WHY NOT TRY TUBE CROSSOVERS?

audioXPRESS

THE AUDIO TECHNOLOGY AUTHORITY

SEPTEMBER 2003
US $7.00 Canada $10.00

Continuing *Audio Electronics*, *Glass Audio* & *Speaker Builder*

Build a High Quality Class A Amp

A "Killer" Center Channel Speaker

An Easy-to-Construct 8″ Two-Way

Refining the Split-Load Phase Inverter

DC Sensor and Timer For Your Loudspeaker

Basic Facts About Filters

www.audioXpress.com

09
0 74470 84697 0

Plus: Building Rozenblit's 80W OTL

NEW !!

DH LABS

SILVER SONIC™

HIGH PERFORMANCE AUDIO CABLES

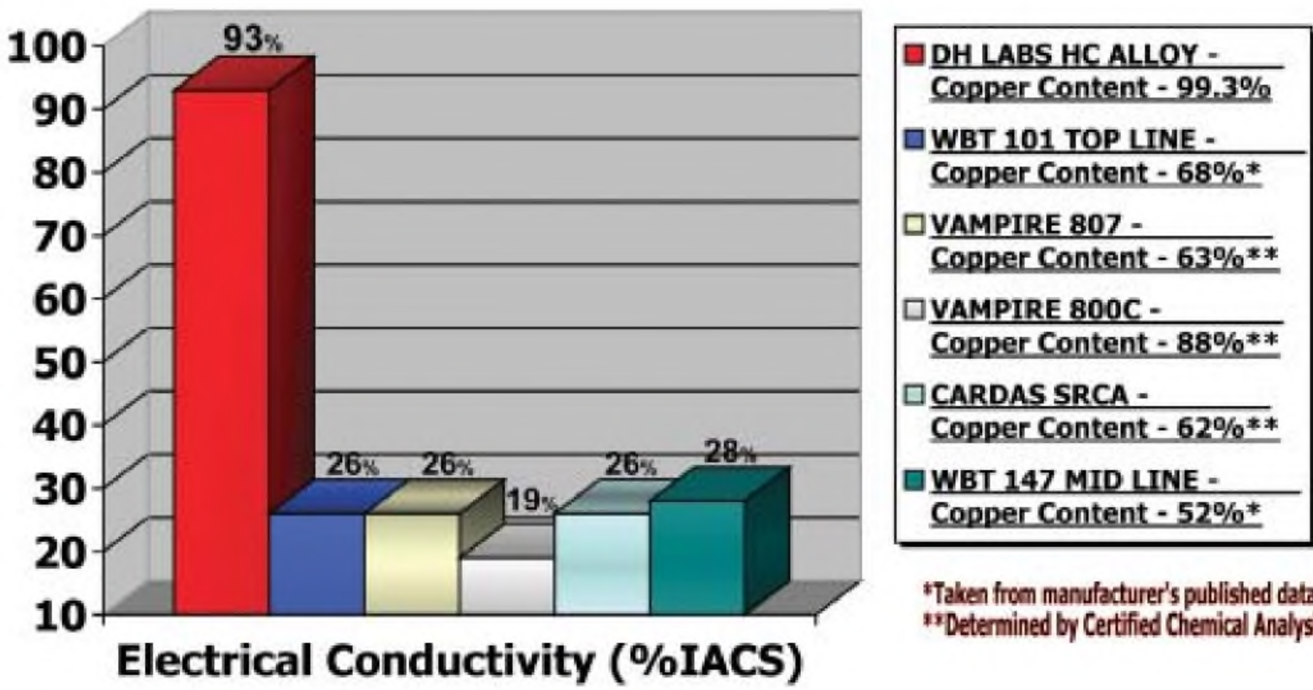
The Ultimate RCA Connector

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

audioXpress

CONTENTS

VOLUME 34 NUMBER 9 SEPTEMBER 2003

FEATURES

MATT'S BIG CLASS A AMPLIFIER
Fans of Class-A designs will appreciate this 20W per channel single-ended version featuring a bipolar input and MOSFET output.
By Matt Tucker 6

TUBE-BASED CROSSOVERS
Using a free design program will help you determine the component values of the tube crossover for your speaker system.
By John R. Broskie 16

A CENTER CHANNEL SPEAKER: A DIFFERENT APPROACH
Determined not to settle for the same old box-on-a-box center-channel/TV configuration, this author provides an attractive, effective solution.
By Daniel L. Ferguson 24

DC PROTECTION AND SWITCH-ON DELAY FOR POWER AMPS
Protect your speaker from unwanted DC with this simple-to-build circuit that also features a switch-on delay.
By Jenoe Keceli 34

THE ABCs OF FILTERS, PART 1
From the simple to the complex, this tutorial covers everything you need to know about filter terminology.
By Richard Honeycutt 38

AN UNORTHODOX TWO-WAY, PART 1
Here's a time-tested design, requiring minimum cabinet construction, featuring an 8" mid-woofer.
By Jon Mark Hancock 46

PHASE INVERTER DESIGN
This author shares some refinements to the traditional phase inverter circuit for improved performance.
By Adriaan Hammer 56

page 46

page 6

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 56

DEPARTMENTS

XPRESS MAIL
Readers speak out 60

BOOK REVIEW
Pro Audio Reference
Reviewed by Bill Fitzmaurice 68

SHOWCASE
Building Rozenblit's 80W OTL
By Ignatius Chen 69

NEW CHIPS ON THE BLOCK
AKM AK4395 and AK4554
By Charles Hansen 72

IN EVERY ISSUE

CLASSIFIEDS
Audio-related items for
sale or wanted 70

AD INDEX 71

YARD SALE
Free classifieds
for subscribers 71

page 24

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

THE STAFF

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

JOHN STUART MILL

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

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

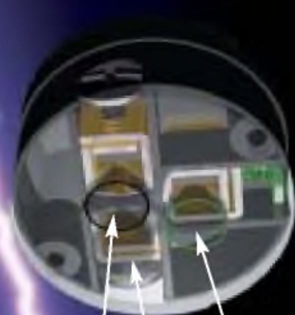
EDITORIAL INQUIRIES
Send editorial correspondence and manuscripts to *audioXpress*, Editorial Dept., PO Box 876, Peterborough, NH 03458-0876. **E-mail: editorial@audioXpress.com.** No responsibility is assumed for unsolicited manuscripts. Include a self-addressed envelope with return postage. The staff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

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

330

AUDIO GRADE™ POWER CONNECTOR

The construction of the 330 is built-to-last and features an elegant polycarbonate rear housing available in either black or clear with a durable Nylon-Tough front housing. Additionally, there is a dust seal to protect the wiring chamber. Marked with "Audio Grade™" and Blue Star™.



WATTGATE™

801-621-1501 • www.wattgate.com


THE ART OF CONNECTION

Matt's Big Class A Amplifier

This article describes the design and construction of a 20W per channel single ended Class A audio amplifier with a bipolar input and MOSFET output. It uses a three-stage topology with local and global negative feedback. **By Matt Tucker**

My senior project at San Diego State University involved one channel of this amplifier, which I presented in December 1999. The amplifier was inspired by my third and final analog design class, in which we designed and simulated an operational amplifier (op amp) using Microsim PSpice. We took all the transistor stages we had learned in previous classes to design a complete operational amplifier. The first two stages of the op amp are almost identical to the first two stages of my audio amplifier. The changes to the input stage included the type of current sources (the ones in the simulation relied on regulated power-supply rails) and a much higher transistor bias (current flow at idle).

The word op amp sends some high-fidelity fans running for cover. However, most op amps are single-ended (SE) Class A amplifiers up to the output stage. Standard op amps are biased very low to keep power consumption down to a minimum. With an increased bias and a Class A output stage, an operational amplifier becomes a nice-sounding audio amplifier.

I chose the output stage topology for two reasons. First, it is the most linear class of output stages. Second, it's unique and interesting to design. A Class A amplifier is made for do-it-yourselfers (DIYers). The heatsinks and the power-supply capacitors add to the cost of a SE amplifier, but you can often find parts at surplus stores.

Looking at the block diagram (*Fig. 1*), you'll note the amplifier is fairly straightforward. A differential input stage, common emitter gain stage, and an output stage make up the amplifier. The current sources in the full schematic (*Fig. 2*) can make the circuit look more complicated than it really is. Let's look at the current source first.

I SOURCE

The current source (*Fig. 3*) gives good current regulation independent of the variations in the voltages being supplied. Although explained before[1], let's take a look at how the current source functions. Looking at *Fig. 3*, at time zero there is no current flow and therefore no voltage drop. The voltage on the base of Qx rises quickly and turns Qx on. Current then flows through Qx and through Rc. As current increases through Rc, so does the voltage drop across Rc.

Eventually approximately .7V is dropped across Rc. That same voltage is applied base to emitter on Qy (ignoring the small Rd drop). This transistor turns on and starts to "steal" current away from the base of Qx. This reaches an equilibrium where the current through Rc and therefore the transistor

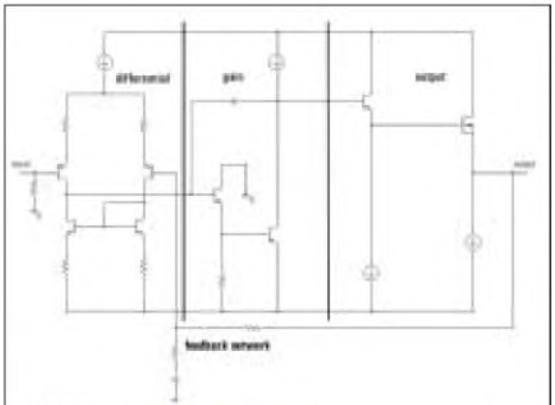


FIGURE 1: Amp block diagram.

A-2127-1

PHOTO 1: Completed amp with power supply.

ABOUT THE AUTHOR

Matt Tucker earned his BSEE from San Diego State University in 1999. He is employed at Mesa Power Systems designing and testing switching power supplies. Before returning to school, Matt worked for several years as an electronic technician repairing consumer electronics and pro-audio equipment. As a former piano, trumpet, and tuba player, he enjoys good reproduction of sound. Matt likes to spend time with his wife, ride motorcycles, and travel.

Over the Top
Music performance and craftsmanship that exceed your expectation in every way
Compass DANCER SERIES
Compass DANCER SERIES DESIGN HIGHLIGHTS:
■ Carbon-paper woofer cone
■ Silk-fabric soft dome tweeter; Ceramic type on some models
■ Innovative magnet flux design with corrective copper elements on all drivers, resulting in significantly reduced third and second harmonic distortions
■ Cast aluminum driver frame
■ Hand picked, matched-pair drivers on all models
■ Phase coherent crossover designed by world renown master of speaker design, built with components of the highest quality and thoroughly tested with digital-based measuring equipment
■ Massive, sonically dead front baffle which places drivers in a time-coherent physical arrangement
■ Multi-chamber reinforced cabinet with solid wood side panels, handcrafted to the highest furniture grade
Behind the Scene
With an abundance of original concepts in loudspeaker design, backed by thirty years experience in manufacturing and matched with an eye for fashion and unparalleled attention to detail, is USHER the ideal original design manufacturer you've always been looking for? Find out the answer today by talking to an USHER representative.
USHER
USHER AUDIO TECHNOLOGY
67, Kai-Fong Street, Sec. 1, Taipei 100, Taiwan
Tel: 886 2 23816299 Fax: 886 2 23711053
Web site: www.usheraudio.com E-mail: usher@ms11.hinet.net

to be biased is .7V/Rc. Choose the value of Rc to give the desired bias current.

DESIGN

The first two stages of the amplifier are straightforward. Please note that I consider the emitter followers (Q13, Q10) as part of the first and second stages, respectively.

The first stage is the differential input, made up of the PNP transistors Q1 and Q5. The input stage has a current mirror (also known as an active load) made up of transistors Q7 and Q8. The stage is biased by the current source Q4 and Q3. The second stage is a common emitter gain stage (Q9).

Mr. Douglas Self's book had some suggestions for current biasing and high beta transistors[2]. Very similar values were used (differential = 4.3mA, emitter follower = 6.5mA). These two stages are very common and can be found in most operational amplifiers and audio amplifiers. They are described in great detail in many books.

In short, the differential amplifier provides a convenient spot for feedback and provides some gain. The common emitter stage provides high voltage gain. The higher the open loop gain (gain without feedback), the more the amplifier gain will conform to the simplified op amp gain equation (1 + R2/R1), or, in the case of this amplifier, 1 + R18/R15 (1 + 15kΩ/604(= 25.83V/V).

THE OUTPUT STAGE

I have read Mr. Nelson Pass' articles regarding his SE amplifiers with great interest. The output stage in my amplifier is single-ended, but it is very different from the stage used in the Zen amplifier. My stage is a source follower with a voltage gain of less than one. The Zen amplifier has voltage and current gain in its output stage.

I determined the output stage specifics by choosing the standard speaker impedance to be used. Knowing that the theoretical efficiency of a single-ended Class A amplifier is 25%, I chose to make an amplifier that was 20W per channel. That meant that with small additional losses I would need to dissipate about 100W per channel. The actual power draw was 112.5W.

Since the output stage uses a constant-current source, I needed to calculate how much current was required. I intended to use an 8Ω speaker with the amp. $I^2 \times R = P$, therefore $I^2 \times 8\Omega = 20W$. Since I wanted a constant 20W, the equation becomes $(.707 \times I)^2 = 20W$ . . . I = 2.24A.

PHOTO 2: Stuffed board mounted in chassis.

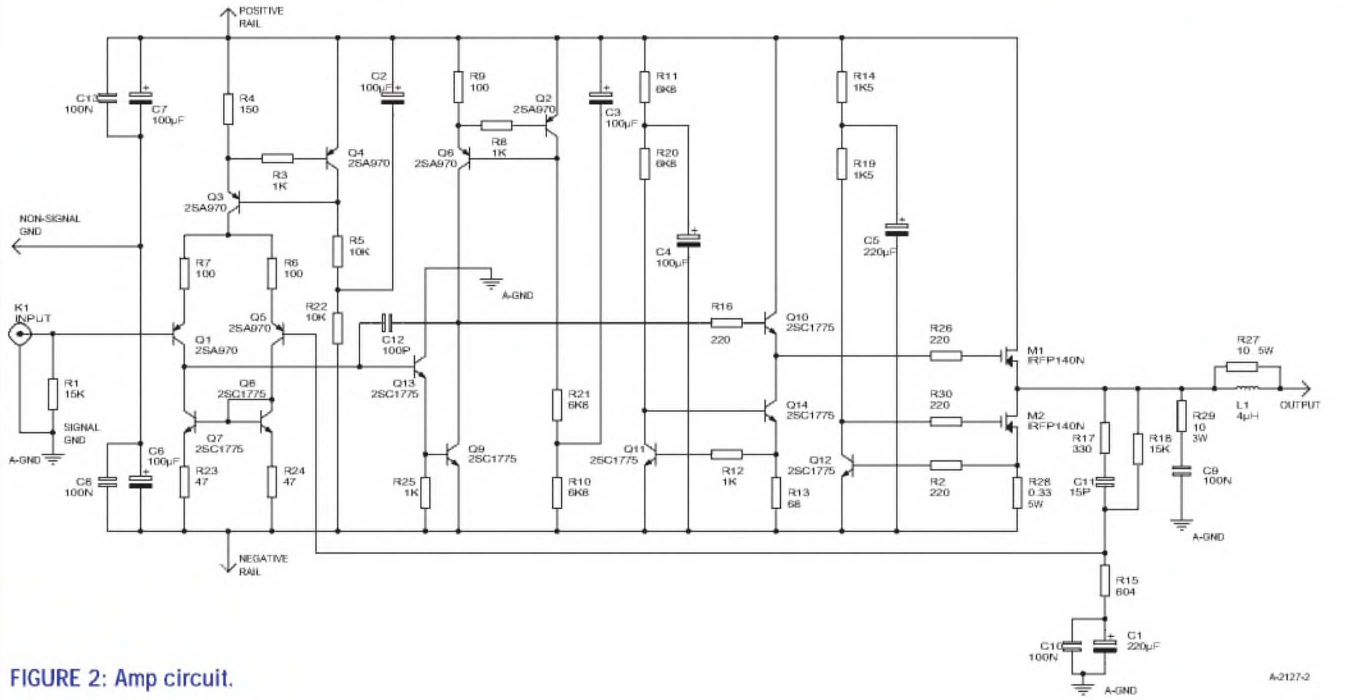


FIGURE 2: Amp circuit.

Using the constant-current source discussed earlier, .7V/Rx = 2.3A = .313Ω. The nearest available resistor in a 5W package was .3Ω. This value caused 2.3A to flow. I now had enough current to give 20W, but I needed to be sure I had enough voltage.

Determine the voltages of the positive and negative rails by V/R = I, and the fact that the amp is designed for an 8Ω load. The equation becomes V/8Ω = 2.24A. Solving the equation gives the minimum voltage of 18V DC. The amplifier cannot swing to the rail because of base-to-emitter (Vbe) and gate-to-source (Vgs) voltage drops. The most important drop is the Vgs of the output MOSFETs.

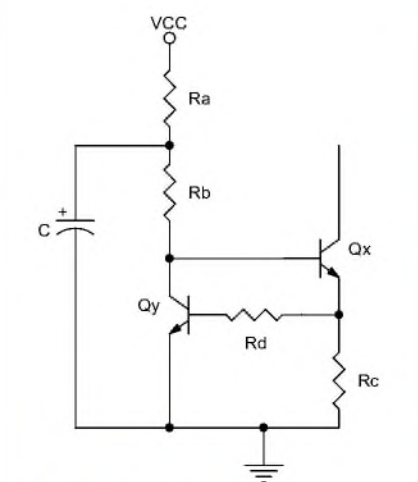


FIGURE 3: Current source.

