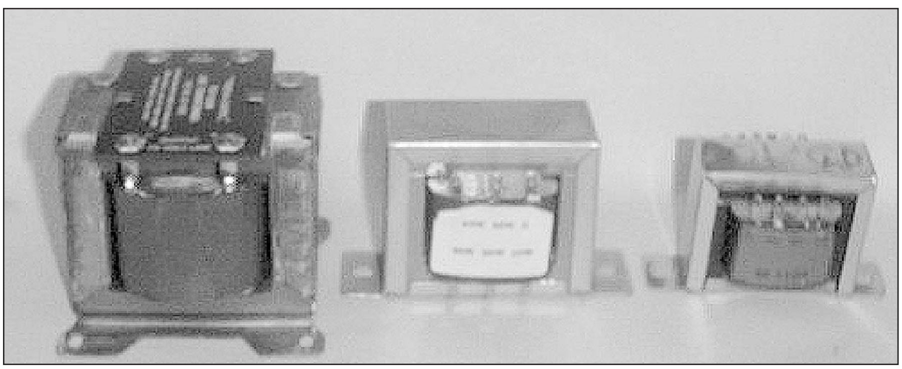


PHOTO 1: A selection of available PA Line transformers.

MAINS TRANSFORMERS FOR OUTPUT

If you work out the RMS plate swing for either push-pull or SE, you can often find a power transformer with about the same approximate turns ratio and voltages. A word of warning: Most are wound with standard winding wire, which has a breakdown voltage of 90V from the center conductor to the outside of the enamel. Many audio transformers use audio-grade winding or magnet wire with a 200V breakdown rating. Additionally, most audio transformers are vacuum impregnated with varnish to improve the breakdown voltage rating and to quiet the transformer.

Power transformers are rated for use at either 50 or 60Hz. For each octave lower you choose to cover, you need to reduce the power rating to half. For example, you can use a 60Hz 100VA power transformer at 50W at 30Hz or 25W at 15Hz. This is why output transformers are usually larger than the power transformers in most hi-fi amplifiers. The low voltage windings often work out well as the voice coil impedance windings.

For SE use, you can again unpack the transformer and add the air gap as mentioned before. A 110V to 9V 2A power transformer repacked makes a good 5W SE output transformer for a 6V6 or 6L6 tube. Don't expect any more than 80 percent efficiency. The frequency response on those I have tested is very good, exceeding 30kHz.

You can use a high voltage secondary of a tube power transformer— i.e., a 300-0-300— for a push-pull amplifier. The efficiency will be low due to the primary windings not being made around the center of the core, plus the DC resistance will be different for each leg, resulting in an imbalance in plate load currents and higher distortion. However, for a guitar amplifier, or just fooling around, this may work out just fine.

MICROWAVE OVEN TRANSFORMERS

These are my favorite devices at the moment. Around 3.6 million microwave ovens end up as worldwide landfill each year. While I am not preaching being green here, it is a great opportunity for recycling, something that I have always had a passion for. My wife calls it hoarding.

I always hate to throw out something that may be useful one day (the Amateur Radio credo). My past experience has shown that after cleaning out accumulated junk, the next week I always need whatever I threw out, and then I must buy a replacement. The moral of the story is don't throw anything out.

The mains transformers from microwave ovens are a great source for the home constructor, because you can easily modify them as both an output and mains transformer for tube projects. Furthermore, they are readily available and usually free from microwave oven repairers or the local recycling center. My source here in Australia exchanges stubbies (a small 300 milliliter bottle of beer usually sold in slabs of 24) on a one-for-one basis for good microwave oven transformers. A Bud-drinking microwave oven repairer in your local area may also be persuaded into a similar arrangement. (The barter principle here in Australia still does not attract government taxes.)

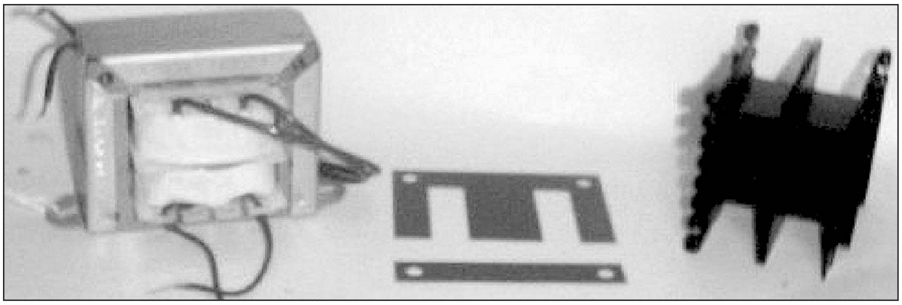


PHOTO 2: A homemade output transformer showing bobbin and E & I laminations.

The transformers come in different forms and ratings. The usual domestic transformer has an intermittent rating of 600 to 700W and is good for 400W continuous. I use them as power transformers for 100W per channel SE amps. The transformers from industrial ovens have an intermittent rating of 1000 to 1500W and are good for around 600W continuous. The cores are welded together, but easy to take apart with an angle grinder. I simply grind off the weld and make a new mounting frame from sheet metal to hold the new transformer together.

Unlike most transformers that are packed with interleaved laminations, these are usually packed with the I-laminations one way and the Es the other (*Photo 2*). This means that you can easily add an air gap for SE use. A single domestic transformer makes an ideal 100W SE output transformer. The high voltage secondary becomes the primary for the plate circuit, and you can use the 110V primary as a voice coil output for a 15R load.

Most of these transformers have one turn per volt for the core. This works out to 110 turns for the mains primary and 2400 turns for the high voltage secondary. If you need an 8R output, it is easy to modify the 110V primary winding to pick out a tap 30 percent from the outside at around 75 to 80 turns, or simply remove the old primary winding (110V) using a cold chisel and a hammer, and wind a new one. Some of my modified SE output transformers are shown in *Photo 3*.

For lower output power of up to 65W, you may not need to take the transformer apart to insert an air gap, due to the core size (one of my favorite thoughts on output transformers is why add an air gap when you can add more core). The core material varies depending on the manufacturer, but commercial ones almost always have high-grade GOSS cores. The measured frequency response of the domestic transformers is not fantastic—usually 3dB down at 7.5kHz, with the low-end usually very good at 10Hz. The commercial high-powered transformers are often good up to 13kHz. The obvious solution (or not) may be to use some feedback around the amplifier to flatten and extend the response.

I have used several of these transformers over time, even with the limited frequency response, in amplifiers using an SE813 for around 50W RMS. The sound has not been disappointing. In one of these, I actually recycled the whole microwave oven, reusing the case, the power supply, and the internal light to show off an 813 inside the oven chamber (even though the 813 produced enough light of its own).

SIMPLE OUTPUT TRANSFORMERS THAT YOU CAN WIND

Over the last 40 years or so, I have seen many methods of constructing output transformers. Some use a myriad of interleaved primaries and secondaries, with nice core material, some wound with litz wire, and so on. These perform extremely well with high efficiency, but have a high price tag (you will get what you pay for). My experience has been that most people who choose to build a tube amplifier love the sound but automatically put older technology and low cost in the same category. The cost of purchasing all new parts to build a tube amplifier is frightening. With this in mind, most of this article shows what you can achieve with little cost and a lot of compromise. One of the best cost savings is to wind your own transformers. For this you need a simple coil-winding machine (described in a future article).

SIMPLE SCRAMBLE WOUND OUTPUT TRANSFORMER

To wind a simple transformer you need wire, a core, a former, and a plan. The transformer described here is not exactly accurate, but within the ballpark, and does not need a lot of heavy-duty design work to get there. Keep in mind my section about power transformers and others that can be pushed into the job before you decide to go down this road. You may wish to keep some of the windings on a power transformer and wind one or more new ones to save on some work. You can also recycle the

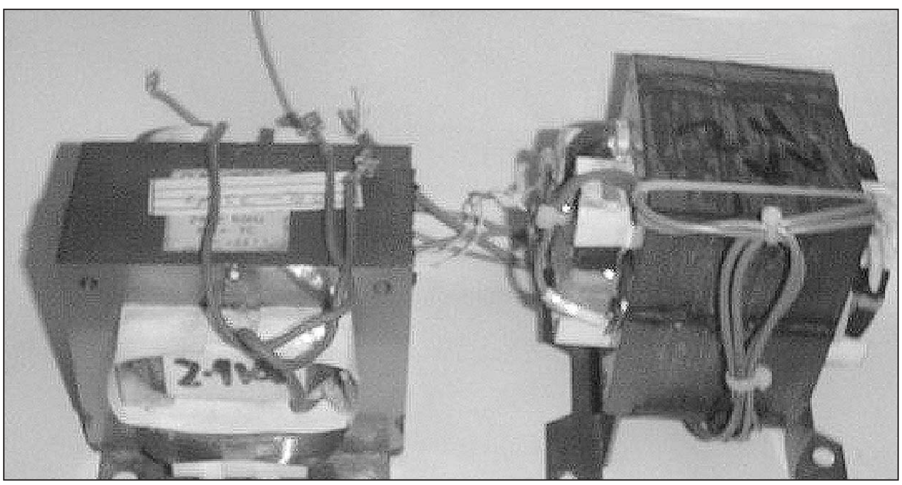


PHOTO 3: Microwave oven transformers modified as SE output transformers.

component parts from the microwave oven transformer here as well, such as the core and mounting frames.

I worked out this simple construction about 40 years ago in my apprentice days. One of the tasks I undertook was to design a 300W push-pull 813 tube PA amplifier for the company I worked for. These were used to provide PA for racecourses, and several dozen were ordered and manufactured. A few variations also found their way to drive multiple quad boxes for local Adelaide rock guitarists. Several of these amplifiers are still in use today, and the owners are most uncooperative to part with them for any amount of cash.

The problem I faced back then was that no two wound transformers tested the same on the bench. This invariably came down to a number of manufacturing and simple assembly errors. An idiot-proof design was needed. Unfortunately, if you have idiots working on a production line, a foolproof design may not be enough (yes, you may quote me on this).

Assuming that you are the brave and adventurous type, this is how to go about it in a few simple steps. Work out the transformer peak-to-peak swing and multiply it by 0.35 for the RMS. As an example, a push-pull 6L6 amplifier running from a 400V DC supply would provide 1600V P-P unless the tube forward voltage drops off approximately 400V, leaving you with 1.2kV P-P. This works out to 420V RMS.

Next, work out the secondary voltage in RMS. For example, for 20W RMS, allow a 20 percent loss in the transformer so the voltage required is for 24W into 8R, approximately 14V. Next

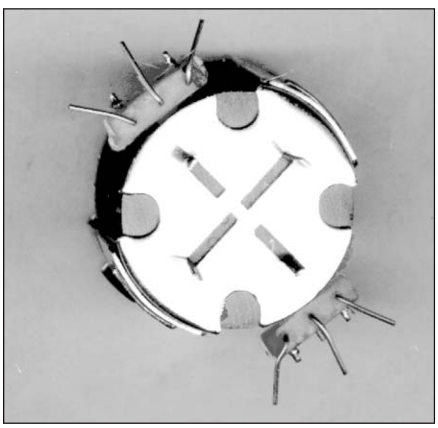


PHOTO 4: 100W potcore ESL output transformer for a midrange panel.

you need a core for this power rating, assuming a lower f3 of 20Hz. You can purchase a core, bobbin, and hardware from a supplier to suit, or take a stab in the dark with any available core around.

The best-guess method I use is to look at a similar-size power transformer to the core you plan to use, and work out the approximate VA of the power transformer. To do this, you add up all of the secondary winding capabilities. This is an easy step if you are going to strip an existing transformer, because you can work out the VA rating exactly.

Next, for a hi-fi transformer, divide the VA by 2.5 (for a 50Hz) or 3 (for a 60Hz) power transformer core. For my example, the core I was going to use is the same as a 100VA power transformer at 60Hz. Divide by three, and you end up with an audio core good for about 30W. Next you need the turns per volt (TPV). As a guide for 5 to 50VA cores using 10TPV, 50 to 100VA use 5TPV, 100 to 250VA 2.5 TPV, and 250 to 500VA 1TPV. This is not exact by any means, but a good rule of thumb.