The gate to source threshold voltage is approximately 4V. Adding 4V to the required 18.0V, the new minimum rail voltage is approximately 22V DC. Accounting for the rail voltages drooping with a constant 200W load, I wanted to find a transformer that would give approximately ±25V DC rails.

The extra 8V needed (4V for each MOSFET) translates into 18 wasted watts. You could use a separate power supply with higher voltage to make the output swing closer towards the plus and minus output rails, but that sounds like a lot of work for a small return. In an amplifier that burns over 110W per channel at idle, what is 16 more watts?

A speaker with a lower output impedance will cause less output power. Unlike Class AB designs, where more current is provided as necessary (given an adequate transformer), a Class A amplifier has only so much current to give before the constant-current source on the output runs out. Lowering the impedance of the speaker to 4Ω will halve your power, but only the power being delivered on the negative swing (where the output is less than 0V and all the current for the speaker must come from the current source).

Going back through the current and voltage requirements, you can see that it would be easy to modify the output power. I looked at many different MOSFETs before choosing the IRFP140N, which has a 100V drain-to-source rating and can handle 33A. It is readily available and reasonably priced.

If you desire more power, you might consider using a fan. Even though the MOSFET can handle more current flow, you can get only so much heat away from a single TO-247 package without forced convection. If you put several MOSFETs in parallel for increased power handling, you should use source resistors to help current sharing. Even hand-matched MOSFETs don't share well[3] above or below the current that they were matched at.

BOARD PRODUCTION

After thoroughly simulating the circuit, I used the Veribest PCB program (bought by Mentor Graphics) to lay out the board. Using a machine that resembles a router, I attached the single-sided PCB to a large metal plate and then routed away the copper that was not needed. For example, if a single trace is needed, the machine grinds away the copper to the left and right of the vertically running trace. The more it grinds away, the more space between that trace and another trace.

The first board turned out the best. I completed the second board about six months later. The amount of copper removed between traces was reduced to speed up the process. This resulted in many shorted traces. The ones I could locate I removed with an X-Acto knife. Two hairline shorts I could not find were removed with a 12V battery and a brief burst of current flow. That method is risky, but it was very effective.

The machine drilled all the holes the same size. On the first board, I enlarged the few holes that needed to be increased with a high-speed drill at school. On the second board, I used a standard drill with a bit that was slightly too large. I paid for this by having several of the copper traces lifted off the board. The board became ugly and messy, but still functioned well (*Fig. 4*).

POWER SUPPLY

I used a toroid transformer for its low magnetic radiation—a Toroid of Maryland #738.182, which has a 385VA rating and a dual 18V AC output. The secondaries are in series to achieve a 36V AC center tap transformer, while the dual primaries are in parallel for 120V AC use.

Originally I had two capacitors in parallel per voltage rail after the bridge rectifier. I used a pair of 29,000µF and 59,000µF 75V capacitors (purchased from a surplus store) for a good reserve. You can use 35V capacitors instead. A slow blow fuse accommodates the large current spike caused by the capacitors charging at turn on. I originally used a 2A fuse when I had just one channel completed. However, the fuse would sometimes blow, especially if the amp was cycled on and off.

I changed to a 5A slow blow fuse and now use a "pi" filter to reduce the ripple (*Fig. 5*). The pi filter started out as .3Ω of resistance connected between the first and second set of capacitors on both the plus and minus voltage rails. The bridge feeds the 29,000µF capacitors and then current flows through the resistors to the 59,000µF capacitors. Most of you have probably seen an inductor used in a pi filter, but with a Class A amplifier and its fairly constant current draw, a resistive pi filter works well[4].

I finalized the power supply by replacing the resistors with thermistors— SG420s, which have a room temperature resistance of approximately 2Ω. When first turned on, the current rush is improved by making the 59,000µF capacitors charge through the thermistor. When the thermistor warms up, the resistance of the thermistors is approximately .15Ω. Because the thermistors have a 23A maximum rating and I have only 4.6A running through them, they never get to their normal operating temperature. This means their resistance never drops to the intended impedance. The use of a larger than needed thermistor gives surge protection and enough resistance to

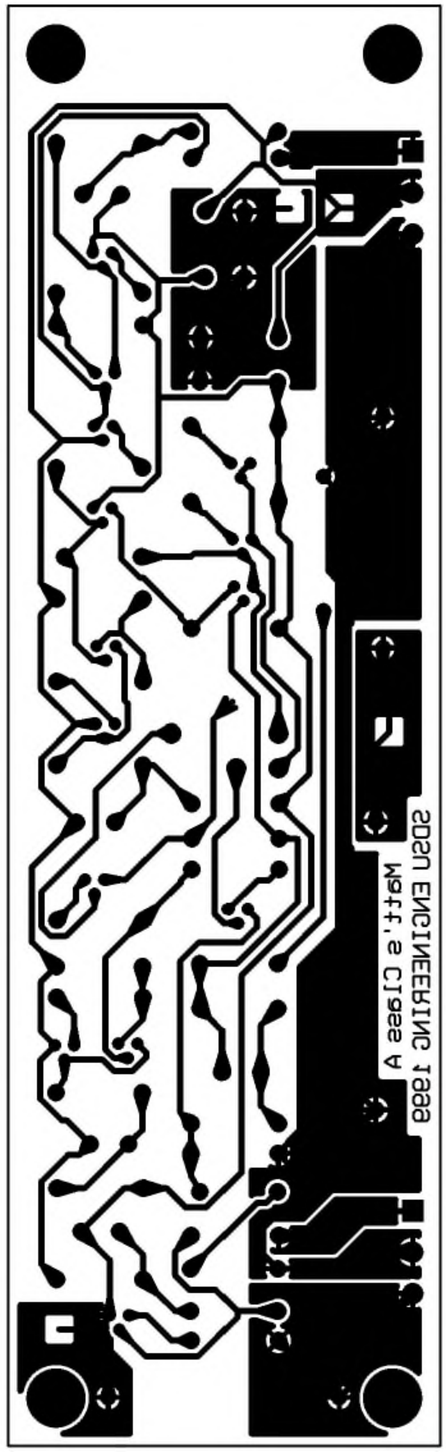


FIGURE 4: PC board artwork.

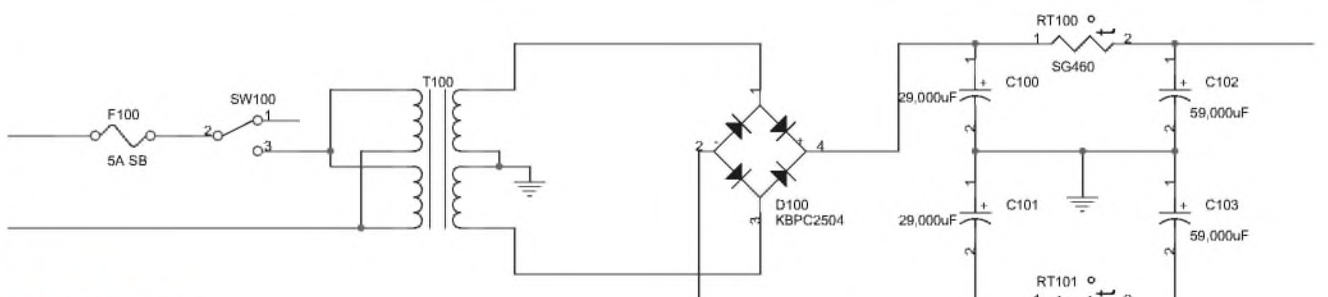


FIGURE 5: Pi filter.

filter out a lot of 120Hz ripple. Two for one deals are rare.

Looking at just the resistance of the thermistors and the 59,000μF capacitors, you can see a low-pass filter with a −3dB point of 1.8Hz has been produced $1/(2 \times \pi \times .15\Omega \times 59{,}000\mu F)$. The effect can be seen by the reduction in 120Hz ripple. Before the thermistor, the ripple on the positive rail is 760mV p-p; after the thermistor, the ripple is 90mV p-p.

HUM

By far the most frustrating problem was the hum the amplifier produced in the speakers. I first approached the hum problem with the idea that it must be coming from the 120Hz ripple (2 × 60Hz because of the bridge rectifier) on my voltage rails. I neglected to measure the frequency with an oscilloscope. I added more capacitance (as though the inrush current wasn't already large enough), and installed a "pi" filter. Neither helped. I did keep the pi filter, which made a huge reduction in ripple.

I finally noticed that the ripple on the output was 60Hz and not 120Hz. This showed me the noise was being introduced before the bridge rectifier. I unbolted the toroid transformer and rotated it while listening for the hum. I found the best spot for the transformer was on its side rotated off center.

If I positioned the toroid just right, I could get rid of almost all the hum. However, if I moved the shielded input cable a small amount the noise would return. I resisted mounting the transformer externally, but I eventually did.

Before I realized my hum was transformer related, I tried numerous changes to the board. I put another RC filter on the input and gain stage, moved grounds around, and tried changing capacitor values. I tore the underside of the board up a bit, but none of these changes helped since my problem was from the transformer.

Photo 1 shows the amplifier with the small power supply box to the right. My hum is now very low. With no inputs connected, there is a 60Hz ripple of 3–4mV that is not audible with my 90dB/1W speakers. I believe that some of my hum problem was due to my RCA input jacks being mounted on opposite sides of the chassis.

If the shielded input cables ran together, they would have very similar 60Hz noise on them and therefore little 60Hz potential between them. Since I ran my input cables far apart, the 60Hz noise was not the same on both cables. The problem increased because the toroid is in between the left and right input cables. This could cause a small current to flow in the ground line and cause hum. This is my theory and possibly the reason you always see RCA input jacks mounted right next to each other.

My transformer is now in a plastic box with one 14-gauge three-conductor AC cord going from the secondary of the transformer in the plastic box to the bridge rectifier in the chassis. A second AC cord provides the primary with 120V, and can be 18 gauge since there is much less current with the 120V potential. I don't use the earth ground wire because the transformer is in a plastic box. All voltages in the chassis are isolated from the primary via the transformer. I no longer use the power switch and fuse on the metal chassis but left them for aesthetic value. The functional power switch and fuse are mounted on the plastic power supply box.

The hum problem is more of an issue

on Class A amplifiers because of the large current draw at idle. Not only do the rails have more voltage ripple at idle, but also the magnetic fields produced by the transformer are stronger. With a Class AB amplifier, very little current is drawn at idle.

The heatsinks have only the output and current source MOSFETs (M1 and M2) mounted to them. A sil-pad thermally couples the MOSFET to the heatsink and electrically isolates the drain of the MOSFET from the heatsink and chassis. *Photo 2* shows the stuffed board mounted in the chassis. The reflection of the coil, 5W resistor, and other components on the board come from the shiny finish of the brass plate that holds two heatsinks together.

I first tried a heatsink having approximately 1500cm^2 of surface area. Even though the amplifier was not yet built, I could simulate the heat dissipation by running 2.3A of current through a MOSFET that had 25V across it drain to source. Since I only had one-quarter of the heatsink (one heatsink of four for a complete stereo amplifier), I tested with only one of the four MOSFETs. The test simulated the idle condition where the output of the amplifier would be zero and there would be 25V across each MOSFET with 2.3A flowing.

After the increase in temperature had steadied, the heatsink measured 144°F (62°C). The temperature was not acceptable. The MOSFETs have a maximum temperature rating of 170°C, but you must consider the human factor; a temperature of 144°F is painful to the touch.

Temperatures between 130–140°F were still hot, so I needed a lower temperature and purchased another heatsink for testing. Its surface area was 2300cm^2 and temperature was 106.1°F. This was appropriate for human contact, so I bought three more heatsinks from the surplus store.

The finished amplifier causes the heatsink temperature to be between 120–130°F at normal room temperatures. The heatsink is very hot to the touch, but it will not burn you and cause only slight discomfort. I recommend even more heatsink surface area. The cooler the FETs run the longer the life they should have. I used four heatsinks (two on each side). I connected the pairs of heatsinks together with a brass plate.

CHASSIS

I used the heatsinks as the frame of the amplifier. *Photo 3* shows the open amplifier in its final configuration. All aluminum panels bolt to the heatsinks. I used .125″ aluminum so the walls would not bend with the weight of the transformer and large capacitors, and later removed the transformer from the chassis.

This was my first time working with metal, so I was learning as I went. I do not give dimensions for the metal, because I bought the heatsinks surplus, so it is unlikely you'll find these exact ones. I brushed the chassis and then chem-coated it with a gold/yellow color. I used five-way binding posts for the speaker outputs.

The boards mount along the left and right long sides of the chassis with two standoff posts. The other side of the board is supported by the MOSFETs, which are bolted to the heatsinks. Mounting the output transistors directly to the heatsink allowed less wiring and kept the signal path of the gate drive short. I punched vent holes in the top of the amplifier, but I should have used thinner metal for the cover to allow for easier punching. I bent and marred my cover a little trying to get through the .125″ metal.

BRINGING THE AMPLIFIER UP

Simulations can take you only so far, but as evidence that a simulation can

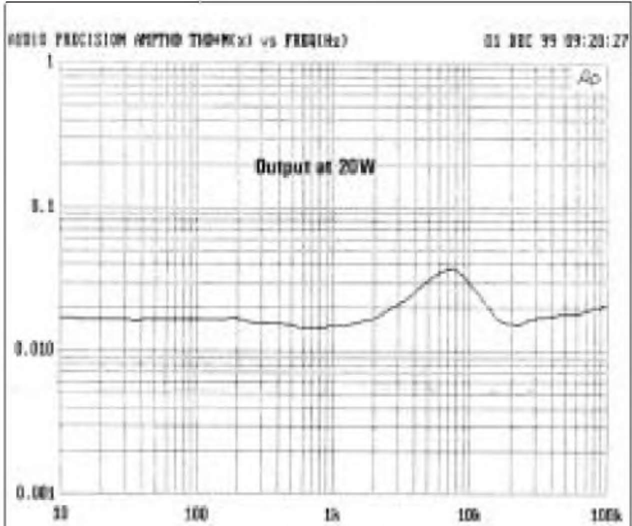


FIGURE 6: Distortion graph.

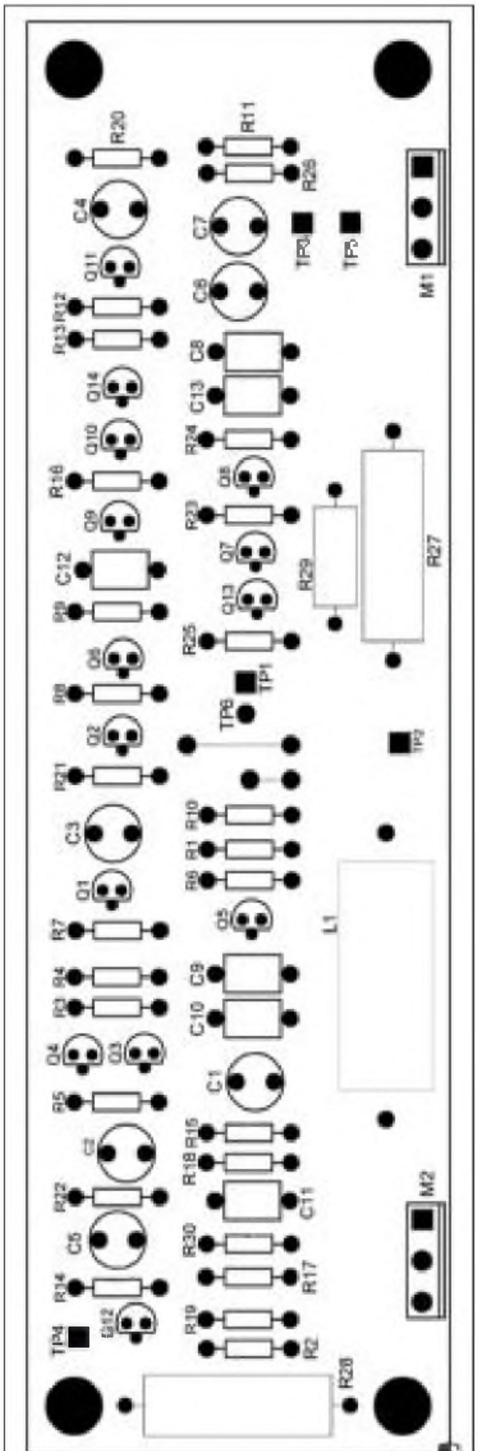


FIGURE 7: Parts placement.

get you started in the right direction, I am happy to say the amplifier worked the very first time I tried it.

You should substitute a large value resistor for R28 (3.3–4.7Ω) to keep current flow low in the output stage and therefore power dissipation low. Fewer electrons will be available for cooking a mistake. Hook up the PS to the AC line (slowly brought up on a Variac® would be really nice). I usually use 8Ω resistors on the plus and minus voltage rails before they enter the board. The 8Ω resistance is enough to keep a short on the board from causing mass component destruction, but low enough to power the circuit (that is, if you are using a high value for R28).

PHOTO 3: Amp overhead view.

Feedback: I consider the amplifier to be DC coupled, but there is room for argument. The frequency response starts to roll off at 20Hz (–3dB at 1.7Hz). The amplifier does not respond to DC because of C1 and C10 in the feedback network. These capacitors block extremely low frequencies from being attenuated by the divider made up of R18 and R15.

The full DC offset is fed back to the differential section. This helps give a low DC output offset. Higher frequencies are passed through the capacitors to ground so only a percentage of the output signal gets sent back to the differential input. The amplifier is DC coupled input to output, but the feedback network keeps the amplifier from having true DC response.

MEASUREMENT/DATA

The input impedance or frequency response is not as important as the sound of the amplifier, but it is nice to have some quantitative numbers. Here is how I measured a few common amplifier specifications.

I measured the gain of the amplifier by giving the amplifier an input that caused the output to be 6V RMS, and then measured the input, which I found to be .227V RMS. (.227V RMS) × (Av) = 6V RMS. The gain is 26.43V per volt (V/V) (only .6V/V off from the calculated gain using R18 and R15 values).

I measured the bandwidth of the amplifier by increasing the input until the output was 4V RMS at 1kHz. The –3dB point is $(1/\sqrt{2}) \times$ (V RMS) = .707 × 4V RMS = 2.828V RMS. The input was swept up in frequency until the output was at 2.828V RMS. The frequency was 201kHz. The input frequency was then lowered until 2.828V RMS was reached. The frequency that occurred was 1.7Hz.

I measured the output voltage offset with the RCA inputs left open. The left channel output was –16.5mV. The right channel had 1.1mV on the output. The Zobel network (also called Boucherot cells) is made of R29 and C9. The 10Ω resistor and .1µF capacitor (R29, C9) present the amplifier with a high-frequency load.

L1 is the output coil that helps protect the output from capacitive speaker loads and forms a low-pass filter with the speaker. The inductance of the output coil in this amplifier is approximately 4µH and was hand-wound. I used 16-gauge magnet wire with 23 turns. The diameter of the coil is 1.5cm and the length is 4.2cm. The exact value is not critical. I wrapped the wire around a large ink marker to get the shape. The coil should not have a core (only air).

I took the distortion graph (*Fig. 6*) before producing the second channel. This means that a little more voltage was available and much more 60Hz and 120Hz noise was present compared to the amplifier as it is now. The distortion is less than .02% THD at most frequencies and increases to less than .04% at 7.5kHz. I do not know why there is a bump in the distortion at 7.5kHz, but I would like to know whether readers have any suggestions. The pi filter and external mounting of the transformer helped with the noise.

GROUNDING

I followed a star grounding scheme. From *Photo 4*, you can see four black Phillips head bolts with a single light-colored bolt in the center of them. The center screw is the center tap of the transformer. The large capacitor charging current spikes travel in this section.

The grounds for the speaker outputs and on-PCB filter capacitors form the "star" ground configuration a few inches below on the plate. The two wires below the star are the grounds that go to the signal input (the shield of the input cables). Although aluminum is not the best conductor, the large plate (left over from the chassis) has very low resistance.

My input jacks are mounted to the back of the chassis and are connected to ground via the shielded cable. I tried

TABLE 1
PARTS LIST

ALL PARTS FROM MOUSER UNLESS OTHERWISE NOTED		
TRANSISTORS		
Q1–Q6	2SA970	620-2SA970
Q7–Q14	2SC1775	620-2SC1775
MOSFETS		
M1, M2	IRFP140N	Digi-Key IRFP140N-ND
RESISTORS		
R1, R18	15k	
R2, R16, R26, R30	220	
R3, R12, R25	1k	
R4	150	
R5, R22	10k	
R6, R7, R9	100	
R10, R11, R20, R21	6.8k	
R13	68	
R14, R19	1.5k	
R15	604	
R17	330	
R23, R24	47	
R27	10 5W	286-10
R28	.33 5W	71-LVR5-0.3
R29	10 3W	283-10
CAPACITORS		
C1, C5	220µF 50V	647-UVR1H221MPA
C2–C4, C6, C7	100µF 50V	647-UVR1H101MPA
C8–C10, C13	0.1µF 100V	140-PF2A104K
C11	15pF 300V	5982-5-300V15
C12	100pF 300V	5982-5-300V100
INDUCTORS		
L1	4µH	see text
POWER SUPPLY		
SW100	10A 125V AC rocker	Radio Shack 275-694
T100	36VCT 385VA	Toroid of Maryland #738.182
D100	25A 400V bridge	583-MP254
C100, C101	21,000µF 35V	539-CGS35V21000
C102, C103	40,000µF 40V	539-CGS40V40000
RT100, RT101	Thermistor	see text 527-CL30
F100	5A SB	Radio Shack 270-1027
not shown	fuse holder 10A	Radio Shack 270-367
MISCELLANEOUS		
Rubber feet		
Hardware		
Wire		
AC cord		
Board standoffs		
Sil-pad		
5-way binding posts		
RCA input jacks		
Shielded audio cable		

lifting the connectors off the chassis, but I did not get improvement in output noise. As I've mentioned, the chassis does not have an earth ground connection.

PARTS

The parts placement is shown in *Fig. 7*. The spots marked TP1 through TP6 correspond to signal ground, speaker output, filter capacitor ground, –25V, +25V, and input signal, respectively. The on-board filter capacitor grounds are separate from signal ground to keep the capacitor charging currents off the ground traces of the board. The two lines near TP6 represent two jumpers, which can be leads trimmed off resistors or capacitors.

The parts list is shown in *Table 1*. All parts are from Mouser Electronics (http://www.mouser.com) except for the MOSFETs, which are from Digi-Key (http://www.digikey.com); transformer (Toroid of Maryland http://www.toroid.com/); and the fuse holder and AC switch (Radio Shack http://www.radioshackcorporation.com/). All resistors are ¼W except where noted. The Mouser part number for the ¼W resistors are 271-VALUE, where VALUE is the value of the resistor (271-1.5K is a 1.5kΩ ¼W resistor).

I listed a Mouser part number for a Keystone-Thermometrics thermistor that is similar to the RTI SG420 that I used. The Thermometrics part is rated lower in current, but it still should have enough resistance when warm to give a noticeable drop in 120Hz ripple. If you would like to use the RTI part, try http://www.pcipci.com/. The capacitors I list for the power supply should provide similar results to the surplus capacitors I used. The chassis design and heatsinks are up to you.

Figure 8 shows the pinout for both bipolar transistors used. Please note that the Mouser website had a specification sheet for the 2SA970 that did not match the 2SA970 that I was sent. *Figure 8* shows the correct pin configuration for the transistors I received. The MOSFETs have a standard pin configuration. With the metal back away from you and the legs down, the pins are Gate, Drain, and Source from left to right. I used sixteen-gauge wire in the amplifier except for the AC cords.

How does the amp sound? Well, I simply say it sounds good. ❖

REFERENCES

1. Nelson Pass, "The Pass Zen Amplifier," *The Audio Amateur* 2/94, p. 14.

2. Douglas Self, *Audio Power Amplifier Design Handbook*, 1996, p. 75.

3. Motorola TMOS power MOSFET transistor data, Q4, 1992, p. 2-7-15.

4. Shrader, "Electronic Communication," Fifth Edition, p. 179.

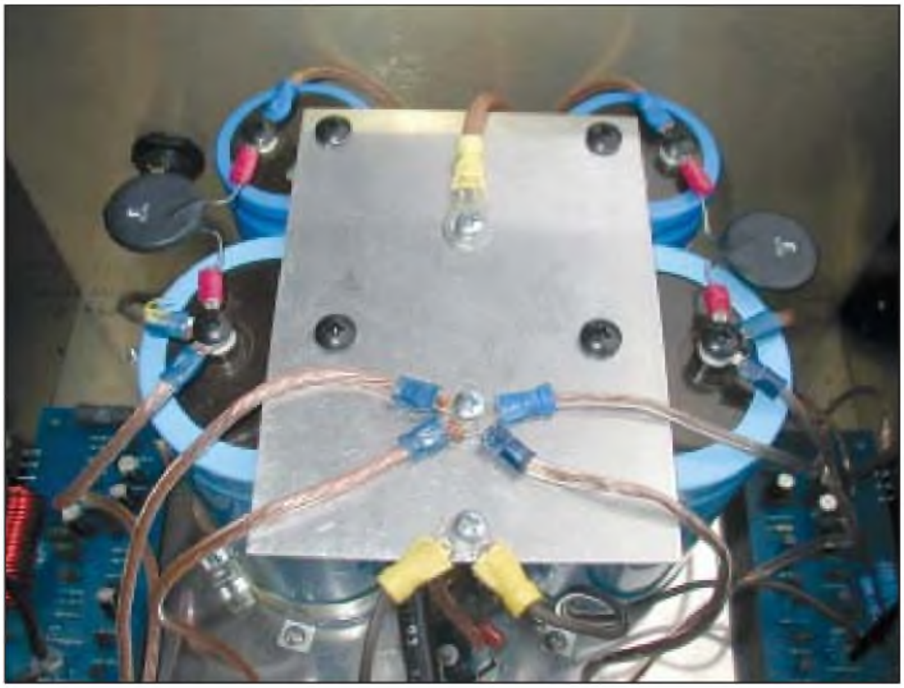


PHOTO 4: Grounding scheme of unit.

Tube-Based Crossovers

Unless a loudspeaker uses a single full-range driver, you will need a crossover to split up the audio spectrum into discrete bands of frequencies. But should the crossover accomplish this task passively or actively? And if actively, with solid-state devices or with vacuum tubes?

By John R. Broskie

By applying two or more filters to the input signal, crossovers give each loudspeaker driver its own band of frequencies in which to operate. The filters used come in two fundamental types: digital and analogue, with the latter further subdivided into either passive or active.

Each style of filter has its advantages and disadvantages. At extremely high frequencies, where active devices become bandwidth limited, passive filters work best.

At audio frequencies, however, the active filter wins, as passive filters bring too many liabilities to the job. For example, loudspeakers use passive crossovers that hold a combination of either heavy, expensive inductors along with expensive, bulky, film capacitors; or small, inexpensive, easily saturated, and poor-tolerance inductors and cheap, low-quality electrolytics placed in series to cancel their polarization.

Adding to these negatives, you must tweak a loudspeaker's passive crossover to work correctly when working into the reactive load that the loudspeaker drivers present.

Active crossovers, on the other hand, forgo using inductors; instead, they rely on inexpensive, yet precise, resistors and small, relatively inexpensive tight-tolerance capacitors to set the crossover frequencies. Beyond the savings in cost, active crossovers offer a greater accuracy and a much improved flexibility over passive crossovers. Tight-tolerance resistors and capacitors are inexpensive to obtain and are easily swapped for different values, allowing easy changes in crossover frequency. Finally, the active crossover allows a more efficient use of the power amplifier's output, as power-wasting speaker attenuator networks are not needed, when the gain can be reduced efficiently by the mere twist of a potentiometer.

TUBE-BASED CROSSOVERS

Tube-based crossovers perform the same function as solid-state crossovers, but use tubes rather than the ubiquitous IC. Why bother with tube-based audio equipment? As they say, if you have to ask, then you haven't listened to a good tube-based audio system. In an active crossover, the tube delivers the same easy-on-the-ear sound it brings to guitar amplifiers and phono preamps.

While realizing a tube-based crossover is no more difficult or complex than the solid-state alternative, it is more expensive. Tubes run on comparatively high voltages, which require comparatively higher prices. For example, a 250V 47µF capacitor costs eight times more than the 25V equivalent. In addition, the tubes themselves are not cheap, as you can obtain a handful of ICs for the same price as one good tube—definitely not the cheapest choice. Although tube gear is not easy on the wallet, it is easy on the ears and eyes. Even when not turned on, tube gear compels attention that no IC-based equipment can match.

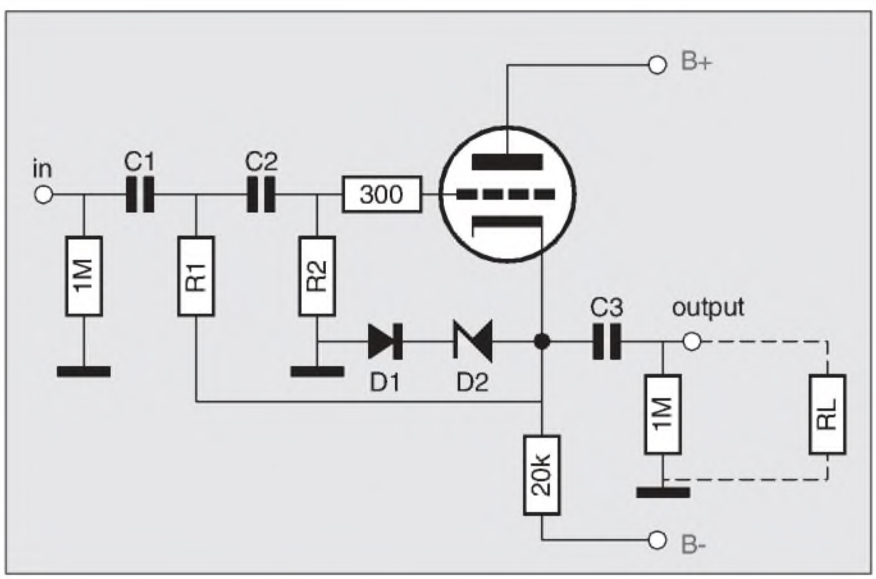


FIGURE 1: Example of a circuit generated by the Tube Crossover Designer software. Note the use of a symmetrical supply voltage. Part values depend on your requirements, and are shown by the program along with this diagram. G-2238-1


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

THE PLAN

This project aims to deliver a tube-based crossover in its most flexible form. Surprisingly, using a printed circuit board can contribute to this goal. While using point-to-point wiring is certainly possible, the PCB not only offers a consistent physical geometry and helps to prevent miswiring, but it allows a great deal of flexibility, as this PCB resembles a cross between a proto board and a typical PCB. Each board holds two tube sockets (four buffers) and many extra holes.

The flexibility is truly staggering. By reconfiguring the PCB, dozens of arrangements are possible, as first-, second-, third-, and fourth-order crossover slopes can be easily accommodated on this PCB. Two channels of matching high-pass and low-pass filters (up to third-order) fit on one board, as does a single channel's three-way crossover made up of first-, second-, and third-order filters (three-way crossovers using fourth-order filter slopes require cascading boards).

But so much freedom can baffle: Which capacitor goes where? Which half of the tube is the input? Unfortunately, the formulas and schematics needed to cover all the possible arrangements would fill this entire magazine. Consequently, we developed a Windows software program to calculate both the component values and their placement on the PCB. A screen of the program in action is shown in *Fig. 1*. You may download *Elektor's* Tube Crossover Designer program at www.elektor-electronics.co.uk. The file number is 020297-11.

The program covers 14 basic filter configurations, which, by cascading PCBs, easily doubles the possible crossover configurations. Unlike many electrical engineering programs, it is easy to use: just key in the desired crossover frequency, the filter type (high-pass or low-pass), the filter order (first, third, or fourth), and the filter alignment (Bessel, Butterworth, or Linkwitz-Riley) to display the part values in a grid. Pressing the "Schematic" and "PCB" buttons toggles the view between the circuit's schematic and its part placement on the PCB.

THE CIRCUIT

This tube-based crossover, of which a randomly chosen variant is shown in *Fig. 2*, makes use of cathode followers to offer both a low output impedance and a buffering of the crossover's tuning elements. Cathode followers provide no gain and little distortion, as all of the triode's potential gain is used as degenerative feedback to lower its output impedance and flatten its frequency response. Because the cathode follower's output follows its input in phase, its input capacitance is free of the Miller effect. Additionally, the cathode follower allows the crossover's output to be recirculated into the filter network to produce a positive feedback loop.

Many filter alignments need some positive feedback to eliminate the need for inductors within the network, and to compensate for the otherwise drooping output from a solely capacitive and resistive network at the transition frequency. The crossover uses either 6DJ8 or 6N1P tubes (or any other dual triode that shares the same pinout and heater voltage and has a low R_p and high mu, e.g., 6BQ7, 6BS7, ECC85, ECC88, 6DJ8, 6AQ8).

Referencing the cathode follower's input to ground sidesteps the need for additional input coupling capacitors. In fact, this circuit has been designed with a goal of eliminating as many coupling capacitors as possible. (While this also serves to cut the cost, the savings are somewhat offset by the need for a symmetrical power supply (±165V), rather than the more conventional mono-polar power supply of most tube equipment.)

PROTECTIVE DIODES

One problem is that the power supply will come up to full voltage much more quickly than the triodes will begin to conduct. This means that at turn-on the triode's grid will see 0V and its cathode will see −165V. Such a large differential can damage the triode by "cathode stripping," wherein pieces of cathodes' surface are pulled away by the severely positive grid when the tube is still cold.

Adding a diode and zener in series solves this problem. When the triodes are hot, the pair of diodes do not conduct and are effectively out of the circuit. When the triodes are cold, they conduct, forcing the cathode to be only a safe −11V relative to the grid.