If you are stripping an existing transformer and the 110V or 240V mains windings are intact and you do not wish to count turns, you can thread ten turns of fine gauge wire through an airspace on the existing bobbin, and measure the no-load voltage with a digital multimeter. Divide ten by the measured voltage to get the TPV of the core. The TPV is even easier to work out: you can simply count the number of turns as you unwind a winding (preferably a low voltage one, which has fewer turns to count), divide the number of turns by the measured winding, and you end up with the TPV.

Multiply the TPV of the core by the voltage you need in order to get the required primary and secondary turns. Knowing the available window size of your core, look at some wire size charts to pick a gauge of wire that will fill 80 percent of the window.

The primary is wound on the former first, using a split former. In this way, each half of the primary is wound at the same time, and will always have the same number of turns even though the primary/secondary ratios may change a little. The primaries can be scramble-wound without interlayer insulation for simplicity in construction. A layer of insulation between the primary and secondary comes next, followed by winding the secondaries in the same manner.

This gives you a transformer with two split primaries and secondaries. For an SE transformer, the primaries are wired in series. For a push-pull, the common wire comes out as the transformer center tap. You can also wire both secondaries in series. If you have enough window space to wind twice the number of turns, you can also use parallel hookup options. An SE transformer needs to be packed with an air gap; a push-pull does not.

This method does not account for issues such as calculating the primary inductance and others that you would normally do to make a high-quality output transformer. This is a quick and simple transformer to make (*Photo 3*)—less than an hour and with minimum cost, and with very close DC resistances for each half of the windings, better than in many commercial versions.

BIAMPING AND ESL TRANSFORMERS

After spending much time playing with ESLs and multiway systems, I discovered a really low-cost solution for output transformers.

In ESLs and multiway systems, it is convenient to have multiple amplifiers and panels, each covering a different part of the spectrum, usually no more than four octaves. An alternative to an iron core transformer is the ferrite or pot core. These have usable core use down to 300Hz for some cores. A small 2″ diameter core is good for up to the 100W power level; the TPVs are low, and the cost and simplicity worth consideration.

I have used pot core output transformers to drive midrange and tweeter panels with much success. The design guides for their use are well outside of this article, but a good reference is the Philips components and materials handbook for potcores (C4 12-81).

For ESL panels, a simple output transformer is shown in *Photo 4*. I wound this output transformer using a split bobbin with both windings made at the same time. This one has 300 turns on each part of the bobbin and is wired as a center-tapped choke. The center tap goes to the DC supply and the other ends to the tube anodes and ESL plates. A negative diaphragm bias voltage minus the DC plate voltage is used.

I have also tested line output transformers (LOPT) and switch-mode power-supply transformers (*Photo 5*). The LOPT tested was from an older black and white set without an integral tripler or rectifier. The frequency response tested was from 1kHz to beyond 200kHz.

The turns ratios and existing windings made these useful for a transmitting tube such as an 813 or 807. The anode is wired to the old primary and the ESL plates to the HV secondary side. A single 10MΩ resistor to ground from both secondary wires provides a return path for the diaphragm bias.

On one occasion, I also tried using the EHT rectifier and a 1nF 5kV filter capacitor to provide an audio-driven bias voltage. I used a lightning gas arrestor as a VR tube. It turned out that it was too hard to regulate, so I used a

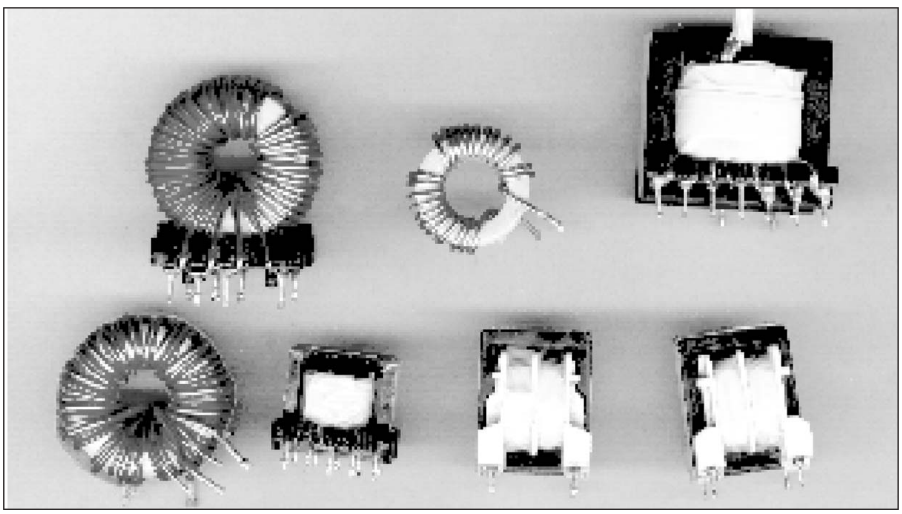


PHOTO 5: Selection of transformers taken from unused computer SMPSU.

more traditional arrangement.

The SMPSU transformers I tested all had useful output from 75kHz to beyond 200kHz, so were not suitable for audio use, but possibly good for ultrasonic pest eradication.

SETTING THE BIAS IN PUSH-PULL AMPLIFIERS

Without leaving the subject of output transformers, it is worth a few lines to explain my method of setting up the bias for a push-pull amplifier. The compromise here is a DC imbalance in the output transformer versus maximum usable plate dissipation of each tube, without a bifilar or split bobbin output transformer. You can never have both at the same time.

Many different ways of setting up the bias current in a push-pull class A amplifier have been published over the years. The most common method is to measure the cathode current and balance it for each tube. The second is to place a DC voltmeter from plate to plate and adjust for minimum voltage. Both of these methods fail for a number of reasons, including assuming that the anode voltage is the same as the supply voltage, failing to deduct the cathode bias and output transformer DC voltage drops to derive a total anode voltage, and assuming that the DC resistances of both halves of the output transformer are the same. My method is not simple but, I believe to be closer to optimum.

First, I pick one tube as the reference tube. I set this one up first, then the other. You need to know the maximum plate dissipation for the tube you are using. This information is available from data sheets or from the web.

Remove the second output tube. Measure the voltage from anode to cathode and the cathode current of the reference tube. The two multiplied together should equal the plate dissipation for a class A amplifier. You can adjust the bias voltage until the sums work out.

Next, install the second tube, and using a low-cost toy compass on top of the output transformer, adjust the second tube bias to ensure the compass does not deflect from north. Turn the chassis 90° and check again. Any imbalance in the output transformer magnetic field will cause the compass needle to deflect. If the core has a zero state of magnetism, then all is well.

The magnetic flux produced in the core of a transformer is dependent on the number of turns and the current that passes through it. Because most output transformers will have dissimilar primary resistances but the same number of turns, the DC voltage drop across them will be different. As such, the primary half with the larger voltage drop will need that tube to draw more current for the same plate dissipation. The offset is that by drawing more current, you may unbalance the output transformer and cause a residual magnetism. I have often found many amplifiers deliver nowhere near the rated output power or end up in class AB2 due to poor setup, or an increase in THD due to the imbalance.

INVERTER TRANSFORMERS

For those interested in car glass audio, the thought of building a DC to DC inverter has often been a problem. There has been little published about inverter transformers, and this is in the Black Art area I am trying to de-mystify.

The best method I found for winding an inverter transformer is to recycle the microwave oven transformer (working out new ways to recycle these is one of my favorite pastimes). I often remove the primary or secondary winding (depending on the output voltage required) with an old woodworking chisel and hammer. The core has 1TPV for the windings, there are usually 2400 turns on the HV secondary, and either 240 turns or 110 turns (depending on country of origin) on the primary.

For example, if I need around 350V DC at 1000mA, I remove the HV winding and add two windings of eight turns of car booster cable bifilar wound for the low voltage primary. If I need 1500V at 200mA DC, I remove the 240/110 primary and wind the same two lots of eight turns for the primary and make a tap on the HV secondary winding at around 45%. Remove the core welding and the "I" laminations with an angle grinder, add a piece of 70GSM card as an air gap, and some new mounting frames, and, presto, you have a good 300–400VA inverter transformer. You can use the commercial microwave oven transformers for higher power up to 1kW. ❖

Review

Audio Interconnects

By Charles Hansen and Muse Kastanovich

This test report concerns line-level audio interconnects, and does not evaluate those interconnects specifically associated with phono cartridges. Supra sent a sample of their balanced EFF-IXLR cable. Because my signal sources and measurement equipment are all single-ended, I did not attempt to make any measurements of this XLR-equipped cable. The cable appears to be identical to the single-ended interconnect except for the end connectors.

As with the speaker cables tested earlier ("Speaker Cables Review," *aX* May '04), the parameters of an audio interconnect are defined as a combination of electrical resistance, capacitance, and inductance. The DC resistance is a function of conductor length and cross-sectional area. As the frequency increases, the resistance of a conductor increases due to skin-effect losses. Also, the lower the dielectric constant of the insulation, the less the reduction (loss) in the wire.

Most audio interconnects are produced in the form of coaxial lines. The capacitance can be calculated from:

$$C = 2\pi\varepsilon_0\, \varepsilon_r\, l \log_e (r_o/r_i)$$

where ε_0 is the permittivity of free space, ε_r is the relative dielectric constant of the insulating material, l is the length, e = 2.718, r_o is inside radius of the shield, and r_i is outside radius of the inner conductor. The value of ε_r varies with material, temperature, and frequency.

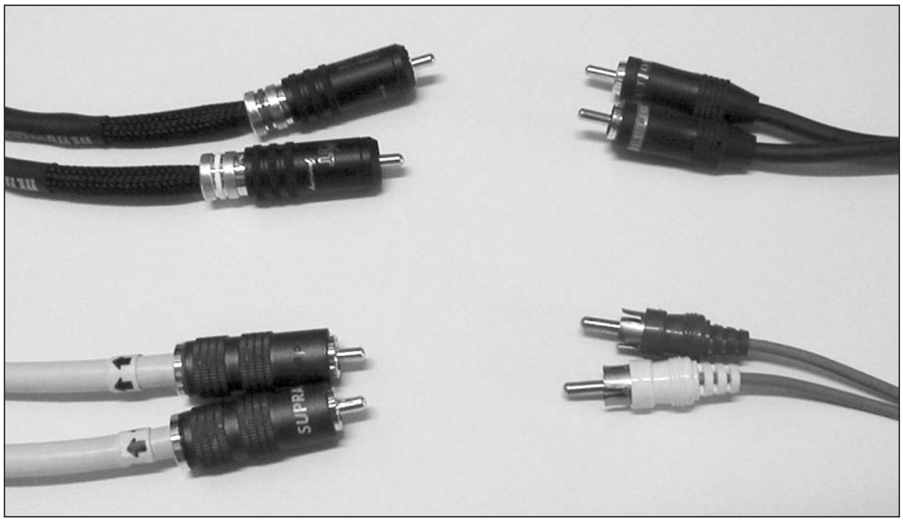


PHOTO 1: Audio interconnect test samples.

There are also audio interconnects made from woven or twisted wires. These round-wire interconnects form parallel wire capacitors. The capacitance of these interconnects is difficult to calculate due to the lack of a cylindrical ground plane, and the increase in capacitance due to the woven or twisted conductor configuration.

Interconnects also have the properties of series inductors. The inductance for two parallel conductors carrying equal and opposite currents can be found by:

$$L = [0.1 + 0.4 \log_e (d/r)]\, l \qquad \text{where } (d>>r)$$

where d is the distance between the wires, r is the wire radius, and l is the length.

Thus, you can see that the impedance of an interconnect will vary with frequency as well as with the distance between conductors. Capacitance varies inversely with distance, while inductance increases in proportion to distance.