THE POWER SUPPLY

The Signal IF-30-30 and IF-30-230 mains transformers are encased in plastic and are designed to be mounted on a print-

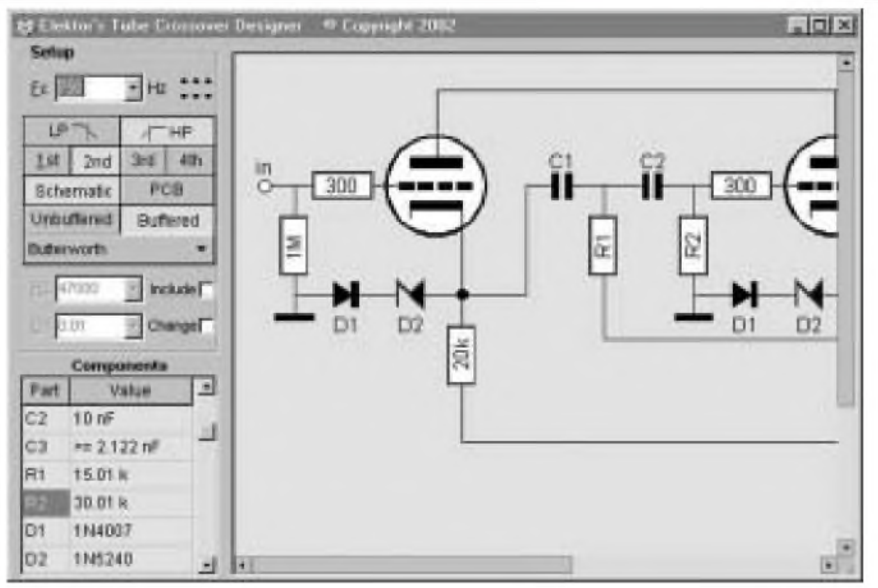


FIGURE 2: *Elektor's* Tube Crossover Designer in action. Complete circuits, parts lists, and component overlays are produced at the flick of a switch. G-2238-2

ed circuit board. Both come with dual primaries, allowing use in countries that use either 115V AC or 230V AC. *Figure 3a* shows the layout for 115V AC; *Fig. 3b*, the layout for 230V AC. The complete circuit diagram of the PSU appears in *Fig. 4*. It is a simple affair. Two power transformers are used, one for the high-voltage bipolar power supply for the cathode followers and one for the low-voltage heaters.

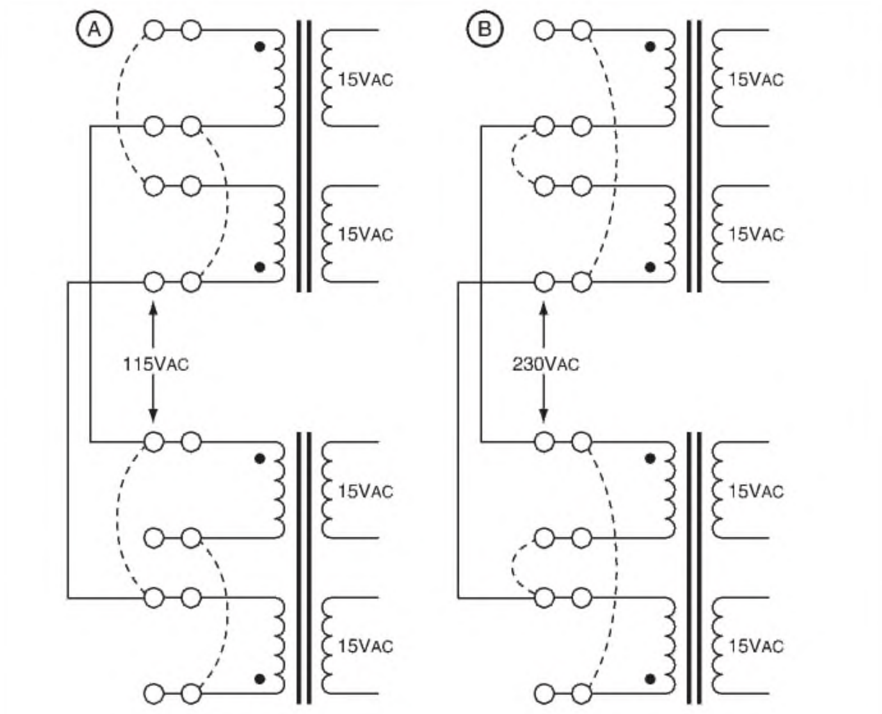


FIGURE 3: Mains transformer wiring for 115V AC mains (*3a*) and 230V AC mains (*3b*). The mains frequency (60Hz or 50Hz) is immaterial.

G-2238-3

Each transformer holds two secondary windings. The high-voltage portion of the power supply uses the two 115V AC windings to produce a center-tapped full-wave bridge configuration that yields the positive and negative 165V rail voltages. Because the cathode follower does such a good job of rejecting power-supply noise at its output, no regulation is used on these high-voltage rails. Instead, conventional RC filters are used.

The low-voltage heater power supply uses two windings to produce a full-wave center-tapped configuration that yields the single positive rail voltage. This raw DC power supply then feeds a fixed 12V three-pin IC regulator. The tubes contain 6.3V heaters, which when placed in series across the 12V power supply see only 6V. This lower voltage is within acceptable voltage tolerances and will only serve to extend the tubes' lives.

FILTER ALIGNMENTS

In audio applications, the Butterworth filter is the most popular type. This makes sense, because it boasts a fairly flat time response, a fairly crisp transition shape, and the flattest passband response. Its chief competitor, the Bessel filter, offers flatter time response, but not as sharp a transition shape nor as flat a passband response. Other filter alignments, such as the Chebyshev and the elliptic, are seldom used in audio because of undesirable characteristics such as ripple in the passband or sharp phase shifts.

The Linkwitz-Riley variation on the Butterworth is particularly useful for feeding loudspeakers, because it yields a flat transition region by providing a –6dB output at the crossover frequency. When two loudspeakers play the exact same frequency without amplitude or phase differences (when using even-order filters, e.g., second and fourth), the sum of their outputs is a twofold increase (+6dB boost) in volume. But as Butterworth filters are designed to have a –3dB output at the transition frequency, the loudspeaker will experience a +3dB hump at the crossover frequency, as even-order Butterworth filters produce an even multiple of 90° of phase shift (i.e., 180°, 360°, and 540°) per number of filter order at the transition frequency. Depending on the phasing of the speaker drivers, the result is either a deep null or a boost at the transition frequency.

On the other hand, odd-order Butterworth filters (e.g., first and third) exhibit an odd multiple of 45° of phase shift (i.e., 45°, 135°, and 215°) per number of filter order at the transition frequency, which gives rise to only a +3dB boost, which compensates perfectly with the –3dB dip in output at the crossover frequency. Thus the odd-order Butterworth filters sidestep the need for an adjustment in output amplitude.

In other words, making a flat even-order crossover requires that the signal not be –3dB, but –6dB down at the crossover frequency, which the Linkwitz-Riley alignment provides for second- and fourth-order crossovers. Still, experimentation is the best path. The advantage an active crossover enjoys is that experimentation is much more readily and cheaply done than with a passive crossover.

FIRST-, SECOND-, AND THIRD-ORDER CROSSOVERS

The first-order crossover is at once the most trivial and the theoretically best crossover configuration; this simple crossover offers the flattest phase and frequency summation at the transition frequency. Unfortunately, its shallow cutoff slopes often fail to both protect fragile tweeters and curb slow woofers from trying to reach too high. Moving up from two-way to three- or four-way crossovers eases the problem of potential damage to speakers as do robust high-quality (i.e., expensive) loudspeaker drivers.

One point to bear in mind is that you can use a first-order filter in conjunction with higher-order filters. One particularly felicitous pairing is a first-order high-pass filter with a third-order low-pass filter. This combination works well with full-range loudspeakers and subwoofers. To realize this arrangement, you need as little as one PCB for stereo, if you choose partially buffered filters.

In fact, you can define the high-pass section's filter by its output capacitor working into the load presented by the following amplifier. While this would

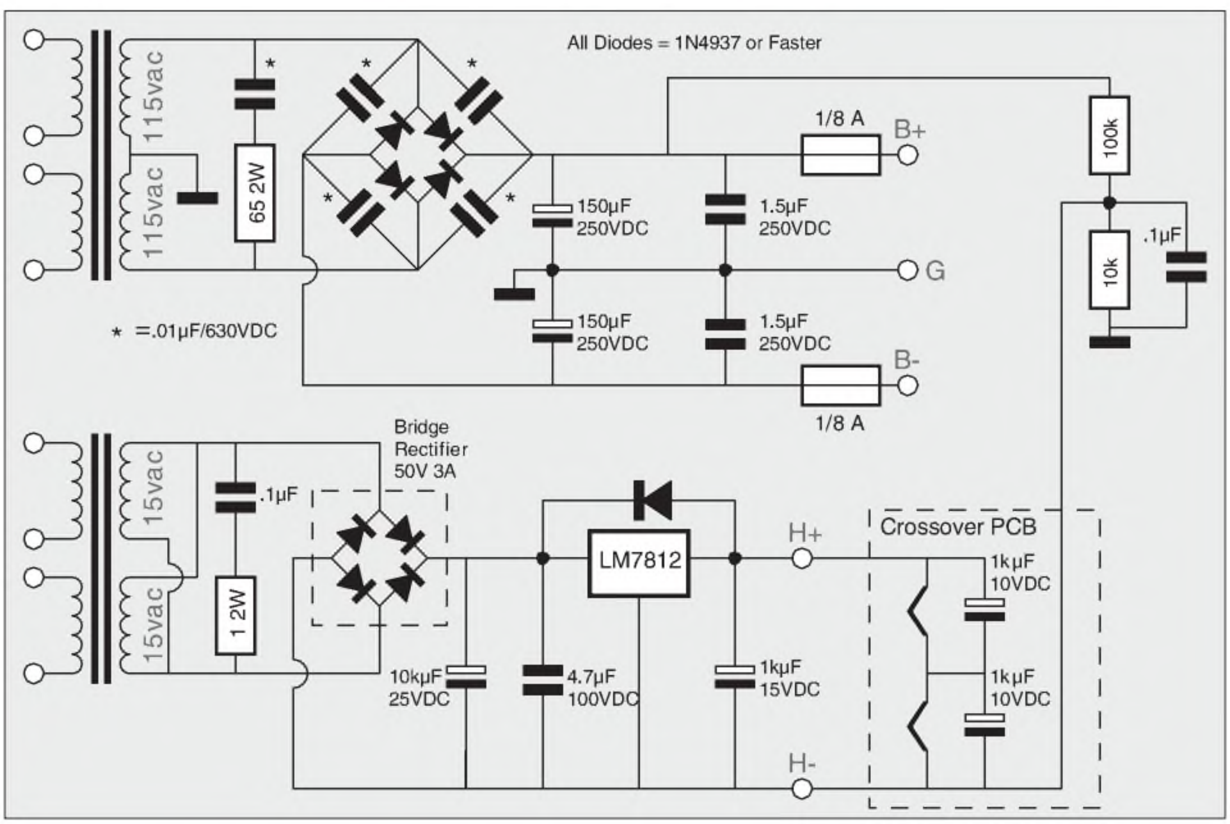


FIGURE 4: Circuit diagram of the power supply unit (PSU).

G-2238-4

eliminate one coupling capacitor from the signal path, it would make the crossover less universal, because the output coupling capacitor would need to be changed to match different input impedances from different amplifiers. Furthermore, using two capacitors adds an additional protective high-pass filter at the low frequencies.

You will need two PCBs for fully buffered filters, however, because both the high-pass filter and the low-pass filter must be nested between two cathode followers. Because of the phase differences in low and high outputs at the transition frequency, you will need to experiment to determine the best integration between subwoofer and satellite, and you will probably need more to reverse the subwoofer's phase by reversing the speaker cable's plus/minus connections to the subwoofer.

As you move up in filter order, more options become available. The second- and fourth-order crossover can be given Bessel, Butterworth, and Linkwitz-Riley alignments. The third-order crossover can be aligned to Bessel and Butterworth alignments, but not Linkwitz-Riley.

For all three alignments, the PCB may be used in two configurations: completely buffered and half-buffered. In the completely buffered version the filter tuning elements are safely tucked between two cathode followers, and you will need one PCB per channel to provide up to third-order filtering. Buffered fourth-order crossovers are not covered, because they would require an excessive number of PCBs. For example, a fully-buffered two-way, fourth-order crossover alone requires 12 cathode fol-

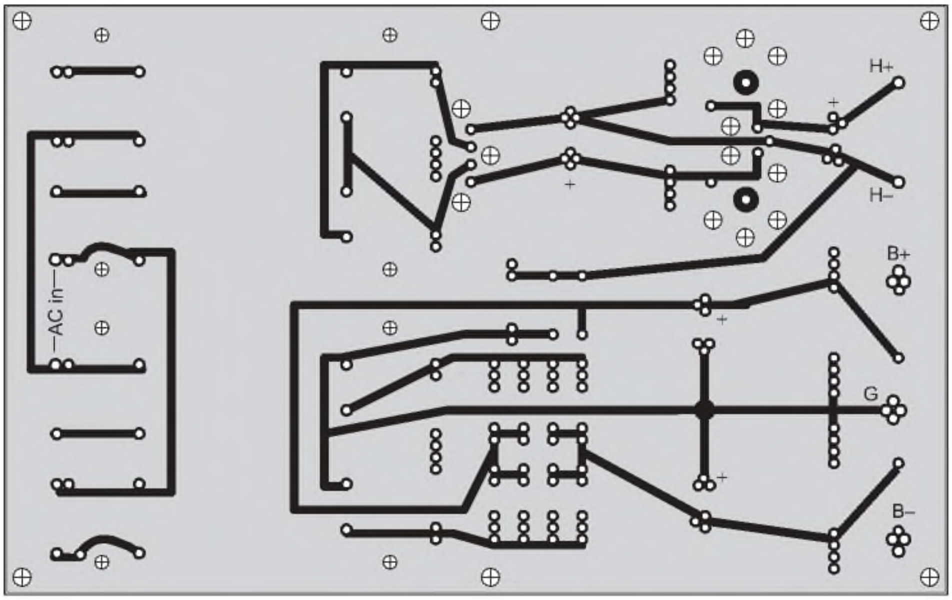


FIGURE 5: Copper track layout of PSU board (67% of true size).

lowers, i.e., three PCBs, which is the power supply's limit; while a fully-buffered three-way requires 16 cathode followers and four PCBs.

In the half-buffered version, although the output is buffered, the tuning elements connect directly to the signal source. Thus the assumption is that the filter will be fed from a low-impedance source, such as a line amplifier or CD player output, not a high-impedance source, such as a potentiometer.

Using the half-buffered version allows two channels of two-way crossovers (up to third-order) or one channel two-way fourth-order filters to be accommodated on one PCB. Three-way crossovers require cascading PCBs or using one PCB per channel for filters of up to third-order. While cascading two PCBs adds more active devices in the signal path, it does offer better attenuation of low-frequency signals for the tweeter.

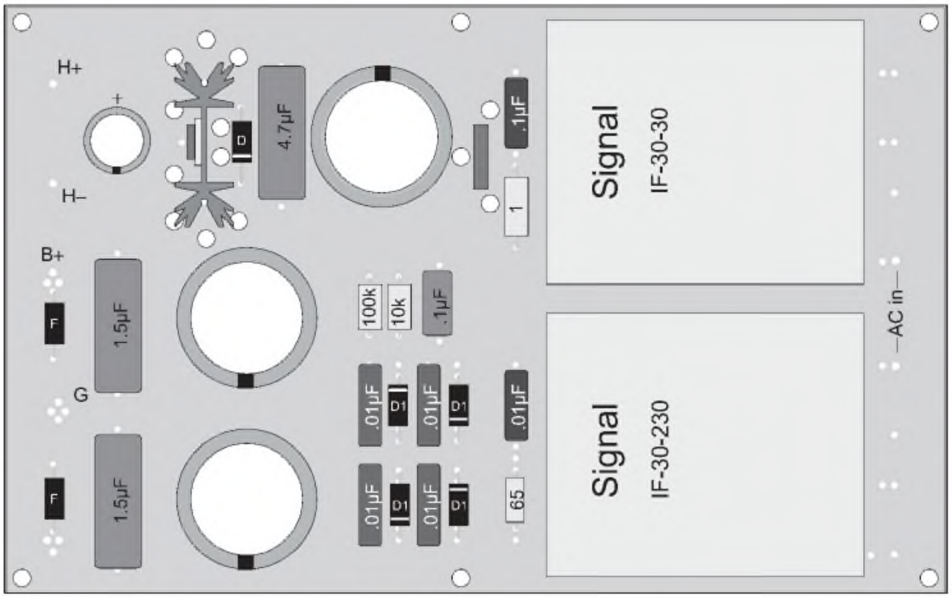


FIGURE 6: Component mounting plan of PSU board (67% of true size).

For example, given crossover frequencies of 500Hz and 5kHz (second-order, 12 per dB octave filters), the first PCB splits the frequencies below 500Hz off to the woofer and sends the frequencies above 500Hz off to the second PCB, which then applies the 5kHz split. The result is that the midrange sees second-order slopes at 500Hz and 5kHz, while the tweeter sees a second-order slope from 5kHz down to 500Hz and a fourth-order slope (–24dB per octave) from 500Hz down. This extra low-frequency filtering helps shield the tweeter from low frequencies and is particularly helpful with horn tweeters, as the normally high loading the horn provides to the diaphragm disappears at low frequencies.

CONSTRUCTION

The PCB artwork and component mounting plan of the PSU board are shown in *Figs. 5* and *6*, respectively. One power-supply board can power up to three crossover PCBs. Both crossover and power-supply PCBs can be housed in a single chassis or in separate enclosures. If you adopt the latter approach, you will need an "umbilical cord," which for safety should be shielded and grounded; furthermore, it should end in a female termination (the crossover chassis holding the complementary male prongs) to prevent inquisitive fingers from touching the high-voltage connections.

The copper track layout of the general-purpose crossover board is shown in *Fig. 7*. While the Crossover Design program displays the layout on the crossover board(s) and the part values, a few guidelines are required. All the filter network defining resistors should be ½W and at least 2% tolerance; the filter network's capacitors should be ≥ 160V DC and at least 2% tolerance. The four cathode follower load resistors per crossover board should be 1–2W devices and should be mounted with a small gap (2–4mm) between the PCB and their bodies. The same holds true for the two 3250Ω voltage-dropping resistors, which should be 2 to 5W devices.

The power-supply PCB holds the power transformers and supporting circuitry. The power transformers are quite heavy and could easily be ripped from the PCB during a fall; thus, they should be both soldered and screwed to the PCB. Four #4 self-tapping screws per transformer (along with eight solder joints) will keep the transformer safely in place.

Unlike the crossover PCB, the power-supply board offers only one optional wiring configuration: 115 or 230V AC input. Use insulated wire to produce these and all other PCB jumpers.

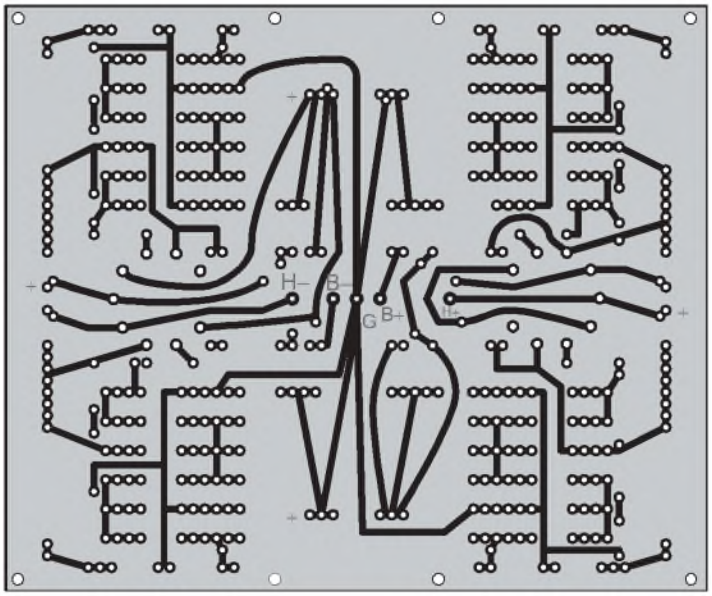


FIGURE 7: Copper track layout of the multipurpose crossover board (67% of true size).

TESTING

Test the power-supply board first without any external connections. I recommend using a Variac®, which allows testing at lower, safer voltages.

The low-voltage portion of the power supply should put out 12V DC with no more than a ±0.5V error. The plus and minus high-voltage portion of the power supply develops much greater voltages when unloaded than when loaded: hence the need to overspecify capacitor voltages. Expect to see voltages closer to 200V than the expected 165V.

First test the crossover boards without the tubes; and then if all voltages are in line with expectations, test the board with the tubes in place. Because the heaters are in series, in the absence of the tubes no voltage will be present across the heater (4 and 5) of the tube socket. But across the two sockets, 12V should be measured. The plate voltage (pin 6) should exceed 165V, and cathode voltages (pins 3 and 8) should be roughly −11V relative to ground. With the tubes in place, plate voltages for either tube should be close to 100V; the cathode voltages, +3V (the cathode resistor's large value effectively defines an autobias circuit in itself).

WARNING

Some of the voltages used in these circuits are potentially lethal. Take all relevant safety precautions to avoid any contact with any point in these circuits while they are in operation.

PLEASE NOTE

This project has not been tested or post-engineered by the *Elektor Electronics* design laboratory.

USING THE CROSSOVER

Quite likely, the loudspeaker drivers will not match in efficiency, requiring an adjustment in crossover output level to achieve balance. You need a white noise generator and a sound level meter for precise adjustment. Or you may choose to do the following method: adjust the level until it sounds correct, then back off on the bass a tad.

If you have any questions or problems, e-mail me at cryquestion@glass-ware.com. ❖

A Center Channel Speaker: A Different Approach

Follow this author's lead to one day constructing your own home theater system. **By Daniel L. Ferguson**

When my job relocated my wife and me to Appleton, Wis., it gave me an opportunity to build the home theater system I'd been wanting for a long time. So, finding a home with a suitable family room layout became one of our top priorities. With that accomplished, the move behind us, and a new 36″ TV finally in place, I began to explore the options for the home theater project.

I was not starting from "square one," because all I needed to complete the system was a center channel and rear surrounds. A few years ago, I built a pair of bi-amplified, D'Appolito-type towers, each containing a pair of 10″ subwoofers. From their inception, I intended that they would some day be the backbone of a home theater system. I spared no expense on their cosmetics and successfully attained a high Spouse Approval Factor (SAF), thus securing their place of honor in our home. Now I needed a center channel speaker design that would complement the towers (*Photo 1*).

M-T-M LIMITATIONS

I have been looking at center channel speaker designs and response curves for quite a while. Today's commercial center channel speakers are almost universally M-T-M layouts that are essentially horizontal versions of the D'Appolito configuration.

Readers of back issues of *Speaker Builder* probably know that a vertical M-T-M configuration can provide outstanding horizontal off-axis response. However, laying one on its side can result in a trough in the off-axis response near the crossover point. Since the crossover frequency is typically within the range of voice and dialog, and a lot of TV watching is done off-axis, it somewhat defeats the purpose of having a center channel speaker. Thus, even Joe D'Appolito has reservations about this application[1].

The other aspect of a "conventional" center channel speaker that bothers me is the somewhat unappealing look of a large, rectangular box perched on top of the TV. It just doesn't look as though it belongs there. On the other hand, if the box is small enough to be unobtrusive, then the bass response suffers and the center just might not blend properly with the left and right channels.

What I wanted was something better in both looks and performance. So here were my design goals: wide, extended off-axis response and an unobtrusive cabinet design that would complement my existing tower speakers.

PHOTO 1: Completed system in author's living room.

PHOTO 2: Assembling the top pieces.

PHOTO 3: Assembling the bottom pieces.

NEW CONFIGURATION

The design I finally settled on is a composite of ideas—many that have appeared in *Speaker Builder* over the years. *Figure 1* shows the enclosure layout. For maximum horizontal and vertical dispersion, I chose a 2″ dome for the all-important midrange associated with reproducing dialog. My thinking was that it should be able to provide both horizontal and vertical coverage, which would be a factor, since the center channel speaker is typically located above the seated listening position.

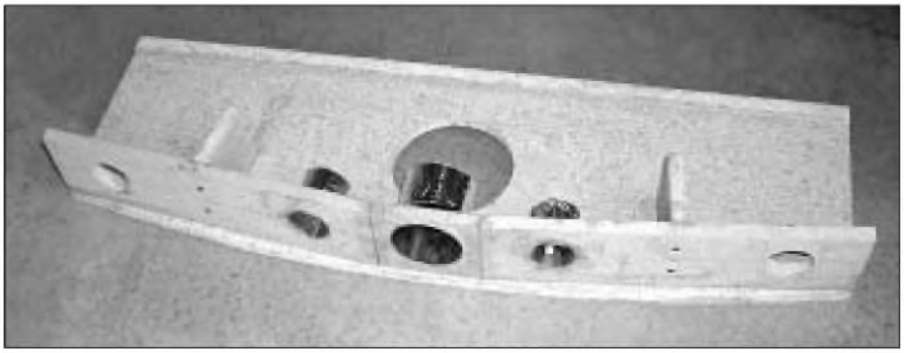


PHOTO 4: Shielding for drivers.

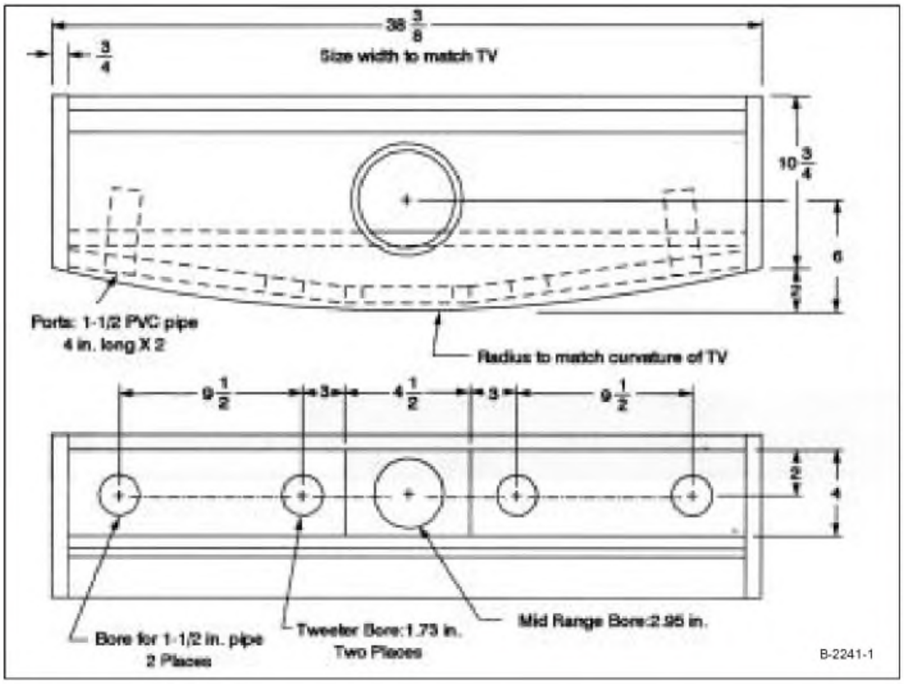


FIGURE 1: Home theater center channel arrangement.

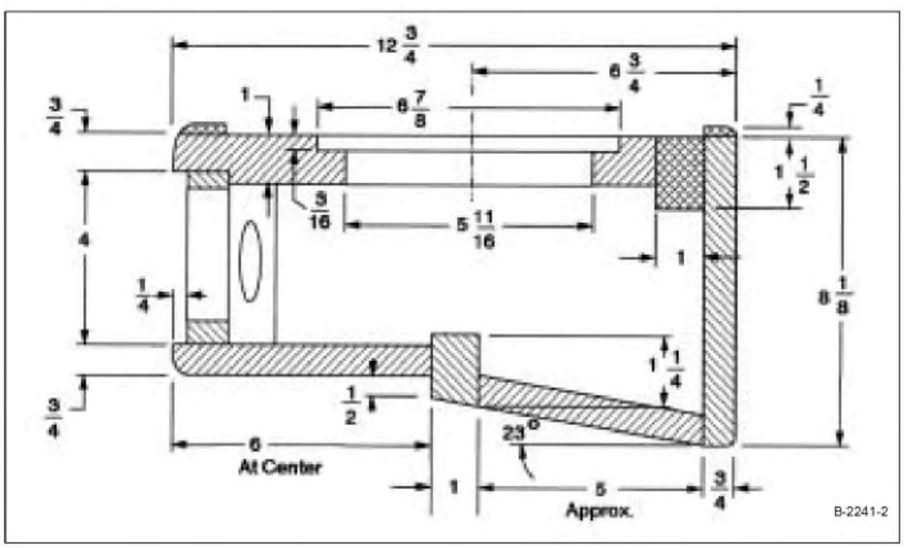


FIGURE 2: Home theater center channel enclosure cross-section.

To ensure wide, even treble response, I flanked it on each side with dome tweeters of similar design. My reasoning was that as you move horizontally from side to side, you are either close to one of the tweeter axes or in the middle where their combined output should be greater than the off-axis response of a single tweeter. For aesthetics, I chose a very low vertical profile, so the only way I could think of to generate similar midbass coverage was to place the woofer horizontally, a la Roy Allison and John Cockroft[2].

The problem with this approach is that for the "Allison Effect" to work properly, the rear of the cabinet should be very close to the wall. In my application, the cabinet is 16″ from the wall. Therefore, I needed to run some experiments to verify the feasibility of the woofer placement before I could commit to the project.

WOOFER EXPERIMENT

To see what type of response curve I could expect from a freestanding, horizontally placed woofer, I installed a 5″ woofer in an oversize test box at approximately the same distance from the wall and height as it would be on the TV. Using my Audio Control SA-3055 real-time analyzer, I took various room response measurements to try to simulate how this setup would perform for a seated listener. The results were amazingly good. While I expected to see the horizontal response fall off and die above 400 or 500Hz, the response actually went out flat to 2,000Hz and beyond.

I then measured at 30 and 45° below the plane of the woofer. While there was some falloff of upper frequencies, it was truly not significant for my intended application. I can only assume that this is due to reinforcement from reflections off the ceiling. With the built-in 360° horizontal radiation pattern this woofer placement provides, I was confident that I had a good match for the dome midrange and tweeters.

DRIVER SELECTION

It was now time to select the drivers. Of course, they all needed to be shielded—especially for a center channel application, since they are essentially in contact with the TV. The choice for the dome midrange was clear: I could find only one—the Morel MDM55. It has an advertised resonance frequency of 380Hz, and I wanted as low as possible crossover point for it.

My design goal was 750Hz, which, as it turned out, was fine. Its off-axis response is advertised to be flat out to 4,000Hz, so the crossover points seemed obvious. This single driver would be able to cover essentially all of the dialog.

For tweeters, I chose the Morel MDT22 for its combination of performance and value. (A candidate for upgrade here is the MDT44, which has more high-end extension, but the cost is triple that of the MDT22.) The MDT22 has a rear chamber like the midrange, which enables it to have a similarly low resonance of 650Hz. This makes the tweeter crossover frequency of 4,000Hz a non-issue.

I looked for quite a while for the proper woofer before settling on the

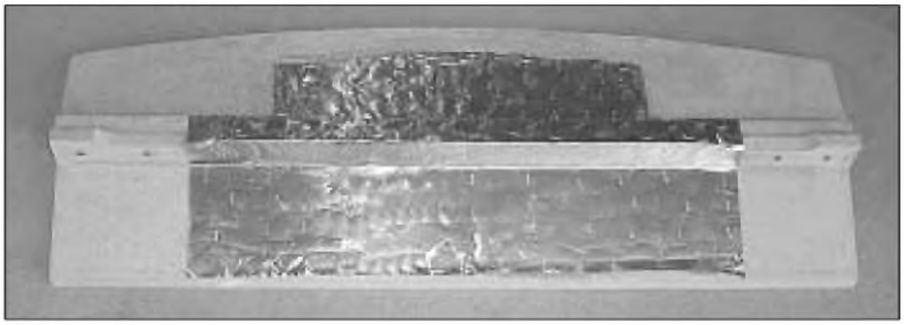


PHOTO 5: Bottom shielding.

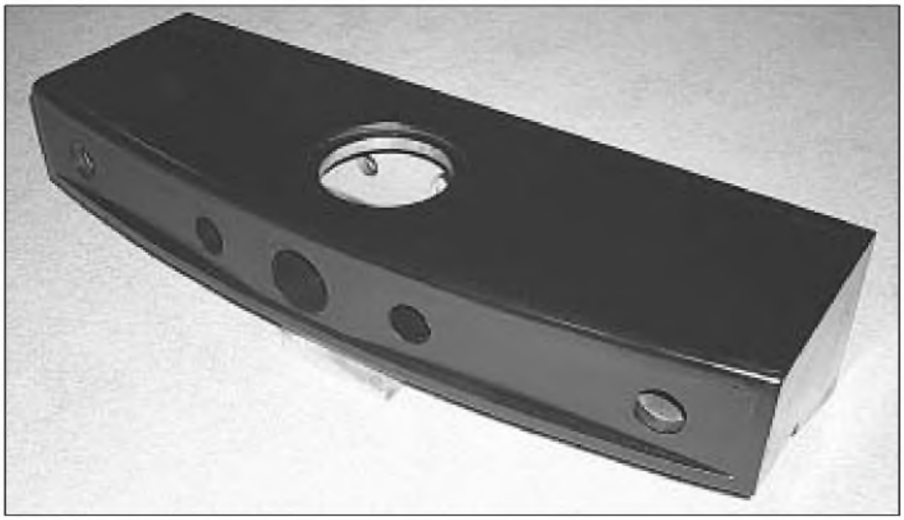


PHOTO 6: Finished enclosure box.

PHOTO 7: Parts to be added.

PHOTO 8: Enclosure finds a home.

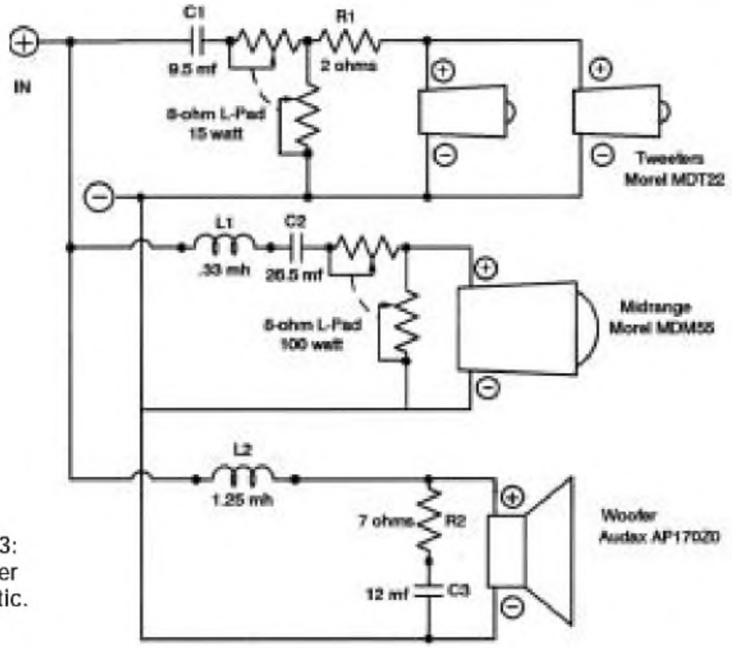


FIGURE 3: Crossover schematic.