The wire resistance is in series between the audio source and line-level input, and increases with frequency

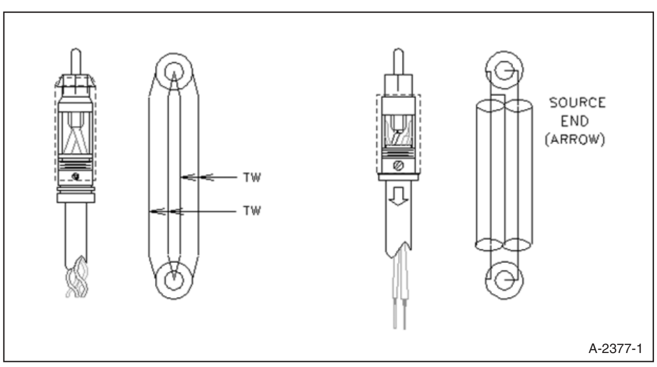


FIGURE 1: Kimber Hero and Supra EFF-IX construction and schematic.

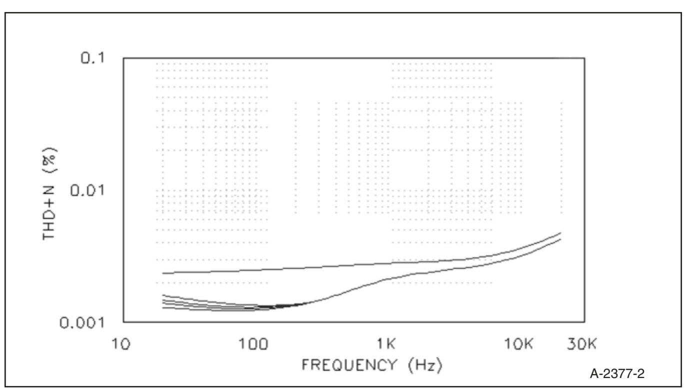


FIGURE 2: THD+N versus frequency.

due to skin effect. The skin effect is essentially the same for a conventionally stranded wire as it is for a solid conductor of the same material and the same net cross-sectional area. The skin depth is the depth below the conductor surface where the current density has decreased by 1/e, and for copper at 20kHz it is about 0.5mm (0.02″). As Bob Pease of National Semiconductor pointed out, when you try to measure pure inductance, you may also measure some of the skin effect. Interconnect capacitance tends to cancel inductance, so inductive phase shift measurements can also be confused with skin effect.

The interconnect inductance is also in series between the audio source and its load. The inductive reactance in ohms is equal to 2πfL. The resistance and reactance do not add arithmetically because the reactance lags resistance by –90°, so the series impedance is the root-sum-square of the two.

Capacitive reactance in ohms is equal to 1/(2πfC). Capacitive reactance leads resistance by +90°. The shunt capacitance will tend to cancel some of the series inductance as frequency increases. Unlike the speaker cables, the capacitance of audio interconnects can affect the high-frequency signal response in the higher impedance circuits associated with line-level signals.

The complex sum of these L, R, and C components determines the equivalent series impedance of an interconnect. The phase shift introduced by an interconnect is proportional to its complex series impedance.

Given the relatively high impedance of line-level audio circuits, an interconnect must provide effective shielding against electromagnetic interference (EMI) as well as convey the audio signal as accurately as possible. Depending on the insulating material, interconnects can also introduce low levels of noise into the signal if the insulator changes its electric field attenuation due to vibration and handling.

INTERCONNECT CONSTRUCTION DETAILS

The interconnects provided for testing comprise two types. One type uses woven wires, while the others are conventional coaxial cables. I also added a sample of the tin-plated interconnect furnished with 1980's era consumer audio equipment. Test sample lengths varied from 0.75 meter to 1.85 meters. *Photo 1* shows the four test samples, clockwise from lower left: Supra EFF-IX, Kimber Hero, Solutions, and tin-plated generic.

Figure 1 (left) shows the woven construction of the Kimber Hero. It consists of four individual VariStrand™ PTFE (Teflon™) insulated wires woven into a tight braid. PTFE is the most stable of the dielectric materials, with a low dielectric constant of 2.1. Two parallel wires carry the signal, and the other two are connected to the ground shells. This is an evolution of the original Kimber PBJ three-wire weave, with an added outer jacket of braided plastic. The outer jacket affords additional protection for the conductors and gives the clamp screw of the WBT connector a surface to bite into. The top of the

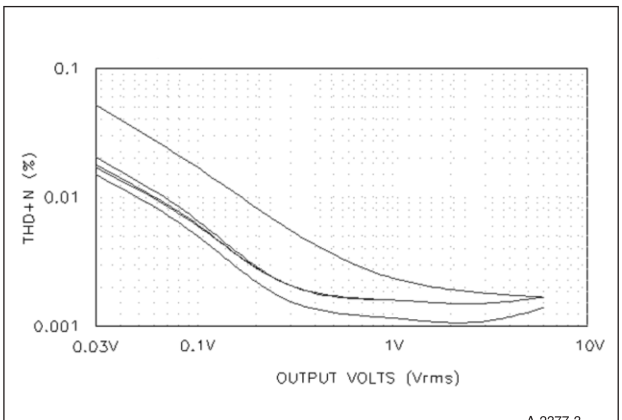


FIGURE 3: THD+N versus voltage.

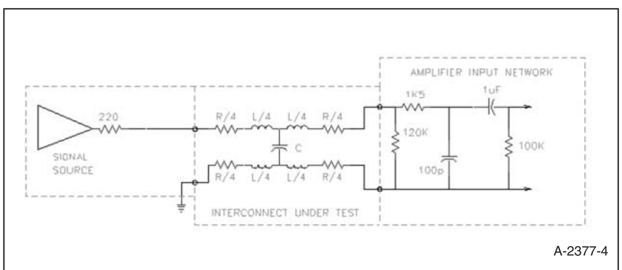


FIGURE 4: Line-level system SPICE model.

gold-plated WBT-0144 plug is tapered such that when you turn the connector barrel counterclockwise (CCW), it draws the eight grounding leaves of the plug tight against the mating jack outer shell, much like drill chuck jaws tighten against a drill bit.

The Supra EFF-IX (*Fig. 1*, right) consists of two tube-shaped conductor layers, each wrapped around a vinyl core. Each conductor is covered by polyethylene (PET) insulation and an individual aluminum foil shield. PET has a fairly stable dielectric constant of about 3.2, which decreases about 3% at 1MHz. Polyester film is slightly hygroscopic—it absorbs water by a factor of 0.2% in the presence of high humidity. A tough plastic outer jacket covers the entire assembly.

The foil shield screen wires are connected to ground only at one end, so signal currents do not flow through the shields. The grounded end is marked by a yellow arrow and signifies the signal source side of the interconnect. The terminations are gold-plated Supra PPX plugs. The four grounding leaves of the PPX plugs produced quite a tight fit on most phono jacks.

The Solutions interconnect was a closeout special at Parts Express. These long interconnects appear to be aimed at the home theater and car audio markets. The conductors are conventional coaxial cables joined in a PVC outer jacket. Molded gold-plated connectors are used on each end. The four grounding leaves of the plugs were a bit loose on most phono jacks.

The "tin-plated" generic 1980's interconnect uses crimped tin-plated plugs with a wide gap between each of its four flared grounding leaves. The plug does not fit fully over the shell of any of the phono jacks I tried, so the shielding is discontinuous at the four gaps at each jack. The joined pair of coax connectors use plain PVC insulation. Ordinary PVC has a rather high dielectric coefficient of 4.7, and its dielectric coefficient decreases by 46% at 1MHz.

MEASUREMENTS

The interconnect samples vary from 0.75 meter to 1.85 meters. I measured the total capacitance, inductance, and resistance of each sample with two types of digital LCR meters. The HP 4261A performs measurements at 120Hz and 1kHz. The AADE L/C IIB uses a resonance measurement method with a variable frequency signal.

I also measured the DC leakage current of the dielectric using a Keithley 480 picoammeter as a measure of insulation quality. A 12V DC gel cell battery biased the interconnects. I made all measurements with the samples laid out in a straight line.

The HP 4261A also measures capacitance dissipation factor (DF), which is an expression of the power loss in the dielectric. DF is the ratio of the equivalent series resistance (ESR) to the capacitive reactance (Xc), and is a measure of the insulator's ability to store energy at a given frequency. A lower dissipation factor is better.

The capacitance is measured between the center conductor and the shell of the plug. Inductance is measured by short-circuiting one end of the interconnect and connecting the LCR meter to the plug on the open end. Resistance is measured the same way, so the L and R per foot is the sum of both the center and ground conductors.

I also took the 1.85 meter length of the Solutions interconnect and mea-

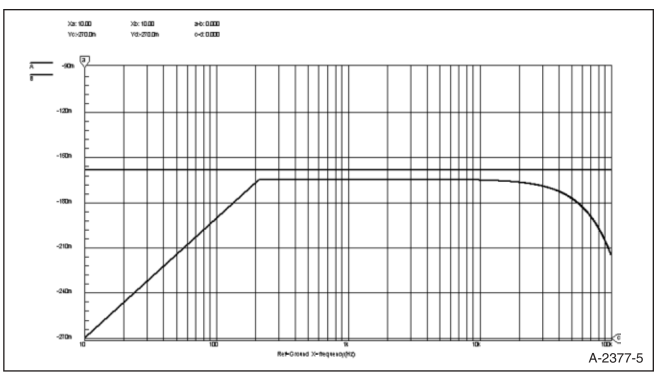


FIGURE 5: Frequency response, 12k resistor versus Kimber Hero after amplifier input network.

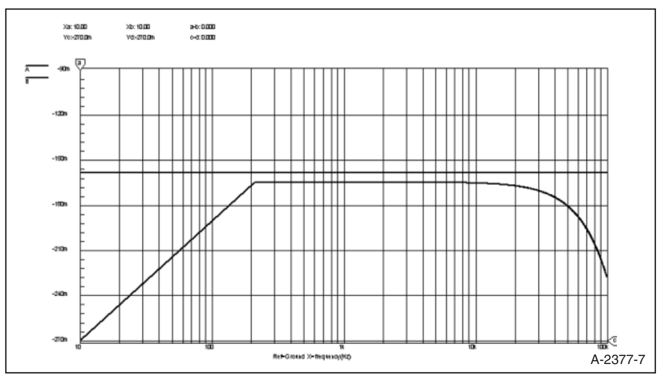


FIGURE 7: Frequency response, 12k resistor versus Solutions after amplifier input network.

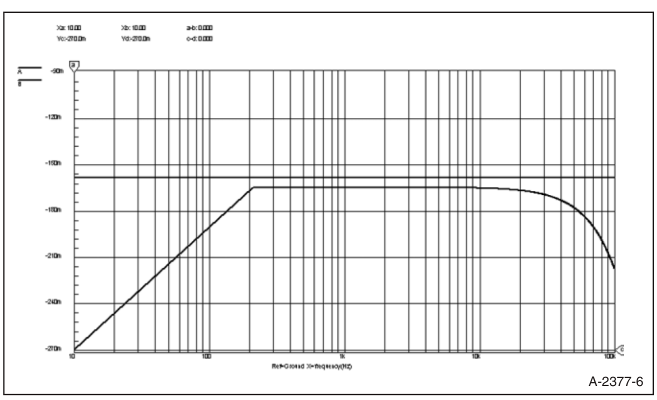


FIGURE 6: Frequency response, 12k resistor versus supra EFF-IX after amplifier input network.

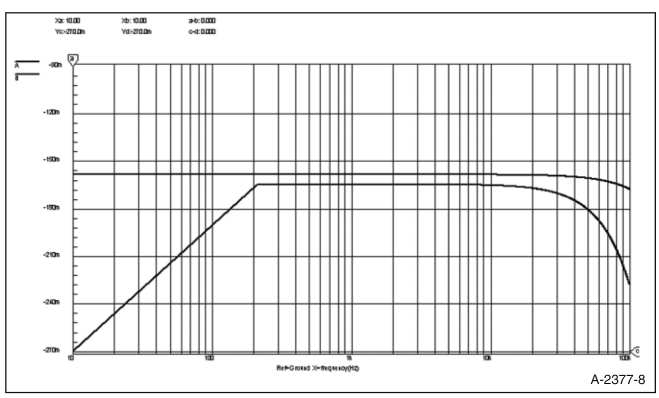


FIGURE 8: Frequency response, 12k resistor versus tin-plated after amplifier input network.

sured its parameters first as a straight run of interconnect, and then coiling its full length in 5″ diameter loops. The total capacitance increased by 4%, while the inductance remained the same.