6½″ Audax AP170Z0. Like the other drivers, it is intended for A/V applications and has the requisite shielding. It also has a polymer basket, which should reduce magnetic radiation even more. Finally, the Thiele/Small parameters seemed just about perfect. The advertised values indicated possibilities for extended bass response, and the graphs show smooth, flat response out to 1,000Hz.

Looking ahead, all of the advertised response curves for the three drivers were flat enough in their planned operating ranges that I was fairly certain that a first-order crossover network would be adequate.

SHIELDING PROBLEMS

When the box of parts arrived from Madisound (info@madisound.com), I couldn't wait to get them unpacked (as usual). Visually, the quality of construction of each of the drivers appeared excellent. Next I checked for magnetic flux leakage by holding each one in contact with the front of the TV's picture tube and then laid each one on top

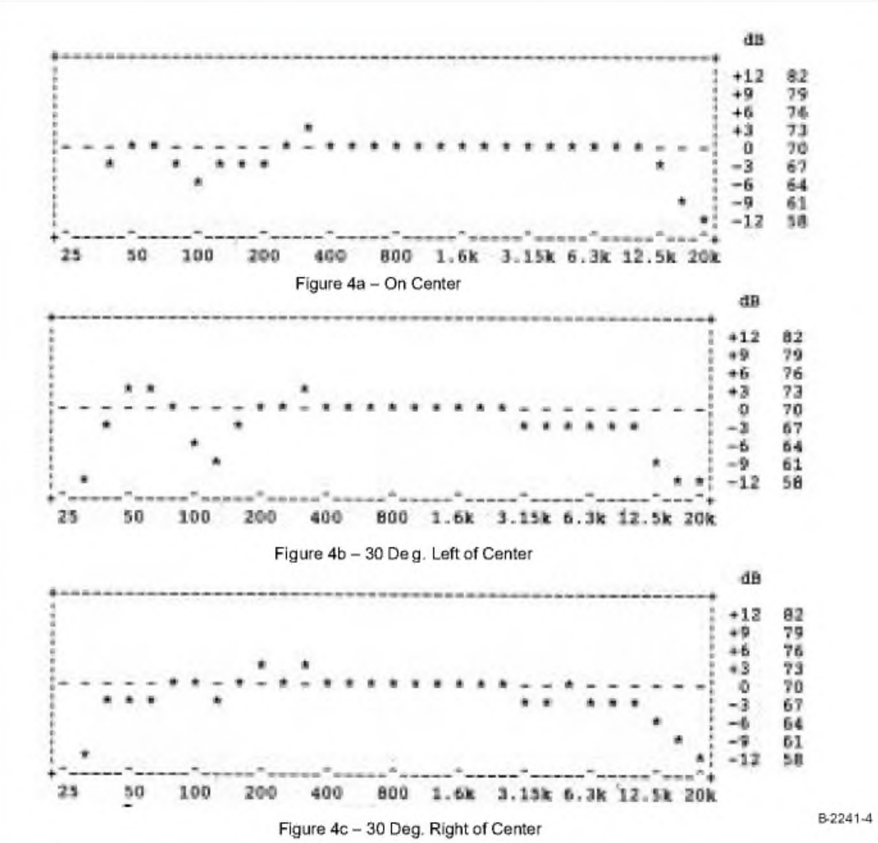


FIGURE 4: Center channel speaker RTA response curves 10° below center.

of the set in the orientation that they would actually be used. There was good news and bad news.

Both the woofer and the tweeters exhibited minimal leakage, but the midrange was a disaster! It caused gross amounts of color distortion in both tests. When I held it in front of the picture tube, you could even hear popping and crackling noises. Clearly not a good sign. (By the way, I found that the best way to check for this is while displaying a fixed test pattern, like the one from a DVD player).

I called Madisound to ask whether they could pull another MDM55 from stock and test it. They placed one in front of a computer monitor and got the same results. They were quite surprised, since Morel clearly claims that this driver is shielded for A/V applications. They offered me a refund, which I declined. This driver was to be the centerpiece for my system, and I was determined to make it work.

SHIELDING RESOLVED

I went immediately to the Internet to see what I could find on magnetic shielding. In short order I found a website for a company that specializes in magnetic shielding products—Less EMF, Inc., (http://www.lessemf.com/). For shielding loudspeaker magnets, their literature recommends two layers of a product called "Magnetshield" (#274), and one layer of "Jointshield" (277). Both come in 4″ wide strips and are sold by the foot. As an added precaution, I decided to line the bottom of the enclosure with "Magnetic Shielding Foil" (276), which is 16″ wide and also sold by the foot. In total, the bill for all the shielding came to $83.30 with shipping.

I suspect most of this is overkill, but I just couldn't take any chances. You may possibly get by with just the two layers of Magnetshield on the three Morel drivers, which is only $4.50 per foot, or a total of around $15 with shipping. It would be worth a try.

As it turns out, installing all the shielding was the one part of the project that was pure drudgery. The metal is thin and sharp and prone to inflicting finger cuts. But in the end it was worth the effort, because it eliminated all visible effects of magnetic flux on my TV. Later on I describe the technique I used in applying it.

THE ENCLOSURE

The enclosure design I chose is somewhat complex to build, and there may be easier alternatives. In any case, it seems to work well and suits its purpose. *Figure 1* shows the overall arrangement and *Fig. 2* shows the cross-section.

I started by making poster board templates of the curvature of the TV and the profile of a cross section that matched the shape of my particular TV. When constructed like this, the box will rest securely on top of the TV with only a non-skid router pad under it to keep it from slipping. The weight does seem a little excessive for the plastic TV cabinet to support, so I added some cleats to the decorative side panels to support the rear of the enclosure, where most of the weight is.

Construction starts by sketching and dimensioning each of the pieces. Note that I chose to make the top out of 1″ thick shelving material because I had some available. This provides the extra thickness needed to flush-mount the woofer and adds to the appearance without grille cloth. I also elected to route a rabbet into the top piece to help hold the three front-panel pieces in position while gluing. If you decide not to flush-mount the woofer, and plan to cover the cabinet with grille cloth, you will need to increase the thickness of the spacers around the perimeter on the top to provide sufficient clearance between the woofer and the grille.

After cutting and fitting the pieces, I glued and screwed the top pieces together as shown in *Photo 2* and the bottom in *Photo 3*. Next came the shielding for the midrange and dome. I started by wrapping the layers of metal around the drivers and taping them in place with some very sticky aluminum foil tape used for sealing heating ducts. I used a Dremel® tool to enlarge the driver holes by trial and error until the wrapped drivers barely fit.

Then I removed the shielding and siliconed them firmly in place in each of the holes (*Photo 4*). In *Photo 5*, in an effort to avoid any possibility of the shielding developing any buzzes, I attached the bottom shielding with a liberal coating of silicone and staples. *Photo 6* shows the completed enclosure after filling, sanding, and applying two coats of black satin spray paint.

Photo 7 shows all the parts laid out and ready for stuffing. I knew that the driver levels would need to be matched to the room experimentally, so I added the two L-pads, which are mounted on the rear of the enclosure.

PHOTO 9: Completed system featuring walnut side panels.

Also shown is the completed crossover, which employs premium inductors and all polypropylene capacitors and is constructed on a blank three-way crossover printed circuit board.

Not shown in this photo are the two 4″ long pieces of 1½″ PVC pipe for the tuned ports. With a gross box volume of approximately $0.85ft^3$, the box frequency measured 48Hz, which was the design target. For the actual measured T/S parameters of this particular Audax AP170Z0, this also results in a theoretical f_3 point of 48Hz.

CROSSOVER DESIGN

I started the crossover design process by installing all of the drivers in the enclosure. First, I ran pink noise response tests on each of the drivers to confirm their sensitivities and suitability for their planned operating ranges. (I protected the tweeters and midrange with suitable series capacitors.) Then I carefully measured each of their impedances using the constant-current method[2].

I spent much time investigating the shape of the impedance curves near the crossover points to see where impedance compensation was needed. While the midrange and tweeter impedances were essentially flat, the woofer needed the usual Zobel network. With the values shown in *Fig. 3* for R2 and C3, I was able to flatten the woofer's impedance to about 6.2Ω above and below the 750Hz crossover point.

Amazingly, the midrange measured an almost constant 8.0Ω throughout the 750 to 4,000Hz range. The surprise here is that the published impedance curve for the midrange is not flat. It shows the characteristic rise you would expect at the higher frequencies.

The tweeters' impedance measured 4.4Ω near the planned 4,000Hz crossover frequency, which was somewhat lower than expected. Since the pink noise tests indicated that the tweeters might not have enough sensitivity if they were connected in series, I decided to connect them in parallel for a combined impedance of 2.2Ω. I inserted a 2Ω resistor in series to keep my amplifier happy. That turned out to be a good choice from a level-matching standpoint.

I calculated crossover values using Ingemar Johansson's LspCAD Lite 3.10 program. The values shown in *Fig. 3* are the nearest match to the calculated values.

For final assembly, I removed each of the drivers, applied silicone caulk to each of their mounting flanges, and reinstalled them. Next, I applied a liberal amount of clear silicone caulk to the bottom of the crossover circuit board and placed it in the right rear corner of the enclosure. I let the caulk cure overnight before proceeding.

The next day I placed the enclosure on top of the TV (*Photo 8*). *Photo 9* shows the completed system with the walnut side panels in place that match the front towers.

SYSTEM TESTING

To measure the system's response, I placed the RTA's mike in a boom stand 1m from the cabinet center and pointed it at an angle 10° below center. This placed it on a line of sight to the seated listening position in my 15′ wide family room. After setting the L-pads for the flattest response (midrange 10%, tweeter 90%), I took three pink noise readings at this elevation—on center and at 30° left and right of center (*Figs. 4a, b,* and *c*).

Considering the ambitious design goals I set for this project, the curves indicate to me that I may have come close to succeeding. Bass response is certainly extended, and the midrange and treble provide the wide coverage I was hoping for. I could have increased the tweeter output for a little more off-axis output, but at higher L-pad settings, the center didn't blend quite as well with my main speakers.

Another possible improvement is to add a bypass capacitor in parallel with R1 to provide some high-frequency lift. A value around 6μF should increase tweeter output by nearly 6dB at above 12.5kHz. The drawback is that the impedance at the top end would fall to 2.2Ω, which may or may not be a problem. At this point, I'm satisfied with the way the system sounds without it.

The one unexpected anomaly is the dip centered at around 100Hz, which could be due to floor bounce. Whatever is causing it is heavily influenced by the room response, since it changes shape considerably depending on where I position the mike. On the right side of the room, which has no openings, the dip goes away.

THE LISTENING EXPERIENCE

So, how does it sound? In a word, quite good. It is crystal clear and non-fatiguing. DVDs with big sound tracks come alive. The dialog is locked into the center, but it blends smoothly with the main speakers so that the center channel doesn't draw attention to itself. I'm well pleased with the final result, and my wife is quite satisfied with the look. She has remarked that it looks like part of the TV.

I've invited quite a few people over to audition the center channel and received rave reviews. One of those reviewers is an old friend who is a non-audiophile and known for his thriftiness. Upon hearing my system, he promptly purchased a surround sound system for his large projection TV, so I guess this thing must work. ❖

REFERENCES

1. D'Appolito, J. A., "Testing the Vifa MTM Loudspeaker Kit," *Speaker Builder* 1/98, pp. 45–48.

2. Bullock, R. M., "How You Can Determine Design Parameters for Your Loudspeakers," *Speaker Builder* 1/81, pp. 12–18.

3. Cockroft, J., "The Shortline," *Speaker Builder* 1/88, pp. 18–22.

PARTS LIST

MATERIALS

1 ea. 1″ particleboard shelving—48″ × 11½″
2 ea. ¾″ particleboard shelving—48″ × 11½″
2 ea. ¾″ walnut side panels (optional)—36″ × 12″
Magnetic shielding from Less EMF, Inc.:
2 ea. Magnetshield—catalog no. 274—4″ × 12″
2 ea. Joint shield—catalog no. 277—4″ × 12″
2 ea. Magnetic shielding foil—catalog no. 276—16″ × 12″
Yellow carpenter's glue
⅜″ staples for shielding
Clear silicone caulk
Water-based wood putty
Black satin spray paint

PARTS

1 ea. 8.0μF polypropylene capacitor
2 ea. 1.5μF polypropylene capacitor
1 ea. 25μF polypropylene capacitor
1 ea. 12μF polypropylene capacitor
1 ea. 2Ω 15W resistor
1 ea. 1Ω 15W resistor
1 ea. 0.33mH air core inductor
1 ea. 1.25mH air core inductor
1 ea. 15W 8Ω L-pad with cover plate
1 ea. 100W 8Ω L-pad with cover plate
1 ea. Crossover circuit board—3-way, 12dB
1 ea. Terminal cup

DC Protection and Switch-On Delay for Power Amps

Easy to build, simple electrical circuit that protects a loudspeaker from unwanted DC, and also provides a switch-on delay function.

By Jenoe Keceli

The idea came as I was repairing a prefabricated transistor high-powered amplifier that did not have any kind of protective circuit. When switched on, these kinds of power amplifiers produce unpleasant static. In this case, or due to an amplifier failure, DC can get to the output, which poses a great danger to the loudspeaker.

The standard solutions for this problem all seemed too complicated and required many components, so I tried to find a simple solution, and I arrived at the idea of the opto-coupler as the sensor element. The application of the opto-coupler significantly simplifies the circuit solution. The protective circuit consists of a DC sensor and a timer that controls the switch-on-delay (*Photo 1*).

The relay carries out the actual loudspeaker switch-on/off. All this happens in stereo, with separate earth-conductors, so that you can also use this solution for amplifiers with balanced output.

FUNCTIONAL DESCRIPTION

Figure 1 shows the schematic of the protective circuit. When you switch on the amplifier, the C3 capacitor slowly charges from the power-supply voltage through the R4 resistor. If the amplifiers function well, the transistor opens and the relay switches on the loudspeakers with a slight delay. In the first second the loudspeakers are protected and the relay switches on only after the operating point of the amplifier has been settled. The delay-time depends on the values of the R4-C3 components.

The signal from the output of the amplifier reaches the opto-coupler input that is functioning as a sensor through an RC filter. The task of the R1-C1 and R2-C2 components is to eliminate the

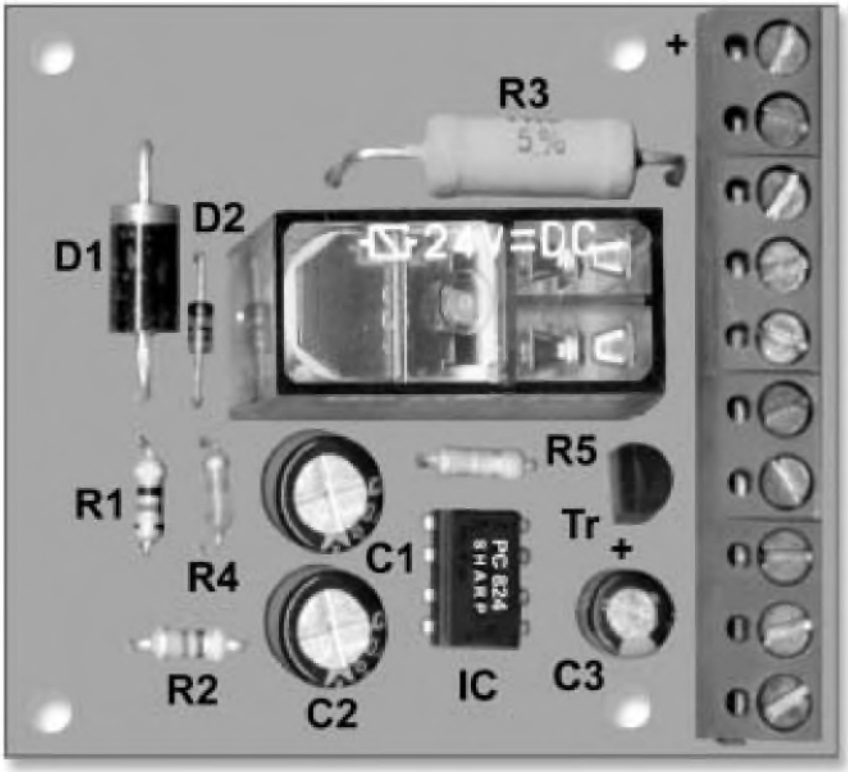


PHOTO 1: The DC protection and switch-on delay circuit.

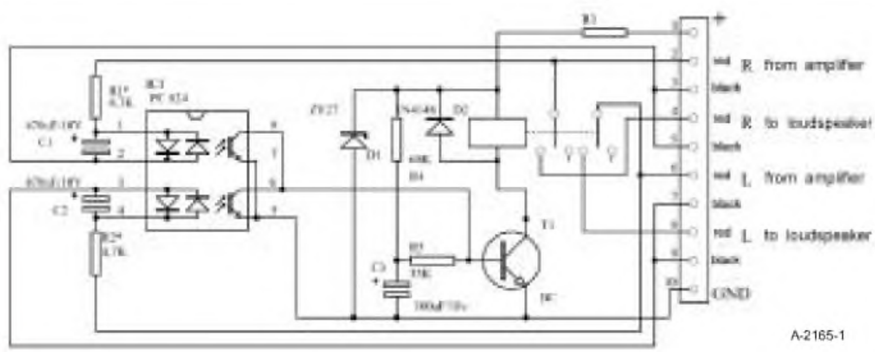


FIGURE 1: The full circuit diagram of the protective circuit. The values of the R1, R2, R3 resistors can vary according to the application. See detailed description.

ABOUT THE AUTHOR

Jenoe Keceli is a long-time electronics enthusiast with 40 years of experience. His main fields of interest are designing and building tube and transistor power amplifiers. He has much experience with sound and disco equipment. He has been publishing his articles in electrotechnical journals. He now runs his own firm, where he builds, repairs, and gives advice.

loud frequency AC. Supposing that the amplifier functions well, there is no DC on the output, so no sign gets to the sensors. The transistor of the opto-coupler does not conduct, hence the relay switches on. In case the amplifier fails, DC will appear on its output; the C1 or C2 capacitors do not hinder the change of the voltage.

The actual diode of the opto-coupler opens its transistor, which bypasses the T1 transistor's base current. As a result, the T1 transistor closes and the relay switches off the loudspeakers. The relays will stay switched off as long as the failure persists.

What is interesting about the opto-coupler is that it exhibits a rectifier behavior on the anti-parallel diodes. The opto-coupler is activated regardless of the polarity of the applied signal on the diodes' inputs. The AC signal that results from normal functioning of the amplifier is filtered out by the C1 and C2 capacitors. It is well known that the electrolytic capacitors are sensitive to correct DC polarity. However, as the diodes

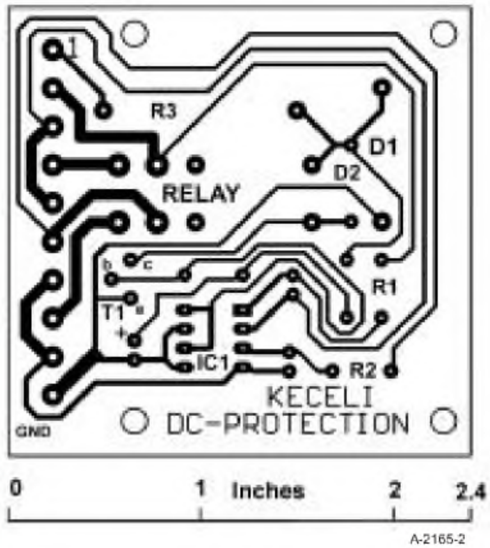


FIGURE 2: Printed circuit board, soldering side.

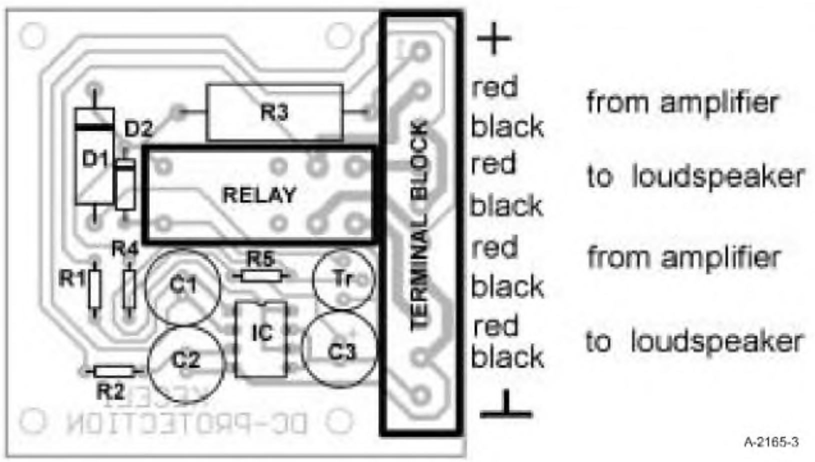


FIGURE 3: The soldering of the components on the printed circuit board. Component side.

TABLE 1
PARTS LIST

R1, R2	Resistor, ¼W 4.7k for a 300W amplifier. For other powers refer to the article.
R3	Resistor 2.2k 4W, for a 72V, 22mA relay. The value of the R3 resistor depends on the power-supply voltage and the relay's coil current rating.
R4	Resistor 68k ¼W
R5	Resistor 33k ¼W
C1, C2	Aluminum electrolytic capacitor, radial 470µF 16V
C3	Aluminum electrolytic capacitor, radial 100µF 16V
IC1	Opto-coupler PC824, LTV824, or equivalent
D1	Zener diode ZY27 (27V 2W)
D2	Diode 1N4148
T1	Universal, low signal NPN transistor, e.g., BC547, BC546, BC548, or equivalent
Re	Relay 24V, 22mA, two contact sets 10–15A, e.g., RP 420024 (Siemens, Schrack), or Finder—Type 4052, or equivalent
Ko	Terminal block—PCB 10 way (0.1″)
PCB	Single-sided PCB

restrict the voltage to low level (remember, DC appears only in case of amplifier failure), there is no risk of damaging the capacitor in this application.

The (lower) cut-off frequency, which is determined by the R1-C1 and R2-C2 constants and which influences the sensitivity of the protective circuit, does not play a significant role in the operation of the protective circuit. Therefore, you can freely choose the values for C1 and C2 capacitors. I advise you to choose the values of R1 and R2 according to the amplifier's output power rating. The suggested value of 4k7Ω is for 300W amplifiers.

You should experiment with selecting the optimum resistor value of R1 and R2 using the actual amplifier. You should begin with smaller-value resistors in normal operation of the amplifier at maximum volume. If the relay does not hold, try to use higher-value resistors. You should increase the value of R1 and R2 as long as the relay becomes stable and the circuit functions well at the full amplifier power. The unstable holding of the relay appears when the value of R1 and R2 resistors is too low because the capacitor still

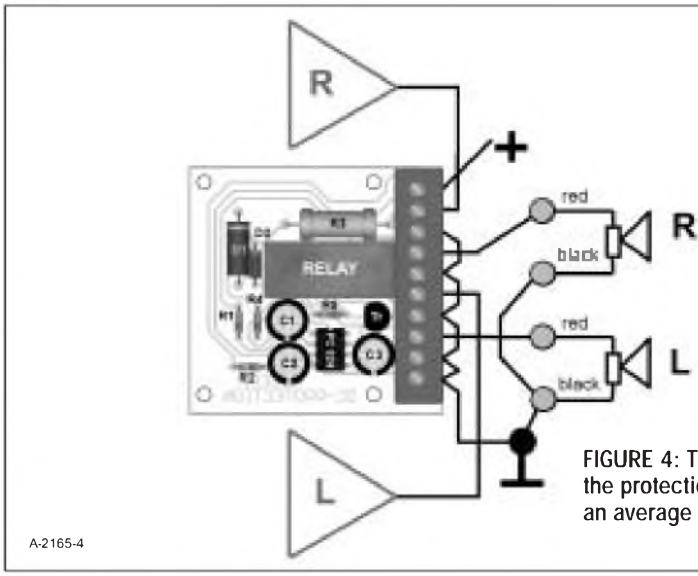


FIGURE 4: The installation of the protection electronics in an average amplifier.

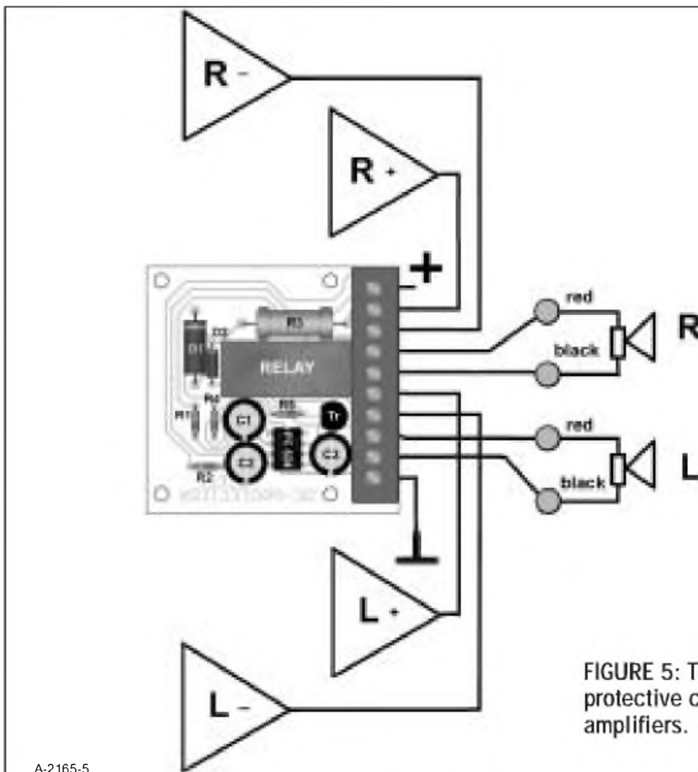


FIGURE 5: The installation of protective circuit in bridge amplifiers.

lets through a small amount of low-frequency signals and the anti-parallel diodes sense this as DC.

Warning: Never choose the R1 and R2 resistors to be less than 500Ω, not even as a test, because it could lead to destruction of the opto-coupler. (Such a low value can only work with low-power amplifiers.)

SELECTION OF COMPONENTS

There are various types of opto-couplers. If you choose a different type, do not forget that it is better to select those with better sensitivity. The T1 transistor can be a universal, NPN low-power device. You should choose the relay contacts between 10–15A, depending on the amplifier; its coil should be 24V. The power-supply voltage of the prototype circuit was 24V.

In this case, application of the D1 zener diode is unnecessary. Instead of the R3 resistor, it is sufficient to use a short wire bridge. As the power-supply voltage is fed from the amplifier, in many cases it would be more than 24V, so it is limited by the D1 zener diode and the R3 resistor to 24V. When the relay drops, the D1 diode takes over the load of the relay-coil, which is why I chose a 2W type resistor for R3.

The value of the R3 resistor is defined by the following simple calculation:

$$R3 = \frac{U - 24}{I}$$

where U stands for power-supply voltage in V, and I is the normal current of the relay coil in A. The formula applies to a 24V relay. Assume that the power-supply voltage from the amplifier is 72V. In this case the value of R3 will be:

$$R3 = \frac{72 - 24}{0.022} = 2181 \cong 2200\Omega$$

Its power rating should be 2–4W. For higher power-supply voltage, besides a higher ohmical value, a higher power rating is required.

CIRCUIT ASSEMBLY

Figure 2 shows the printed circuit board, and *Fig. 3* shows the arrangement of the components.

The printed circuit board should be conventional—single-sided with four mounting holes and a PCB mounting terminal block. Because the thickness of the copper foil did not prove to be thick enough at higher currents, you should strengthen the already broader tracks using soldered wire between the contacts of the relay and the terminal block. Since the R3 resistor might develop excessive heat, I suggest you solder it with longer legs to place it further away from the printed circuit board and to prevent the board from burning.

INSTALLATION

With average stereo amplifiers, where you need to deal with two separate amplifiers, one pole of the loudspeaker is on earth potential, so you can earth one of the poles of the input. Disconnect the red loudspeaker wire and patch through the protective circuit and connect the circuit to the ground. *Figure 4* shows the installation details.

Reduce (with the help of R3) the power-supply voltage of the power amplifier to the supply voltage of the protective circuit. Do not forget about the proper selection of the R1, R2, R3 resistors as described earlier. Occasionally, you can still find bridge amplifiers, especially among the higher-power amplifiers, or in car hi-fi equipment.

In these stereo-channel cases, you have two pairs of amplifiers that are biased in opposite phase. Both wires of the loudspeaker receive an active sign, and there is no earth potential, so bind in the protective circuit according to this circumstance. Free the inputs from the earth potential and connect them to the active signal (*Fig. 5*). You can even install two protective circuits in bridge amplifiers so that both amplifiers get their own protection.

After the installation you can forget about the few components you have used to make the protective circuit. In case of eventual failure of the amplifier, the loudspeakers will be turned off and saved from destruction. Saved, just like you from a financial catastrophe! ❖

The ABCs of Filters, Part 1

If you imagine "pancake syrup" when you hear the word "Butterworth," then you need to push yourself away from the breakfast table and study this article to learn more about filter terminology.

By Richard Honeycutt

I was in the fifth grade, studying the *Radio Servicing Course Book* from Supreme Publications, when I first encountered the term "electrical filter." I found that a filter was a bit of circuitry that allowed some frequencies of alternating current to pass more easily than others.

Today—and for the last 30 or so years—terms that were first applied to electrical filters are used to describe the design and analysis of closed-box, vented, and other loudspeaker systems. While filter terminology is quite helpful to those acquainted with it, for others it simply adds another layer of jargon to a field already rife with obscurities. This article traces filter terminology from very simple passive electrical circuits, through the development of the analogous electrical circuit to represent microphones and speakers, into the culmination of speaker analysis techniques in the work of Leo Beranek and its subsequent codification by Neville Thiele. (Fans of Dr. Richard Small should not feel slighted that he is not mentioned: his work is basically a very valuable systematization of Thiele's, so our story ends before his work began.)

CAPACITORS AND INDUCTORS

Every electrical filter includes at least one of two reactive elements: the capacitor and the inductor. A reactive element is one whose electrical behavior varies in direct or inverse proportion to frequency.

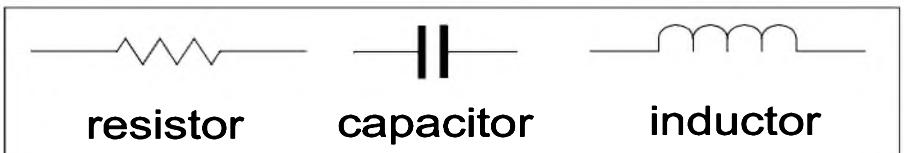


FIGURE 1: Schematic symbols.

A capacitor includes two electrical conductors separated by an insulator. When you apply a voltage between the two conducting plates, one of the plates becomes positively charged, the other negatively charged. This involves one plate having more electrons than protons in its atoms, and the other plate having fewer. When you break the connection to the voltage source, the separation of charge remains. In fact, if no charge is allowed to move, the same number of volts the source supplied can still be measured across the capacitor plates. Thus the fundamental property of capacitance is that it opposes a change in voltage.

If you make an attempt to change the voltage across a capacitor, the capacitor accepts or releases charge in order to prevent the voltage from changing. When a capacitor is charged, it stores energy in the electric field that represents the attraction between the excess electrons on one plate and the excess protons on the other plate. If the capacitor is discharged through a resistance, the current in the resistance produces heat. This heat is the final form taken by the energy that was stored in the electric field.

If you apply a DC voltage to a capacitor, no current flows, except for the initial current required to charge the

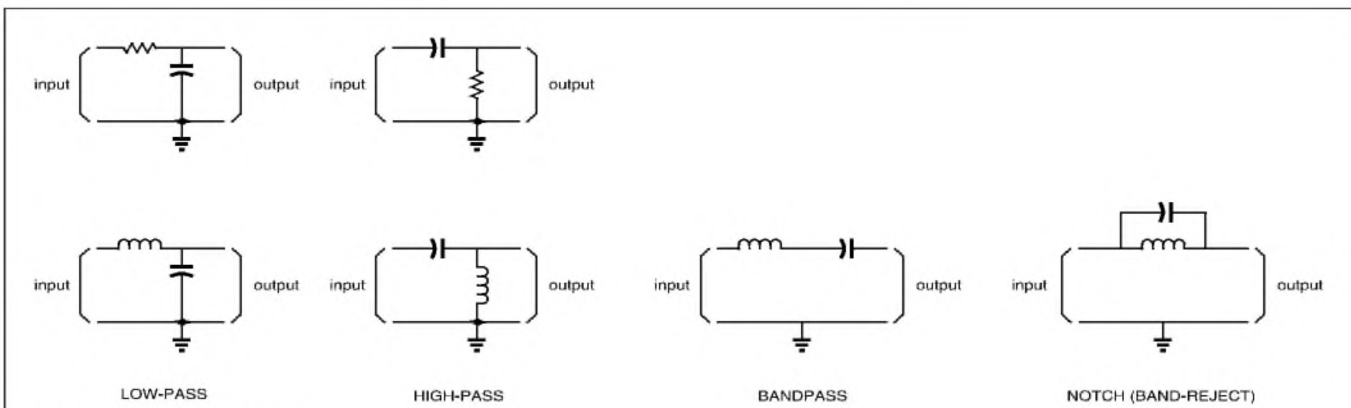


FIGURE 2: Basic filter circuits.

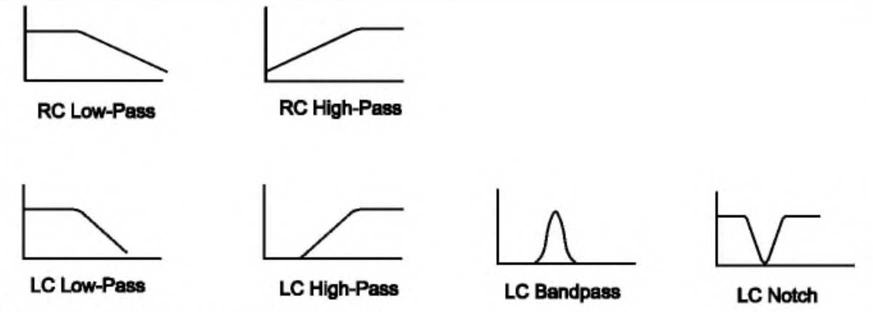


FIGURE 3: Filter responses.