Using the measured data, I calculated the C, L, and R per foot, the characteristic impedance ($Z_0=\sqrt{(L/C)}$, independent of length), the insulation resistance (Rp), and the interconnect resonant frequency. The data is summarized in *Table 1*.

The Kimber and the Supra interconnects came with characteristic data/meter. I compare this specification data with my measured data in *Table 2*, adjusting for the test sample length.

TESTS

I measured the THD+N vs. frequency for a 1V rms signal directly across a non-inductive 10k Caddock power film resistor at the end of each interconnect. I also measured a 12″ length of RG58/U test cable terminated with BNC connectors. This cable measured 38pF, 0.002%DF, 97nH, and 39mΩ. The source impedance of my distortion test set is 600Ω.

Figure 2 shows the THD+N vs. frequency for each interconnect with the test set 80kHz low-pass filter engaged. Top to bottom at 20Hz, the samples are Kimber, tin-plated, Supra, Solutions, and RG58/U. I consider the RG58/U data to represent the residual noise level of the distortion test set oscillator.

The big surprise is the Kimber Hero, which had higher THD+N levels than the others. After some investigation of the residual distortion products with my analog scope connected to the monitor output of the distortion test set (after the THD test set notch filter), I found a distinct 150kHz signal hidden among the random noise. It seemed to peak a bit in magnitude at each zero crossing of the test sine wave. I could increase the overall magnitude of this signal by wrapping my hand around the outer jacket.

The results were the same for both samples of the Kimber, and reversing the interconnect made no difference. When I engaged the steep 22kHz LP filter I use for testing CD players, the Kimber distortion level decreased below those of the other interconnects, to just above that of the RG58/U cable. I repeated the testing the next day, with identical results. Increasing the load resistance to 100k made no difference.

The residual distortion product for the other interconnects consisted of random noise. I could easily increase the noise in the tin-plated interconnect by bringing my hand near either plug adjacent to the gap in the grounding leaf. The random noise level also increased when I engaged the jack shell with a minimum amount of force. I had quite a few of these interconnects on hand, and substituting a different sample made little difference in the noise level. I could also generate noise spikes by moving the coax where it entered each plug.

TABLE 1
MEASURED AND CALCULATED PARAMETERS

INTERCONNECT	KIMBER HERO	SUPRA EFF-1X	SOLUTIONS	TIN-PLATED	UNITS
Measured C	78	111	625*	170	Pf
Diss Factor	0.003	0.004	0.080	0.070	%
Measured L	330	401**	592	545	nH
Measured R	39	75	397	368***	mΩ
Length	1	0.75	1.85	1.5	M
Insulation	PTFE	PE	PVC	PVC	
Leakage current	0.1	0.2	3.5	1.0	pA
C per M	78.0	148.0	337.8	113.3	pf/M
L per M	330	535	320	363	nH/M
R per M	39	100	215	245	mΩ/M
Characteristic Z_0	65.0	60.1	30.8	56.6	Ω
Calc Rp	680.5	358.6	3.2	13.4	MΩ
Resonant Frequency	99	75	26	52	MHz

Notes:
* Solutions capacitance measured 625pF left, 593pF right
** Supra inductance measured 412nH at the arrow end
*** Tin-plated resistance measured 325mΩ left, 368mΩ right

Next, I measured the THD+N versus the voltage at 20Hz across the 10k load resistor. *Figure 3* shows the data, top to bottom, at 0.03V: Kimber, tin-plated, Supra, Solutions, and RG58/U. The Kimber again had the highest THD+N levels, and it remained that way until the test signal reached 3V rms, where the audio signal was finally able to overcome the 150kHz. The noisy tin-plated interconnect was the next highest, although its THD+N dropped rapidly above 0.3V rms.

Next, I applied a 1V peak-peak square wave with a frequency of 100kHz, and measured the rise time in nanoseconds across the Caddock 10k power film resistor with my analog oscilloscope. The results for each interconnect are shown in *Table 3*.

The Kimber Hero square wave response was fast and clean, without any peaking or ringing that would indicate the presence of the 150kHz signal I observed during the THD+N testing.

To further investigate the possibility of RF pickup of the Kimber interconnect, the only one without an outer shield, I used my FM alignment generator to drive a 98MHz test signal through each interconnect into a 50Ω dummy RF load. I set the output for 27,400µV (100dBf, or 100dB above 1 femtowatt).

I measured the conducted RFI with an Electro-Metrics PCL-11 RF current probe connected to a Boonton 42B microwattmeter. None of the four interconnects showed any sign of RF leakage. When I fed the RF signal through a length of 16-gauge zip cord, I measured an RF leakage of 2.5nW (64dBf). That unmodulated signal strength is sufficient to drive any FM tuner into full quieting.

As with the speaker cable tests, in order to evaluate frequency response and phase shift for equal length interconnects, I used SPICE simulations.

SPICE SIMULATIONS

Using the measured parameters, I modeled each interconnect as part of a SPICE model line-level audio system in order to determine its effect on magnitude and phase. *Figure 4* shows the line-level system model I decided on. I used a source impedance of 220Ω, which is at the upper end of a solid-state preamp, yet close to the cathode-follower output impedance of a tube preamp. The line load is the input network used in the NAD-214 power amplifier. The interconnect models are 1-meter length T-section equivalents based on my actual measurements.

I modeled the DC resistance portion of the cables as equal R/4 segments. This is not a totally accurate representation of the Solutions and tin-plated interconnects. About 60% of the total resistance is in the center conductor of the Solutions interconnect. For the tin-plated, this jumps to 75%. The Kimber and Supra both had the ideal 50/50 resistance distribution.

As with the speaker cable simula-

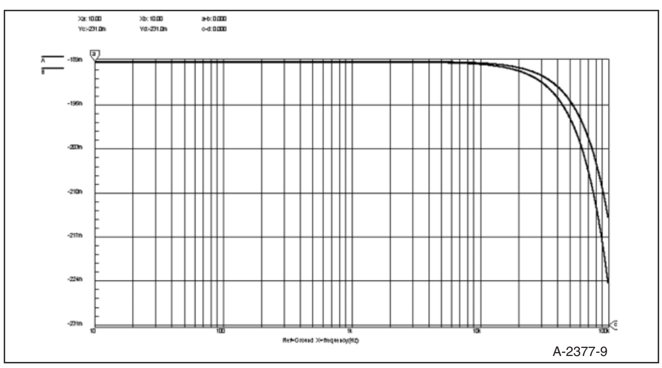


FIGURE 9: Frequency response, 12k resistor versus amplifier input network, "worst-case" interconnect.

TABLE 2
MANUFACTURER'S SPECIFICATIONS VERSUS MEASUREMENTS

PARAMETER	KIMBER SPEC	MEASURED	SUPRA SPEC	MEASURED
C	76.5pF	78pF	58pF (77pF/M)	111pF
L*	401nH	330nH		
R	33mΩ	39mΩ		
Z_0			75Ω	60Ω

(* Kimber specifies both series and parallel inductance. I have listed the series inductance here since that is the way in which the inductance of a cable manifests itself with respect to the resistance. If the inductance were to be in parallel, the DC resistance measurement would be zero.)

tions, these line-level simulations should be considered comparative rather than absolute.

The signal source model was set for an open-circuit output voltage of 1V rms at 1kHz. I ran simulations with a 12k resistor directly across the signal source, and with each interconnect in series with the power amplifier input network load. I used two identical amplifier channels in the simulations to avoid interaction between the 12k resistor and the interconnect/amplifier input network models. The vertical scale is in dB relative to 1V rms.

There was no way for me to calculate the skin-effect resistance for the various interconnect configurations. I concluded it was probably irrelevant, because the milliohm resistances of the interconnects are negligible compared with the source and load impedances.

Figures 5–8 show the frequency response for each interconnect. The upper line is the response across the 12k resistor, and the lower curve is the response across the output side of the amplifier input network. The response across the resistor is flat in both magnitude and phase shift from 10Hz to 100kHz. The drop in response at the high and low ends of the lower curve is due to the input network RC components. In order to differentiate between the interconnects, I recorded the simulation attenuation and phase shift at 50kHz.

Figure 5 shows the response model for the 1-meter Kimber Hero. At 50kHz, the attenuation after the input network is –0.177dB and the phase shift is –3.4°.

Figure 6 shows the response model for the 1-meter Supra EFF-IX. At 50kHz, the attenuation after the input network is –0.178dB and the phase shift is –3.7°.

Figure 7 shows the response model for the 1-meter Solutions. At 50kHz, the attenuation after the input network is –0.181dB and the phase shift is –4.5°.

Figure 8 shows the response model for the 1-meter generic tin-plated interconnect. At 50kHz, the attenuation after the input network is –0.178dB and the phase shift is –3.6°.

INPUT NETWORK

I also modeled a "worst-case" interconnect consisting of the 342pF Solutions capacitance, the 133.7nH L/4 of the Supra, and the 61.33mΩ R/4 of the tin-plated interconnect. At 50kHz, the attenuation after the input network is –0.182dB and the phase shift is –4.6°.

The SPICE simulation magnitude and phase shift data for the line-level system models are summarized in *Table 4*.

Finally, to isolate the effect of the interconnect alone, I made SPICE models that drove the fixed 12k resistor and the amplifier input network through the worst-case interconnect. I used independent voltage sources, 220Ω resistors, and interconnect models so there would be no interaction between the two responses. The frequency response of these two simulations, at the far end of the worst-case interconnects, are shown in *Fig. 9*. At 50kHz, the attenuation at the 12k resistor is –0.160dB and the phase shift is –1.08°.

CONCLUSIONS

I am at a loss to explain the 150kHz signal seen with the Kimber Hero during the THD+N testing. I did put both the Kimber and Supra models in my system, and found no ill effects from a brief listening test. Despite the fact that the Kimber has no dedicated outer shield, I don't believe this is RF pickup. Twisted-pair wires do an excellent job of rejecting electric fields. I don't know of any commercial or amateur radio source that broadcasts at that frequency. Since the magnitude of the 150kHz signal rose slightly at the zero crossings of the test sine wave, it may be some sort of very low-level parasitic resonance in my test setup. This 150kHz is not distortion in the strict sense; it falls into the noise portion of THD+N. Most good amplifiers and preamps would roll off their response below 100kHz, and I don't believe it would even be present outside the test environment.

In any case, it would be of no audible consequence in an audio system. Removing the effect of this 150kHz signal with the steep 22kHz LP filter puts the Kimber Hero noise level measurement slightly above the short RG58/U cable with the same filter.

As with the speaker cables tested earlier, there are again measurable LRC differences between these four interconnects. In all cases, the amplifier input network RC impedances dominate the response across the audio band. The difference in attenuation at 50kHz between the four interconnects is only 0.004dB despite the data from *Fig. 9* that shows the "worst-case" interconnect accounts for about 0.160dB of the modeled system attenuation. Similarly, the difference in phase shift at 50kHz between the four interconnects is 1.1°, with the data from *Fig. 9* showing that the "worst-case" interconnect accounts for about 1.03° of the modeled system phase shift.

CHARACTERISTIC IMPEDANCE

When you think of a transmission line, you naturally think of coax cable. All but the Kimber Hero is composed of some form of coax. As with the speaker cables, even the longest practical interconnects are an infinitesimal fraction of the audio signal wavelength.

The wavelength of a 20kHz audio signal, even after accounting for the drop in signal velocity propagation factor due to the interconnect configuration, is over 39 kilometers. The wavelength of the 50kHz signal I used for taking attenuation and phase shift SPICE data is almost 16 kilometers. Irrespective of the extremely small l/λ ratio, any attempt to match the characteristic impedance of any interconnect from the low output impedance of audio sources to the high input impedances of line-level audio components is doomed to failure.

The source impedance, Rs, can vary from tens of ohms for discrete solid-state line stages to as much as 5k for a shunt-regulated push-pull (SRPP) tube stage. The input impedance, Rin, can vary from a couple thousand ohms for the Pass Zen MOSFET amplifiers to 100k or more for tube amplifiers. Indeed, the characteristic impedances of all the interconnects I tested is 65Ω or less.

There are no reflections from a transmission line when its reflection coefficient, ρ, is zero. Reflections are at a maximum when ρ is ±1. Again ignoring the extremely small l/λ ratio in an audio interconnect, the reflection coefficient at the source end is defined by

$$\rho = (Z_0 - Rs)/(Z_0 + Rs)$$

TABLE 3
100KHZ SQUARE WAVE RISE TIME

INTERCONNECT	KIMBER HERO	SUPRA EFF-1X	SOLUTIONS	TIN-PLATED	RG58/U
Rise Time, nsec	40	60	90	70	20

TABLE 4
SPICE SIMULATION RESULTS

INTERCONNECT	KIMBER HERO	SUPRA EFF-1X	SOLUTIONS	TIN-PLATED	"WORST-CASE"
Attenuation, 50kHz	–0.177dB	–0.178dB	–0.181dB	–0.178dB	–0.182dB
Phase shift, 50kHz	–3.4°	–3.7°	–4.5°	–3.6°	–4.6°

The source-side reflection coefficient for my 220Ω SPICE model into a 65Ω interconnect is a high –0.54.

The reflection coefficient at the termination end (the input to the line-level load) is defined by

$$\rho = (Z_0 - R_T)/(Z_0 + R_T)$$

The termination-side reflection coefficient for a 65Ω interconnect into my 12kΩ SPICE model is –0.99.

The best interconnect is no interconnect at all. Integrated amplifiers have an advantage here for the connection from preamp to power amplifier. When an interconnect is necessary, short lengths of low capacitance cables will be the most accurate. As for materials, the highly stable Teflon (Kimber) and PET (Supra) insulation should give the best electrical performance.

Based on the noise content of the distortion residual signals, I would avoid all tin-plated connectors. Gold plating precludes any corrosion problems and the unpredictable audio consequences of the oxide film buildup that forms on tin plating.

Manufacturer's response:

I cannot explain the capacitance deviation for our Supra EFF-IX. However, our specifications refer to the cable itself, without connectors. We do not specify the capacitance of terminated cables in our product catalog.

The EFF-I design is for minimizing the skin-effect, as you say, but not for the way you approach the effect of it. You say that it is probably irrelevant owing to its small miliohm resistances. Yes, there is an efficiency loss problem, only in RF applications, whereas for audio frequencies there is not, but another problem; the dynamic *influence of the skin-effect. Music is not comparable to single RF signal, but is a complex signal of different frequencies and amplitudes of current and voltage. The difference in skin depth for 20Hz and 20kHz is rather big. As shown in our catalog and on our website, the skin depth for 20Hz is 14.7mm whereas for 20kHz it is only 0.46mm. (Please convert if you prefer inches) Formula is also available from our website www.jenving.se.*

This means that the resistance varies dynamically with the signal content in real time (i.e., the current pumping in and out inside the conductor). That is audible. *We are sensitive to dynamic changes and this is like changing cable, continuously.*

Please mention our US distributor: Tonian Laboratories, www.tonianlabs.com (They frequently advertise Supra in your magazine).

Tommy Jenving
Jenving Technology AB

CONTACT INFO

Kimber Kable
2752 South 1900 West
Ogden, UT 84401
01-625-5530
e-mail: info@kimber.com
Kimber Hero $150/pair
Terminations: WBT-0144 plugs

Supra Cables/Jenving Technology AB
Bastebacka 112-113
S-459 91 Ljungskile, Sweden
+46 (0)522-698990
e-mail: supra@jenving.se

LISTENING TESTS KIMBER AND SUPRA INTERCONNECTS

By Muse Kastanovich

Under review are the Kimber Hero ($150) and Supra EFF-IXLR ($141) balanced interconnects (*Photos 1 and 2*), Kimber Hero ($150), and Supra EFF-IX ($118) unbalanced interconnects (*Photo 3*). All Kimber interconnects are 1m long, and the Supras measure ¾ meter.

LISTENING TEST # 1

The first listening I did was with the balanced interconnects in my friend Randal's system. He has a pair of B&W 800 speakers, Krell KSA 200S power amp, Krell KBL preamp, and a Denon S10 CD player. I used the interconnects between the CD player and preamp. On hand for comparison was the pair of Audiotruth Diamond X3 pure silver balanced interconnects that normally inhabit the system.

Listening to "Safe Journey" by Steve Tibbetts (ECM 1270), and switching from the Audiotruth, I noted the Supra EFF-IXLR had a touch less top octave information. The drums sounded a bit more like they were made out of plastic, and there was less sustain and natural ringing in the treble. The midrange sounded good and rich with the Supra, however. It definitely had softer-sounding edges to the transients, and was less dynamic sounding than the Audiotruth.

PHOTO 1: Kimber Hero balanced.

The Kimber Hero balanced had a more agile presentation of delicate sounds than the Supra. The Hero was a bit more articulate, and had more air in the top octave. At the same time, it was punchier and more dramatic. I found the Hero to have a well-balanced sound from top to bottom.

Now listening to the HDCD "Time Capsule" by the B-52s (Reprise 9 46920-2), I started out with the Hero as the reference. Switching to the Audiotruth, I found it had more top octave, but the midrange was slightly recessed and not as rich sounding. It sounded a bit more grainy, which I did not attribute to revealing more recording flaws. The bass was also not as tight with the Audiotruth.

Again with the Hero as the reference, the Supra seemed to have a little bit of a thin midrange. Its treble was a bit ragged, and the bass was thinner than with the Hero. Somehow, it seemed noisier, perhaps because the music sounded more heavily processed and artificial. Also, voices sounded kind of muffled, as though I were listening to them through a curtain.

LISTENING TEST #2

I switched to my own system for the remainder of the auditioning. This consists of a Rotel RDD-980 CD transport and Assemblage DAC-3 D/A processor connected with Sound & Video Digiflex Plus BNC cable. Amplification was provided by my home-built STAMINA single-ended MOSFET integrated monoblock amps (based on the Pass Zen/Bride of Zen). Speakers were my modified pair of B&W Matrix 804s.

The first comparison I made on my own system was with the Supra EFF-IX after familiarizing myself with the sound from the unbalanced Kimber Hero RCA, using the HDCD "Triple Quartet" by Steve Reich (Nonesuch 79546-2). The Supra was softer and more muddled. The top octave sounded a bit recessed, and the flute became lost in the mix. The bass was more nondescript than with the Hero, and the marimbas sounded a touch muffled.

Now I started with the JPS Labs Superconductor interconnect, which normally serves as the D/A amplifier interconnect in my system. The Superconductor has a very involving sound and the perfect frequency balance to complement the rest of my system.

I made the switch to the Supra and found it rounded off the sound a bit, kind of like a classic tube amp. The marimbas were a bit too metallic and thin, and the strings sounded a bit too thin, as well. Some decay of the individual notes seemed to be missing. Also, the soundstage was more vague, with slightly fuzzy instrumental positioning.

PHOTO 2: Supra EFF-IXLR balanced.

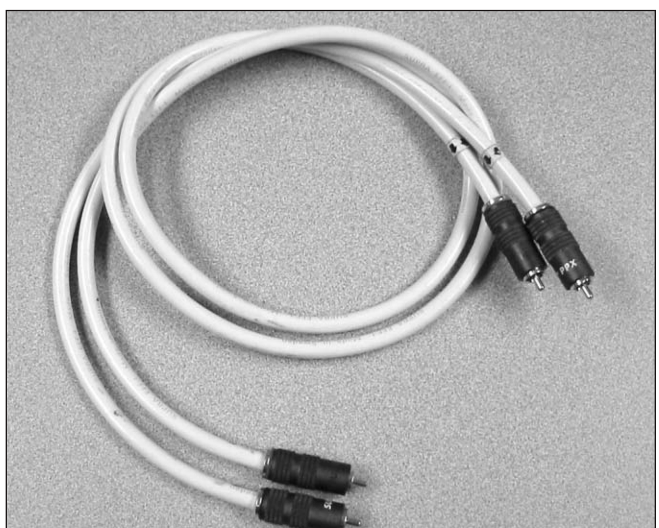


PHOTO 3: Supra unbalanced.

The dynamics seemed a little soft in comparison with the JPS Labs.

Next up for reference was the hybrid silver/copper Silver Streak interconnect by Kimber. A bit more expensive than the Hero ($240 instead of $150), it is, nevertheless, one of the most affordable pure-silver interconnects on the market, and a good all-around performer. Listening to one of my favorite test pieces for clean studio-rock, "Tambu" by Toto (Legacy JK 64957), I detected that the Hero had a little less top octave air, and that transients were slightly softer and slower. The imaging was not as pinpoint-precise, and perceived outlines of instruments were fuzzier. Also, the soundstage was not quite as wide.

I should emphasize that sonic differences between these two interconnects from Kimber were less than those between any others I listened to. There is a distinct familial resemblance, and I was very much liking what I was hearing from both the Hero and the Silver Streak.

I also briefly switched back and forth between the Hero and an old pair of Kimber PBJ interconnects ($75). To cut to the chase, I preferred the sound of the Hero in just about every area of reproduction. The more expensive model had better dynamics, frequency extension, transparency, and so on. The PBJ has always been a good value as an entry-level interconnect, but be warned that once you get used to what the Hero can deliver, it's difficult to settle for less.

When comparing the Supra to the Silver Streak, I believed there was less glow around individual images. The Supra's soundstage was narrower and shallower, and the tone was a little thin. The Supra had strong treble, but slightly recessed midrange, bass, and top octave extension. Also, the dynamics sounded weaker. The vocals sounded more processed and artificial than they did with the Silver Streak.

WRAP-UP

There you have it, eight different interconnects, mixed and matched. The situation is a bit simpler because the balanced Supra and Kimber interconnects are both virtually identical balanced versions of their unbalanced counterparts. I listened to both versions in the interest of thoroughness, but the sonic qualities of balanced and unbalanced versions are nearly the same in both cases.

I was very impressed with the Kimber Hero. The fact that it can compete quite well sonically with some more expensive silver cables makes it a must-audition if you are shopping in this price range. In fact, I preferred it in some ways to the JPS Labs Superconductor, which has been my reference interconnect for several years. The Kimber Hero has now replaced the Superconductor in my main system, and will serve permanent duty there because of its high transparency, powerful dynamics, and beneficial frequency balance.

The Supra EFF-IX, on the other hand, did not end up on the favorable end of any of the comparisons I made. As always, your mileage may vary, and it might sound better in other systems. I cannot recommend it since it is possible to get more transparency and smoother response from other choices at this price. ❖

Xpress Mail

UNORTHODOX TWO-WAY

I have been reading Jon Mark Hancock's series of articles about the design and construction of an unorthodox 2-way (Sept., Oct., Nov. '03 *aX*). It was very exciting to find a relatively easy construction project that reflected considerable thought, design, and attention to producing a speaker that rivals commercial products.

It is obvious that a lot of attention has been paid to address the components in this design with regards to their quality and characteristics of sound. While preparing to assemble these components for construction, I discovered a surprising issue. Although the drivers and cabinet are well within a reasonable price range for a DIY project, the crossovers cost approximately $400 assembled from parts distributors.

Is this due to the characteristics of the tweeter and/or its relationship to the system? Would choosing a different and possibly a more expensive tweeter such as the Scan-Speak D9500 simplify the crossover? Not having heard this design, I would be reluctant to build it due to the cost of the crossover in relation to the drivers.

The last issue I would like to address not only relates to this article but to most other DIY projects. Only about 2% of the article addresses the quality and characteristics of the sound of the unit. You usually get "they sound great" and a few other comments.

The cost of this project is $900 or so for parts plus a few additional miscellaneous items. If I knew this design was comparable to commercial products such as Harbeth, Spendor, Reference 3a, and so on, I would build it without question.

I look forward to the future projects you mentioned that are still on the drawing table.

John Czosnyka
estes1300jc@yahoo.com

Jon Mark Hancock responds:

Thanks for your interesting letter and kind comments. The articles as presented go into a lot more than just the details necessary to build the speakers, because I hoped to convey the thinking and philosophy behind the design choices I made, as well as the proper setup and how the design takes that into consideration (in the discussion on boundary response and room placement in the first part). As well, I included a fair number of detailed measurements, particularly some I believe are quite relevant to evaluating sound quality. The off-axis response plots are very relevant in that regard. There are many more that could have been done, but as it was the article was longer than initially envisioned for a "simple" two-way dynamic system.

PHOTO 2: Another speaker design (MTM) from author Hancock.

Regarding the cost of the components, I offer a few comments. First, I acknowledge that the crossover is a significant part of the total cost. It is quite critical to the performance of the system, as it allows (in conjunction with the driver characteristics) a lower crossover frequency, and a superior (in my opinion) trade-off for directivity and bass extension.

PHOTO 1: Two-way design by author Hancock.

As I mentioned in the article, a single 8″ woofer will perform similarly to two 6½″ or 7″ woofers, but at only a little over half the cost. The drawback to an 8″ driver is the directivity with increasing frequency, which mandates a lower crossover point. Actually, most systems with 6½″ or 7″ midwoofers are crossed over too high with regards to driver directivity.

The most telling evidence of the benefit (besides how it sounds!) is the provided plot for frequency response starting on-axis and going to 45° off-axis. Note how there is no flare, notch, or other irregularities in the range from 1kHz to 3kHz in the crossover region, generally the Achilles heel of more conventional designs, including well-regarded commercial designs published in mainstream magazines such as Stereophile.

My only other comment regarding the "sound" is that these are voiced (with the crossover in the article) with a full baffle step compensation, and intended to be set up in a listening position well out from the wall, as described in Part 1. If you plan or need a different positioning, contact me and I can give you some alternative crossover designs. These are not voiced to be "bright" or for-

ward; they have smooth but detailed character with good image depth and ease of long-term listening. Voicing is similar to Avalons; you can find reviews and measurements online, which will give you some further hints.

The crossover is only slightly more complex than a similarly implemented Linkwitz-Riley 4th order, a not unusual configuration for a two-way system; while it gives substantial benefits in optimizing the driver power handling and system behavior, due to the high slopes achieved in the first 48dB of attenuation, and the notch which is tuned to the LF driver peak.

The crossover's overall expense can vary depending on the specific component choices; it is possible to reduce the cost by using parts akin to the surplus GE polypropylene caps from Madisound (which are very good quality), or by winding your own inductors, as I usually do. In this case, I opted to use and specify good quality "off-the-shelf" components which would be easy to acquire from multiple sources, as I hoped that would make things easier for the average constructor.

It's interesting that you mention the idea of using a more expensive tweeter as an alternative to lower the cost. I chose the Vifa XT25 because it offers unusually good performance for the money (in an effort to keep the driver cost reasonable), especially a low resonance frequency and flat, extended response. But the "reference design" for this configuration uses a Scan-Speak SS9800, which has a nearly identical impedance curve, also very flat response, and nearly identical efficiency. It offers benefits of the very low distortion SD1 motor and lower overall distortion at high SPL. It's essentially a "drop in" upgrade, but at nearly $200 apiece, it ups the ante substantially. I use them in my own systems.

Another tweeter I've used with similar characteristics is the Hales Reference; Photo 1 shows another system based on this fundamental 8″ two-way approach (with another nod of my hat to my old friend Charles Hansen. This is probably the "ultimate" expression of this design; crossover topology is still basically the same, though some component values are tweaked, obviously. Similar commercial speakers (Avalon) sell in the $10K plus range.). By the way, another option to lower the price tag is the HiVi M8n, a stamped frame version of the woofer, which is ~40% lower in cost, but seems to perform the same in my own tests. The magnetic system and diaphragm appear to be the same.

Your last question is interesting—and it's why I think audioXpress has independent reviews on some of the projects offered by quasi-commercial sources, such as the Seas Thor project by Joe D'Appolito. Actually, I felt a little awkward about the amount of "coverage" I gave to describing the performance, as that's akin to reviewing one's own creation, hardly something it's easy to be objective about! Which is why I also mentioned my friend with the large ESL and ribbon system who likes them quite well, too—enough to build a set for himself.

To date there have been quite a few of these built, considering this isn't any kind of commercial endeavor, including sets in an MTM configuration with a suitably modified crossover (Photo 2). It has been A-B'd with some other somewhat well-known DIY systems, such as the Usher 7″ two-way, and compared quite favorably in musical balance, LF extension, and imaging. But these are decisions that are quite personal, and ideally you need to make them for yourself.

While measurements don't tell the whole story about a speaker, they can give you much useful information about their potential neutrality and accuracy. Possibly you saw the review of the Reference 3A speaker in Stereophile a few months back, one of the commercial speakers you mention in your letter. While the subjective review in the article waxed fairly eloquent over his feelings about how this speaker sounded, John Atkinson identified a number of problems in the measured performance, including a pronounced resonance in the midwoofer around 800Hz–1kHz, which was very visible in a major bump on the axial frequency response, a significant glitch in the impedance curve, and a very pronounced energy storage ridge in the waterfall plots.

The on-axis response through the crossover region was not particularly smooth, and became considerably worse as you moved off-axis to 30° or more. These are all factors I can readily hear in a speaker, when present, and I'm guessing John Atkinson can, too, as he commented on them and wondered why the other reviewer didn't make note of them. But then, I guess, to the other reviewer, they "sounded good," at least compared with what he was used to. Whether they would to you or not, I can't predict.

I understand your reluctance to make this large dollar commitment to something you haven't heard, or haven't seen reviewed by someone whose opinions you understand and trust. If you're approaching DIY from primarily a dollars and cents perspective, that's pret-

ty sensible. In that case, I'd be inclined to recommend the "safe" route of buying a unit you're familiar with commercially, though I don't think you can equal the performance for the cost of materials—in my case, I spent a little over $700 on parts altogether.

This is why I did mention my equipment used, as well as some recordings, to give you an idea of where I'm coming from, and what kind of associated gear is recommended. The results will still be decent with just an HT receiver and a low-cost DVD player, for example, but you'd be missing out on what might be achieved by at least having some good separates, such as the Aragon and Ayre gear my MkIVs are hooked up to at home, now. All elements influence the sound, including your room, setup, and cables.

The danger of DIY is that once you get caught up in it and experience its indirect rewards (personal satisfaction from construction and perhaps personal innovations and tweaks), it seems to take on a life of its own, quite independent from the end result. Painters don't stop painting just because they enjoy their latest creation hanging on the wall!

I hope this has given you some food for thought, and answered your questions adequately.

READY TO SERVO

Thank you very much, Daniel L. Ferguson, for your very informative article on a servo subwoofer via voice coil feedback (Nov. '03 *aX*). This low-cost approach seems doable! I have a pair of subwoofers, each with four 15″ drivers that happen to have dual voice coils. And only one of these voice coils is used on each driver, so I am already ready to servo!

Your article states that 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." I know you are ruling out ported designs with this remark, but I wonder whether this also applies to an open baffle arrangement such as mine. With open baffle, the rear wave is free to come around and cancel out the front wave. As a result, there is a rolloff in delivered sound amplitude as the frequency is lowered, since cancellation effectiveness is a function of wavelength. However, using a parametric equalizer, this effect is counterbalanced to deliver a fairly flat response, which is free of box resonances.

I think that I would still need to use this equalizer to compensate for the open baffle cancellation effect and that the output of this equalizer would then go to the servo system to ensure that the desired response was actually delivered by the drivers. Do you see any problems with this? (I guess I would need to use the feedback from one driver to determine the control signal for all four drivers in the sub.)

Another question I have about the servo approach is that of speed of response. Most articles on servos cite the benefits of low distortion and accuracy, but it seems to me that quickness is also a key issue. I have done A-B comparisons between my current system and a prototype employing another open baffle design, using a subwoofer pair that each had six 12″ drivers. During these tests, it was fairly clear that the 12″ driver system was better due to quickness of the drivers. Will a servo system tighten up the bigger but slower response of the open looped 15″ drivers?

Again, thank you for your contribution. If you have taken your circuit any further to where a PWB has been laid out and/or the circuit assembled, I would be interested in purchasing a pair.

John Daniewicz
Cedar Park, Tex.

Daniel L. Ferguson responds:

There should be no reason a feedback system could not be applied to an open baffle (or dipole) woofer system. The difficulty comes from the fact that these systems are inefficient (as you are obviously aware). Therefore, they require both a very large cone area and high excursion to generate sound pressure levels equivalent to conventional box systems. This translates into multiple large drivers, and therein lies the rub. Each driver in a servo system must have its own dedicated controller and amplifier. Thus, your system would need eight separate servo systems, which most of us would find impractical.

You are correct concerning the continued need for equalization. The open baffle cancellation effects must still be mitigated in order to maintain a flat response. A servo driver in this application would attempt to exactly replicate the response curve imposed by the equalizer. If the equalizer were properly designed, you could expect lower distortion, flatter frequency response, and, yes, *improved transient response. The trade-off is that—in all likelihood—output would be reduced since the drivers* must *be operated within their linear region.*

Regarding improved transient response, I can attest that the servo has a positive effect. I have recently done some square-wave testing that clearly shows this, and the improvement is quite audible.

Thanks for your interest in my article. I have continued to work on the concept and have recently developed a working 12″ model which I hope to have published in audioXpress *in the not-too-distant future.*

SOUND DIFFERENCES

I'd like to thank Charles Hansen and Muse Kastanovich for their comprehensive article "Speaker Cables Review," *aX* May '04. Some comments:

1. Regarding conductor spacing (p. 53): Capacitance varies inversely with distance; inductance increases with distance (the opposites were stated).
2. A 20kHz wavelength (p. 60) in free space is 9.31 miles; in a dielectric of 3.2 constant (PET), it is 5.21 miles. So much for my nitpicking.
3. An inexpensive but good low-impedance cable:

   Paralleling nine lengths of 50Ω RG58A/U RF coax cable results in a low-loss cable with 5.56Ω Z_0. RG58A/U cable has a polyethylene dielectric; dielectric constant = 2.26, dissipation factor = 0.02% at 100MHz. Propagation velocity = 0.665 times light speed. Approximate parasitics are 20.6mΩ, 77.5nH, 31.0pF per foot.

   Nine paralleled lengths of this common and cheap RF cable have the following parasitics per foot: 2.29mΩ, 8.61nH, 279pF. The inductance is lower than that of all cables tested in the article. The resistance is less than that of all except the Goertz cable. The capacitance is higher than that of all except the Goertz. Such a cable assembly is, of course, well-shielded into the MHz range.

   You could use six lengths for a characteristic impedance (Z_0) of 8.33Ω, 12 lengths for 4.17Ω, and so on. Over frequency ranges where the speaker impedance is fairly close to Z_0 and fairly resistive, the cable's reactances appearing at its input are significantly reduced (eliminated by distributed transformation with a perfect match).
4. Distortion

   In Figs. 2 through 6, traces B and C show THD variations at the cable input and output, while driving a speaker. This is *not* cable distortion: note that above 70Hz, distortion with the cable and speaker is lower than with a resistive load (with or without cable, same THD, as pointed out, traces A). Apparently the cable impedance and speaker load are simply modifying the amplifier's distortion spectrum. The THD increase below 50Hz could be due to woofer back-EMF nonlinearities. With the first three cables (*Figs. 2, 3, and 4*), this THD increase is greater in proportion to cable resistance, which is consistent with the speaker being the bass THD source.

   But the Monster cable (*Fig. 5*) and zip cord (*Fig. 6*) don't fit this pattern. However, if these cables were actually the source of the bass distortion, it would have shown up with the 8Ω resistor load; it didn't.

   Some multi-carrier RF transmission systems are very sensitive to IM distortion, so many RF cables have been measured. The best can have distortion levels below −160dBc at 100W, 1GHz; that's one millionth of a percent! But even plain old RG58A/U has a level of about −120dBc; that's 0.0001%, and that's at 1GHz, 50,000 times the highest audio frequency! So the distortion of even the worst speaker cables would appear to be completely inaudible.
5. A Challenge

   I'm very skeptical about claims of cable audibility. Mr. Kastanovich said that, compared with the Goertz cable, bass through the Supra was "rich but a little flabby." This description would be expected with sufficient lowering of damping factor. However, the cable resistance (all that matters in the bass) is only 22.2mΩ higher than with the Supra. With a 4Ω (minimum) speaker and 0.8Ω output amplifier, the maximum attenuation difference between cables is 0.04dB. Damping factor with 4Ω load would be 4.97 with the Goertz and 4.84 with the Supra, a 1.03% difference. Moving one's head ½″ can make a 0.04dB change.

I once compared two tube amps at a very low level (my specific interest) and thought there was a very slight difference. But when I set up an instant-switching A/B test, with the same music segment, repeated often enough to make me aware of exceptional bias and other thought-induced distractions (not to mention millimeter head movements), *there was no difference*!

1. I believe the audiophile community would be well served (possibly at the expense of cable manufacturers) by "conducting" one or both of the following tests:
2. A blind cable comparison using instant switching (with, of course, ultra-low resistance relays).

   Have an assistant install one pair of speaker cables from a variety that the auditioner had previously evaluated, and see whether the auditioner can identify which cable is in place.

Please accept that my skepticism is stated with full respect for the authors. If the proposed blind tests are performed with identifiable results, I will accept that without question. In any case, I thank Mr. Hansen and Mr. Kastanovich for a most comprehensive, detailed, and interesting article.

Dennis Colin
Gilmanton I.W., NH

Charles Hansen responds:

1. *I did indeed use distance while the apparently dyslexic portion of my mind was thinking proximity.*
2. *I should have mentioned the relationship between wavelength and the square root of dielectric constant.*
3. *I don't think I ever heard of using parallel lengths of coax for speaker cables. What an interesting idea!*
4. *Absolutely correct! I should have stated in the conclusions that the variations in distortion with frequency are due to the speaker impedance variations changing the loading on the amplifier, and not "cable distortion."*

I had a real struggle with the results with the Monster cable, despite its 10′ length. I repeated the THD tests for the Monster cable several times over a few days, and still got the same data.

Another correction: In Table 2, the Goertz measured resistance is 1.60mΩ, not 1.60Ω.

Muse Kastanovich responds:

First of all, I must disagree completely with Mr. Colin's assertion that "the distortion of even the worst speaker cables would appear to be completely inaudible." He reminds us that the distortion measured in RG58 coaxial cable at 1GHz is approximately 0.0001%. If this number were similar to the distortions at all audio frequencies, then I would agree that it is so low as to be inaudible.

However, a 1GHz signal is actually easier for a cable to transmit than one at audio frequencies, since it travels exclusively on the surface of the conductor. At audio frequencies electromagnetic signals travel on the surface of conductors, and inside them, and combinations of the two, with varying skin depth. There was an excellent technical article on this phenomenon in Stereophile *a few years back, but I'm unable to reference it right now.*

This is part of the reason different cables do *sound so different in listening tests. Also, speaker cables do not normally enjoy the benefit of a perfectly matched, resistive source impedance, nor a perfectly matched, resistive termination impedance when playing in a stereo system.*

No doubt the RG58 result of 0.0001% distortion came from a test where these impedances were perfectly matched to the cable. Add to that the fact that music is not a single frequency, steady-state sine wave, and you can see that listening results might not have much to do with these types of distortion tests at all.

When I wrote in my original article that the Supra cables had a bass presentation that was "rich but a little flabby," I was referring to the attack and decay of individual bass notes, not just frequency response. The listening statements I make are actually very complex observations of the differences in the feel and impact of musical sounds and qualities over time. For space considerations in audioXpress *I must keep these brief and cannot fully explain each one. Frequency response is only one facet of reproduction, and not even the most important one in my view.*

As for the blind listening tests Mr. Colin suggests, they sound like fun. I would be more than happy to participate in such tests given that a few conditions are met. First, I *would need to be involved in the design of the tests so that a switch box of sufficient quality was used, or a proper one could be custom built. Also the listening room, associated equipment, listening time, volume, and other conditions would have to be appropriate so that differences could be easily heard as they are in my own home on a regular basis. Please note that I would not demand overly expensive or pedigreed components be used. You can achieve a large percentage of the best sound with carefully selected affordable equipment.*

Blind listening of this kind has fallen out of fashion in recent years. There is a good reason for this. The negative results that came out of the Stereo Review *tests of the 80s/90s proposed the laughably preposterous conclusions that there were no audible differences whatsoever between any power amplifiers or CD players!*

I have personally never heard two different components of any kind that did not have musically important differences in their sound qualities. The differences are usually a little smaller for cables, but still audible and often musically important. In fact, I have heard differences from changing a single resistor in an amplifier I was building from one brand to another (exact same resistance value).

I fully acknowledge that not everyone has the same ability to hear differences, nor does everyone care equally about the subtle differences that are to be found. To me these differences are important to the musical experience. When all added together, they produce a large difference in sound quality, and a much more satisfying connection to the original performance that is being reproduced.

CORRECTION

I am from India and a great fan of *audioXpress*. I am writing to point out a serious flaw in one of your articles ("Restoration of Peavey TKO 80 Bass Amp, Part 2," Nov. '03, p. 40). The pin configuration for positive and negative supply of preamp IC NE5532 was incorrectly printed. Such mistakes cannot be tolerated from a magazine which claims to be the "audio technology authority."

Manu Balasree
manu_b1235@asianetindia.com

Charles Hansen responds:

Mr. Balasree is correct concerning the error in my IC power supply pin labels (Figs. 13 and 14, p. 40). Pin 4 is –DC and pin 8 is +DC. While I certainly appreciate his bringing this error to my attention, I hope he doesn't actually live or work in an environment where "mistakes cannot be tolerated."

6SN7 HUM

I recently took an opportunity to investigate why the Russian 6SN7 causes very annoying hum in my DIY 300B single-ended amplifier. I am using the JE Lab Design.

If I use US-made 6SN7 tubes from RCA, Philco, Sylvania, and so on, it's hum-free. I bought two Electro-Harmonix and three Russian-made 6SN7s. They all cause a lot of hum noise.

I have tried to add a 100Ω hum pot at the filament. It helps a little, but it won't cure the problem.

I read many articles and schematics. I noticed that the potential of the filament should not be too large from the grids. Most direct-coupled circuits would provide a +60 –80V DC to maintain the potential between the filament and two grids to a reasonable level.

I added a 20k resistor to the 100k bleeding resistor at the B+ line. This gave me +80V DC at the tap. I have the center tap of the filament supply soldered to it. Wu la la, the hum is gone. And now, I can try out the Russian 6SN7s, which I've had for a long time.

Johnny Tang
kmtang@direct.ca

HELP WANTED

I'm in the process of building Mr. Hamley's power amplifier circuit (*TAA*, 1/88, pp. 51–3). I am updating it with current mirrors and a transistor bias circuit. Could you recommend substitutes for the SK133[5] and SJ48[5] in a TO-3P package?

Might it afford any improvement to use three output transistors in parallel over the indicated pairs?

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

Readers with information on this topic are encouraged to respond directly to the letter writer at the address provided.–Eds. ❖

Hi-Quality MM IC Preamp

from page 13

tridge at a 5cm/sec recording velocity (0dB). This produced a margin of about 73dB, which you could improve by either adding shielding material around the transformer/power supply or moving it farther away from the active circuit board.

The hum level was never an audible problem, even though my transformer is only a bit over 6″ from the main circuit board. Another test involved injecting very low level white noise into the preamplifier and verifying the RIAA playback curve characteristics. Finally, I verified the 44dB gain level at 1kHz.

The big question is how the unit sounds, given its very low cost and simplicity. All of the impressions that I will impart are after the unit had 300 to 400 hours of break-in, and a two- to three-day warm-up for listening. I used a Shure V15 series cartridge in a damped low-mass tonearm on a belt-drive turntable for all listening.

I was very pleasantly surprised by the sonics from the start. The preamplifier quickly proved to be generally smooth and neutral, with a slight leaning toward the warm side. It produced a very nicely defined soundstage that could be very wide and fairly deep when the recording had such characteristics. It also had very good detail resolution and transparency.

I was particularly pleased with how well it reproduced instrumental sounds with convincing body and dimensionality. Delicate high-frequency sounds from harps and triangles seemed to float in the space behind my speakers, with very clean and natural transient attacks and decays. The preamplifier was also easily able to reproduce the many sonic recording differences (and shortfalls) of some of my records.

In sum, I believe that this unit offers a level of performance that is far beyond the price of construction, and that would easily embarrass many commercial solid-state phono preamplifiers that cost much more. If you are interested in getting into vinyl playback, do yourself a favor. Build this low-cost preamplifier and save your hard-earned dollars to purchase a better turntable and cartridge! ❖

Book Review

Katz: Mastering Audio

Reviewed by David Moulton

Mastering Audio *(#BKB88, $39.99), available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail custserv@audioXpress.com.*

Bob Katz is a well-known, dedicated, passionate and highly accomplished recording and mastering engineer, as well as a prolific contributor to a range of Internet listervs, including ProAudio and Sursound. I've followed his career for years now with considerable respect and admiration. Bob's new book, *Mastering Audio: the art and the science* (Focal Press) is a welcome addition to the small library of contemporary books about professional audio.

If you are a student of recordings, a budding recording engineer, or an audiophile who wishes to obtain an inside look into how recordings are made and mastered, this book is essential. It fills a hitherto unpopulated niche dealing with the final creative stage in mastering a modern recording.

Beautifully printed and illustrated, *Mastering Audio* is divided into five parts. The first three cover the meat and potatoes of mastering, including facilities, formats, standards, preparation, the mastering process, signal processing, and the craft of mastering. Part IV is a very brief epilogue, and Part V is an excellent collection of Appendices, covering everything from the effects of radio broadcasting to a small but well-written glossary. There is also a good index.

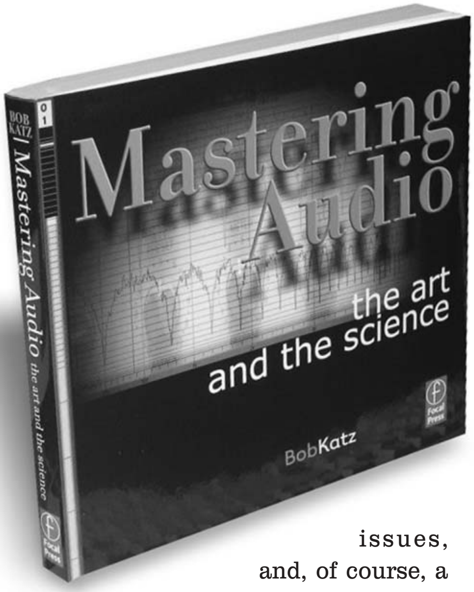
Part I: Preparation begins with the concepts related to mastering, a general philosophy of mastering, a welcome section on documentation, Bob's take on possible DAWs and formats to choose and use, the processes and procedures used in mastering, and the librarian's business of verification, archiving, and making back-ups (if you actively dislike the idea of becoming a librarian, don't even think about becoming a mastering engineer!).

This is followed by an extensive discussion of signal flow in the mastering suite, some excellent ear-training suggestions, and discussions of resolution, dither, levels management, dubbing, and monitoring—worthy and welcome topics, all.

Part II: Mastering Techniques is worth the price of the book all by itself. Beginning with a discussion of album assembly, Bob takes you through editing, levels adjustments, equalization, the use of compressors (should be required reading for everyone who ever makes an audio CD of anything), noise reduction, and other types of signal processing that find their way into the mastering room.

Part III: Advanced Theory and Practice contains plenty of that, but also some material that is, to me, critical basics, not advanced at all. The first of these is a section about calibrated monitor systems. Second is a section on metering, including, of course, Bob's own K-System of metering, which involves three different nominal levels choices. This section includes considerable discussion of the relationship of acoustic sound pressure levels to indicated meter levels and even touches on the growing issue of *metadata*, which is transmitted control data that is going to make our lives really complicated in a couple of years.

A chapter on analog and digital processing takes you through that particular thicket of mythology and hype, including a discussion of some of the measuring tools that are available. Chapter 17 finally addresses some advanced "tricks of the trade," and even detours out of mastering into plain old recording engineering to get at some important issues. There are also sections on sample rates, jitter, word clock



issues, and, of course, a closing chapter on tips for getting out of the trouble we so often find ourselves in, right up to our earlobes!

The remainder of the book, **Part IV: Out of the Jungle** and **Part V: Appendices** is a wrap-up, including some advice for those of you who would like to try this at home, some editorializing, and a closing poem by Bob, which is a nice touch, not to mention a gutsy inclusion. The Appendices are a grab-bag of excellent information, including file formats, tape and file prep, labeling and logging practices and templates, some decibel and level info, bandwidth/Q tables, digital speed and capacity info, more about Bob's K-System, a bibliography of books and test CDs, and bios of Bob and his editor, Eric James, in addition to the previously mentioned radio info and glossary.

So, there's a lot here—a wealth of information. Generally, I find it thorough, accurate, and up-to-date. On that basis, I believe this book should be an essential part of any audio professional's or enthusiast's library. However, I do have some quibbles to share with you, not to disparage the book, but rather to line your expectations for it up with what you'll actually find when you buy it. You need to keep in mind that Bob is a mastering engineer by profession, not a teacher or a writer, and in his writing he tends to be hyperactive, impatient, and intense. He wants to tell you everything at once, and he tends to throw himself into what it is he wishes to say.

As a result, the writing style is a little herky-jerky and overly busy, which makes the book harder to read and to make sense of than I would like. Lots of inserted "grabbers" and "audio myths" add to the clutter for me, and tend to break into the flow, rather than illuminating it. Too many acronyms also get in the way. Finally, Bob's excellent and chatty footnotes are placed at the ends of chapters instead of the bottoms of pages, making them difficult and annoying to use.

I often found myself wincing at places where perspective wasn't clear, where teaching examples were missed, and where it was apparent to me that Bob sometimes is not particularly sensitive to (a) what confuses people and (b) what they'd really like to find out at any given point.

An example: an early chapter called "Decibels for Dummies" is actually a fairly advanced discussion of digital levels management. There is hardly a mention of decibels at all, and no discussion of the basic principles of what decibels are or where they came from, what levels "mean," or how we should think of them in a fundamental way. Instead, he dives straight into the deep end, starting with 0 dBFS, overs, normalization, and loudness.

I found myself becoming uncomfortable (and I know this stuff!). I suspect many readers may find it daunting. My recommendation: keep the faith! Bob does get to decibels (if much later in the book). And yes, most of the explanations you will need are there, just not necessarily in the right learning order. It may take a bit of digging.

Bob also occasionally indulges in some of the mythology he is generally careful to avoid or expose, so there are moments when you need to take a deep breath and a grain of salt. On page 52, for instance, talking about the need for re-dithering, Bob writes, "Unfortunately, many processor and DAW manufacturers still have not recognized this fact, and this partly explains why some digital devices sound pure and sweet, while others sound cold and harsh."

The axiomatic assumption that some devices sound pure and sweet while others sound cold and harsh tends to evaporate when it is carefully tested under controlled circumstances (Bob and I have conducted a number of correspondences about such topics, and I know he knows this as well). From my teacher/journalist stance, I would at the very least be inclined to qualify such a statement, saying something such as, "Many professional listeners blame the lack of re-dithering for a perceived degradation of sound quality, which they claim leads to a sound that is comparatively cold and harsh as opposed to pure and sweet. Such blame is, of course, difficult if not impossible to prove. Nonetheless, the belief persists."

Such a statement is a little more wordy, complex, and cautious, but it is also more meticulous (just like mastering has to be!) and more carefully describes the reality of the situation. Also, there is a missed opportunity here to discuss the politics of "what you tell clients" as opposed to "what you can prove in a test lab." Be on the lookout!

I didn't find many outright errors or omissions in the book. The biggest problems I found in this regard were in the monitor calibration chapter (14). Bob makes the case that we need calibrated, known levels (which I absolutely and unequivocally agree with), but inexplicably fails to mention the primary reason for such a need, the Equal Loudness Contours (which illuminate the fact that as we change level we are also changing perceived spectrum, timbre, and equalization). There is also a minor misstatement here about the relationship between Sound Intensity (which is directly related to power) and Sound Pressure Level (which is derived from the square root of power).

More important is some ambivalent editing and use of the term "monitor gain." Bob correctly defines gain and monitor gain, but sometimes incorrectly uses it in the text, which is confusing. Finally, the process to be used for monitor calibration is not described well, so it was difficult to follow (a) what monitor calibration is supposed to accomplish, (b) what you need to do in order to calibrate, and (c) why you need to do it.

These are all minor bumps in a really nice road. Look, you may encounter some confusions as you read *Mastering Audio*, there will be parts that are less than entirely clear, and some of the layout and hypertextual pace may annoy you. For all that, Bob's discussions of compression and equalization are probably the best and most comprehensive I've ever read, and his worldview is powerful and relevant. As I've said earlier, if you're interested in how recordings are made or how to make them, buy this book! Thanks for listening. ❖

Dave Moulton is a recording engineer, teacher, and loudspeaker developer. He is also author of Total Recording, Golden Ears Audio Ear Training, *and* Ten Essential Lectures About Audio. *You can complain to him about anything at his website, www.moultonlabs.com.*

Classifieds

VENDORS

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

JENA LABS
Our websites are our information centers for interconnects and cables, AC power solutions, DIY and OEMparts, Immersion Cryo services, and dealer list. **jenalabs.com jenatek.com**

Summer Sale!
www.borbelyaudio.com
Selected Borbely Audio kits in North America:
www.audiokits.com
Selected Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio/

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**.

Rockler Woodworking & Hardware
Innovative solutions for all your speaker building needs
www.rockler.com
1-800-403-9736 Free Catalog

Mic and phono preamps, meters, filters, etc. all US built. **TDL Technology, Inc., 505-382-3173 , www.zianet.com/tdl**

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

Yard Sale

FOR SALE

Pair of Dick Olsher's Black Dahlias, $200. Send a stamp for list of other speaker inventory. Patrick, (530) 885-3163.

Aragon Digital Processor Model D2A Mark II, mint condition, $375. For use between CD transport and amplifier. Call (203) 929-0212.

WANTED

Audio Transformer Design Manual and *Transformer Design and Manufacturing Manual* by Robert G. Wolpert. Back issues of *The Sensible Sound* (numbers 2–5, 8, 10, 11, and 32) to complete my collection (I have available numbers 14–18, 20, 21, 23, 26). Contact Angel Rivera (941) 798-9227 (Fl.), rivera_angel@hotmail.com, or alrivera@tampabayrr.com.

AD INDEX

Classic Circuitry

Haynes Duophase 14W Amplifier (1934)

Submitted by Philip Taylor

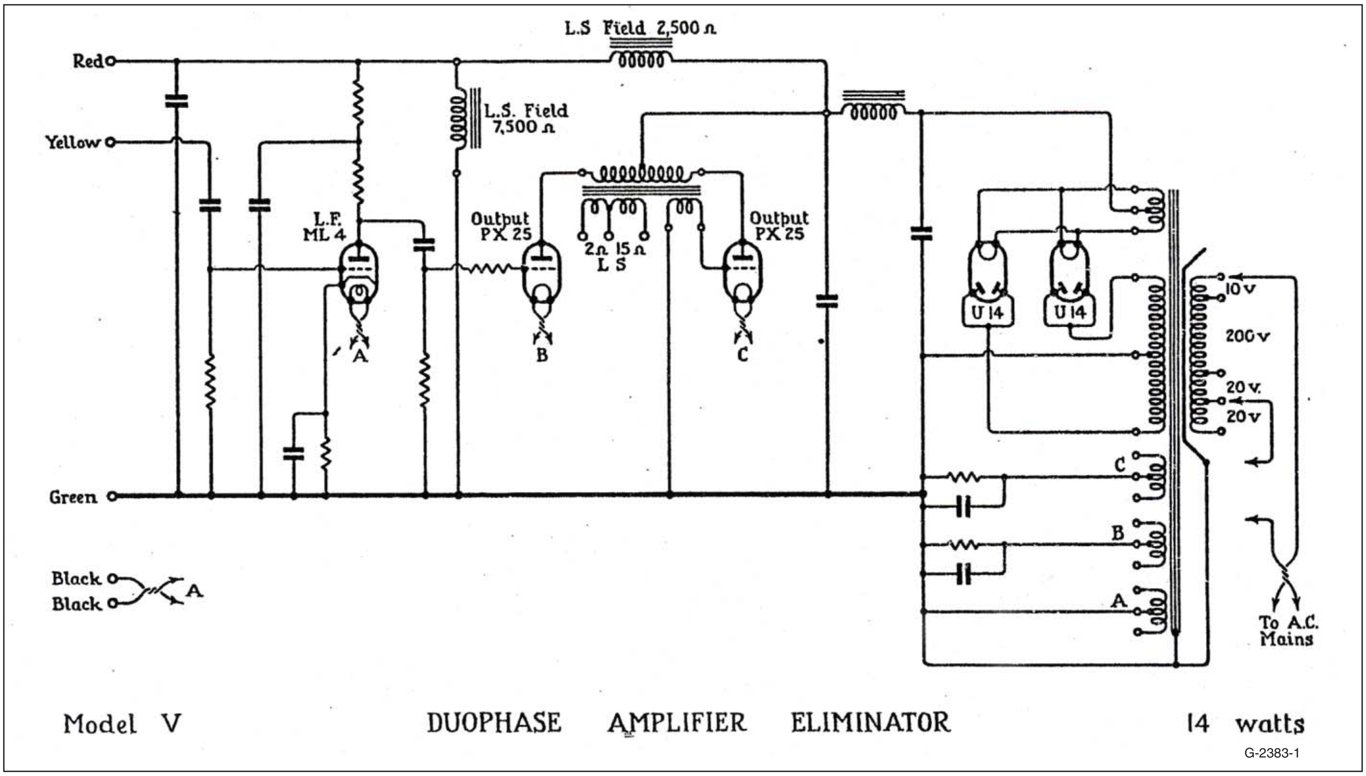


FIGURE 1: Haynes 14W amplifier circuit from 1934.

DUOPHASE—ITS ADVANTAGES (MERITS OF THE OUTPUT STAGE USED IN HAYNES RECEIVERS)

1. **No Back Coupling—Improved Bass Performance.** The pair of output valves, working in opposite phase, prevents the setting up of heavy currents, varying at low frequency in the H.T. supply and the grid bias circuits.
2. **Avoids Valve Distortion.** The distortion due to curvature of valve characteristic is largely removed by the two oppositely phased valves.
3. **No Voltage Amplifying Intervalve Transformer.** The output stage, comprising its pair of valves, works with a much lower input signal voltage than is customary, and avoids the use of a voltage increasing intervalve transformer.
4. **Fewer Components.** Less equipment is required to feed the output stage, with a consequent reduction in possible causes of distortion.
5. **All Intervalve Couplings Are by Resistance.** The preceding stages are entirely resistance coupled and the use of an iron cored intervalve transformer of high inductance is avoided.
6. **No Parasitic Oscillation.** The input couplings of each of the output valves are dissimilar, and the grid of one is tied down to earth by a coil of very low reactance. In consequence, the output valves have no tendency to break into oscillation.
7. **Avoidance of Hum.** The input to the "secondary" valve of the Duophase stage is derived from a winding of almost negligible reactance, representing a considerable step-down ratio. In consequence, there is no possibility of A.C. ripple being fed to the grid by stray A.C. fields arising from the disposition of the mains smoothing equipment.

Despite a lot of yakking, near compulsory attendance at audio jumbles over the last ten years, and a lot more years visiting vintage radio swap meets, I have never seen a Haynes amplifier and have never met anyone who owns one. I suspect they were in very limited production and would suffer at the hands of incompetent and disinterested repairmen.

It would certainly be very interesting to measure the phase splitting action in this Haynes circuit, especially at the edges of the audio spectrum. In those far-off days, I should imagine the designers would be aiming for about 80Hz to 10kHz maximum, and perhaps not as wide as that in 1934. ❖