B-2233-3

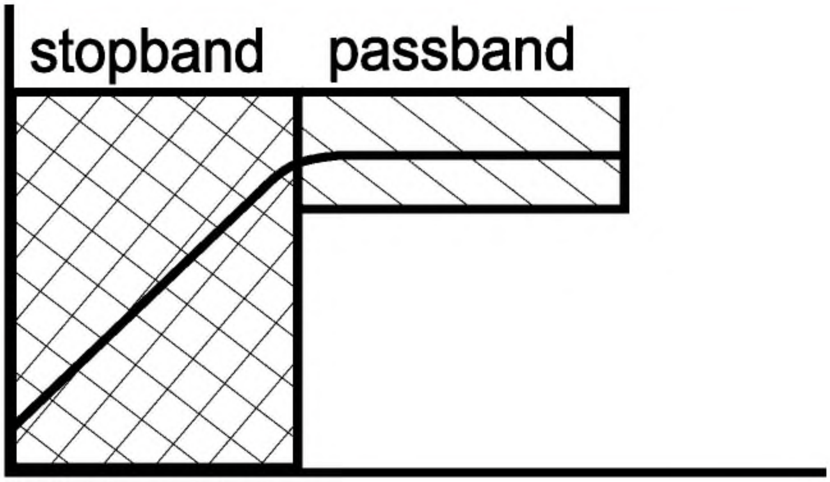


FIGURE 4: Pass- and stop-bands.

B-2233-4

plates. The value (C) of the capacitor is stated in farads.

If you apply an AC voltage to a capacitor, the voltage source is changing all the time, so the capacitor is always charging or discharging. Even though there is no current through the capacitor (how can current flow through the insulator?), there is current in the circuit containing the capacitor. The more rapidly the applied voltage changes in magnitude and polarity, the more current flows. A capacitor's ability to control the current drawn from an AC voltage source is called its capacitive reactance. Capacitive reactance (X_C) is calculated from:

$$X_C = \frac{1}{2\pi fC} \quad (1)$$

where f is the frequency and C is the capacitance.

Notice that capacitive reactance is inversely proportional to the frequency (f) of the applied voltage. This means that if you connect a capacitor in series between an AC source and a load, the capacitor will prevent the load from drawing as much current at low frequencies as it can at high frequencies.

The other reactive element is an inductor, more commonly called a coil. While an inductor need not take the form of a coil, it usually does. When current flows in the coil, it sets up a magnetic field around the coil. While current flows in the inductor, energy is stored in the magnetic field.

When the current stops, the field collapses, inducing a voltage in the coil. The induced voltage can cause a current in an external resistor, producing heat. This heat is the form taken by the energy that was stored in the magnetic field.

If a current flows in an inductor and the current changes, the magnetic field also changes, which induces a voltage in the coil. The polarity of the voltage is such that it opposes the change in current. The fundamental property of inductance is to oppose a change in current.

Since alternating current is always changing in polarity and magnitude, an inductor limits the amount of AC in a circuit. The faster the AC changes, the more limitation. This property is called inductive reactance, and it is calculated from:

$$X_L = 2\pi fL \qquad (2)$$

where *L* is the inductance in henrys.

You have, no doubt, already noticed that capacitance opposes change in voltage while inductance opposes change in current, and capacitive reactance is inversely proportional to frequency while inductive reactance is directly proportional to frequency: in other words, capacitance and inductance are exactly alike and totally different. Two things having this relationship are called duals. Probably the duals most familiar to you are male and female, but we won't pursue that line of inquiry.

RESISTORS

The other building block of passive filters is the resistor. A resistor controls current in any circuit in which it is a series part, and it makes no distinction among DC, low-frequency AC, or high-frequency AC. Naturally, the property of a resistor that does this is resistance.

Resistance, capacitive reactance, and inductive reactance are all measured in ohms. All three properties control the current in a circuit, though reactance affects only alternating current. But resistance functions by means of collisions between subatomic particles in a conductor, and thus changes electrical energy to heat.

We say a resistor dissipates power. Reactances function through storage and release of energy, and they do not dissipate power. *Figure 1* shows the schematic symbols for a resistor, a capacitor, and an inductor.

Any time you connect a capacitor and an inductor, either in series or in parallel, and add electrical energy to the circuit, there will be some sloshing of that energy back and forth between the electric field stored in the capacitor and the magnetic field stored around the inductor. At some particular frequency, this sloshing is a maximum; this maximum sloshing is called resonance, and the frequency at which it occurs is called the resonance frequency.

BASIC FILTERS

Figure 2 shows the four basic filters: lowpass, high-pass, bandpass, and notch (or

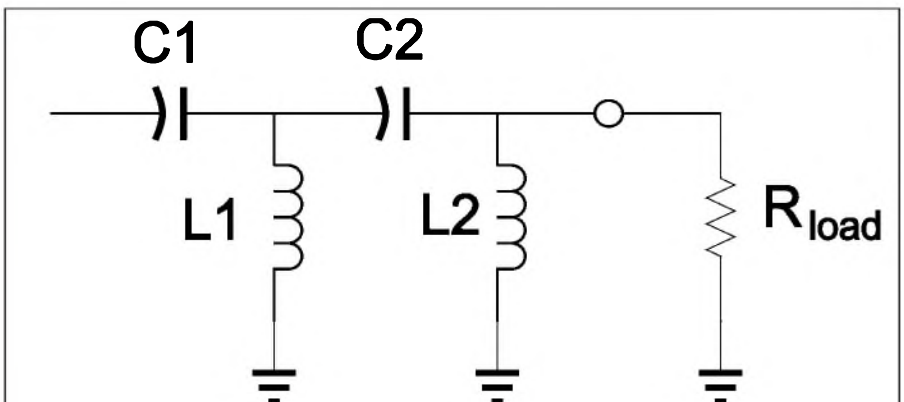


FIGURE 5: Loaded fourth-order high-pass filter. B-2233-5

band-reject). Notice that although each filter must contain at least one reactive component, you may use more reactive components in order to obtain the desired frequency response. Thus the figure shows both *RC* and *LC* versions of low-pass and high-pass filters.

Figure 3 shows typical frequency-response curves for each of the filter types shown in *Fig. 2*. For each curve, the vertical axis is gain, or V_{out}/V_{in}, usually expressed in dB. The maximum gain of any of these filters is 0dB. Thus 0dB is at the top of the graph, with negative dB values (fractional gain) below. Of course, the horizontal axis is log frequency.

Since our ultimate goal is to examine the relationship of electrical filters to loudspeakers, and since the portion of a loudspeaker's response that is related to the box parameters (and indeed the portion that is algebraically predictable at all) corresponds to a high-pass filter, we will concentrate on that type of filter from here forward.

The frequency response of a filter can be split into two segments, the passband (or portion of the frequency range in which signals are passed with little attenuation) and the stopband (or the portion of the frequency range where the attenuation changes). *Figure 4* illustrates these segments.

By common usage, the point at which the passband and the stopband meet is the point at which the voltage gain of the

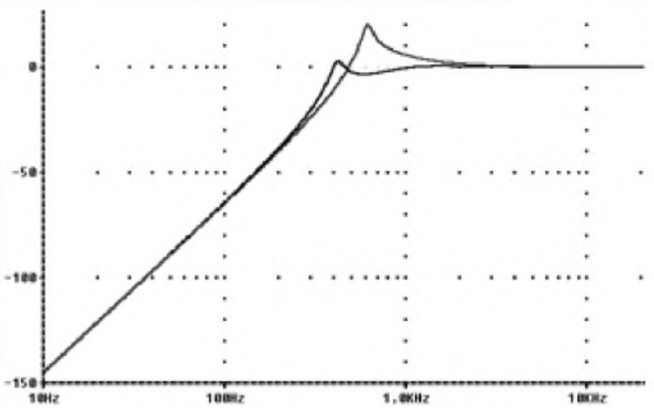


FIGURE 6: Effect of section tuning. (Upper curve is for $C_1 = C_2 = 3\mu F$, $L_1 = L_2 = 20mH$; lower is for $C_1 = 30\mu F$, $C_2 = 3\mu F$, $L_1 = 2mH$, $L_2 = 20mH$.)

B-2233-6

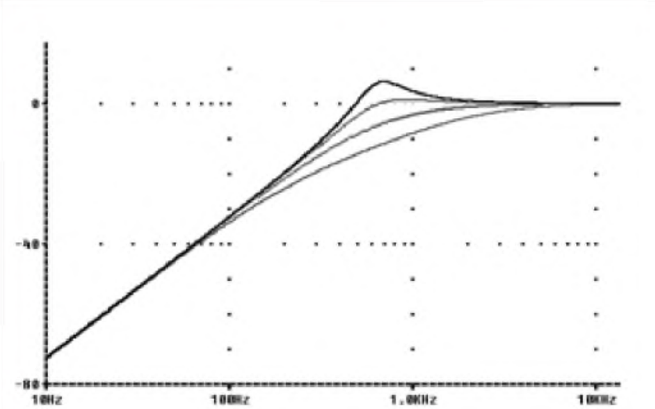


FIGURE 7: Effect of Q on the response of a second-order filter. (Curves represent Q of 0.25, 0.5, 1.0, and 2.0—from lower curve to upper curve.)

B-2233-7

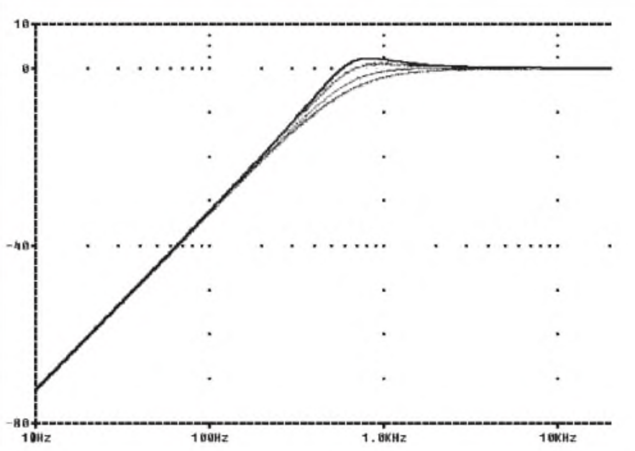


FIGURE 8: Second-order filter families. (From bottom to top: Bessel, Butterworth, 1dB Chebychev, 2dB Chebychev.) B-2233-8

filter is 3dB less than (0.707 times) the gain in the main part of the passband. This point is called by various names: cutoff frequency, critical frequency, or half-power point. (A 3dB drop in power represents a 50% decrease.)

Now look again at the response of the high-pass filters in *Fig. 3*, specifically in the stopbands. Notice that the slope of the response is greater for the LC filter than for the RC filter. In fact, the slope of the stopband response of a low-pass or high-pass filter depends upon the number of reactive devices in the filter. This number is called the order of the filter.

Each of the *RC* filters in *Fig. 2* has one reactive device, the capacitor (they are first-order filters), and its slope in the stopband is 6dB per octave. An octave represents a doubling or halving of frequency, so this means that an RC high-pass filter's stopband response slopes downward to the left, the gain decreasing by 6dB each time the frequency is halved. This characteristic can also be described as a 20dB per decade drop, a decade being a factor of ten in frequency. The LC filters in *Fig. 2* have stopband slopes of 12dB per octave (40dB per decade), since they have two reactive devices (they are second-order filters). If two of the LC high-pass filters shown in *Fig. 2* were cascaded, the resulting high-pass filter would have a stopband slope of 24dB per octave (80dB per decade), and so forth.

A LITTLE MATH

Before we get into the equations relating to high-pass filters, it is important to recall that even though resistance and reactance are both measured in ohms, they cannot be directly added. The reason for this is that resistance is not associated with storage of energy, and is thus independent of time; whereas reactance involves energy storage, which is a function of time. The total current-limiting effect of a circuit containing both capacitance and resistance is called the impedance. Impedance, Z, in a series circuit, is expressed as:

$$|Z| = \sqrt{R^2 + X^2} = \sqrt{R^2 + (X_L - X_C)^2} \quad (3)$$

Actually, impedance and reactance are complex quantities; that is why the left-hand side of the equation is the magnitude of impedance, rather than just impedance. You can see that resistance and reactance are like the sides of a right triangle of which the impedance is the hypotenuse.

Now, notice that if you disregard the input impedance of whatever is connected to the output of either high-pass filter in *Fig. 3*, the input voltage is just divided across two components: a resistor and a capacitor or an inductor and a capacitor. In both cases, the voltage across the capacitor is the output voltage. For the *RC* filter, the voltage gain of the filter is:

$$A_V = \frac{X_C}{\sqrt{R^2 + X^2}} \quad (4)$$

and the cutoff frequency is:

$$f_C = \frac{1}{2\pi RC} \quad (5)$$

For the LC filter, the voltage gain is:

$$A_V = \frac{X_L}{X_L - X_C} \quad (6)$$

and the resonance frequency is:

$$f_R = \frac{1}{2\pi\sqrt{LC}} \qquad (7)$$

The cutoff frequency will be a bit lower than f_R, depending upon the load resistance, which is always connected in parallel with the inductor. Equations (6) and (7) ignore the small resistances that are always present in the voltage source, capacitors, inductors, and wires.

For simple filter circuits, the equations are simple. However, as the order of a filter increases, the voltage-divider approach to developing equations becomes messy very quickly.

Figure 5 shows a fourth-order high-pass filter with a resistive load. Other versions of a fourth-order high-pass filter might also have resistances in series or parallel with the capacitors and/or inductors. But for this relatively simple version, the voltage-gain equation looks like this:

Aside from the critical frequency of the filter, this equation quantifies two other effects. One is the interaction between the two second-order filter sections, C_1L_1 and C_2L_2. Even if both sections have the same critical frequency, their impedances interact in different ways if different component values are used. *Figure 6* shows an example.

The other effect hidden in the equation is the "quality factor" (Q) of the filter. In general, Q is the ratio of energy stored to energy dissipated in a device or circuit. In the case of the filter of *Fig. 5*, the Q is expressed as $R/2\pi fL_2$, where the frequency (f) is generally considered the critical frequency of the circuit. (If the resistance were in series with the inductance, Q would be the inverse of this value.) *Figure 7* shows the effects of varying the Q of a second-order filter circuit.

In order to more easily deal with the complexities of higher-order filters,

$$A_V = \frac{16\pi^2 f^2 L_1 L_2 R C_1 C_2}{R\left[1 - 4\pi^2 f^2 \left(L_2C_2 + L_1C_2 + L_1C_1\right) + 16\pi^4 f^4 L_1 L_2 C_1 C_2\right] + j\left[2L_2\pi f - 8\pi^3 f^3 \left(L_1L_2C_2 + L_1L_2C_1\right)\right]} \qquad (8)$$

mathematicians analyzed the polynomials (such as equation (8)); polynomials representing certain filter circuits, and having certain values of Q, are known by the names of the investigators who first studied them in depth. The more common of these polynomials—and thus the more common families of filters—are Bessel, Butterworth, and Chebychev. (The latter has a variety of spellings.)

Bessel filters exhibit the flattest phase change with frequency, while Butterworths have the flattest response within the passband. Chebychev filters have bumps or ripples in the frequency response, and are categorized by how many dB variations the ripples encompass. *Figure 8* shows second-order filters corresponding to each of these families. Note that there are curves for both a 1dB and a 2dB Chebychev filter.

Actually, these curves represent circuits like the high-pass LC filter shown in *Fig. 2*, but with a resistor connected across the output to represent the load. This resistor's value determines the Q. Varying the value of the resistor changes the filter family. It also changes the critical frequency; in fact, the four curves shown in *Fig. 8* do have different critical frequencies. To keep the critical frequency the same, you would need to change the values of L and C also.

MECHANICAL SYSTEMS

Now that we've looked at electrical filters, let's examine a simple mechanical system: a weight suspended from a spring. In its simplest form, such a system has only two elements: the mass that gravity acts upon, and the stiffness of the spring. Any real system will also have some mechanical resistance (called damping) from motion through the air and from internal dynamics of the spring.

When you lift the mass above the rest position, it gains potential energy due to its higher position in the gravity field. As gravity pulls it down, the mass gains velocity, so that at the rest position, all the excess potential energy it gained from being lifted has been converted to kinetic energy. As the mass falls below the rest position, it slows down as its kinetic energy is converted to the potential energy of the stretched spring.

If there were no damping, the mass would bob up and down forever, as energy sloshes between the potential and kinetic forms. The presence of resistance causes the amplitude of motion to decrease as part of the energy is converted to heat and lost from the system. In this system, mechanical resistance may be represented by the air flow around the mass and material deformation in the spring.

You may have noticed the analogy between the action of a damped mass-spring system with its sloshing of energy between the potential and kinetic forms, and an LRC filter with its sloshing of energy between the electric and magnetic field. There are many other analogous points between these two systems. For example, the momentum of a moving mass opposes a change in direction of motion, just as the magnetic field of an inductor opposes a change in the direction of current.

Because of these similarities, it is possible to model a mechanical system as an LRC electrical circuit element. Inductance can represent mass, inverse capacitance† can represent stiffness*, and resistance can represent damping. The force applied to the system is modeled as voltage, and the volume velocity (cubic meters of air moved per second) is modeled as current.

Just as Ohm's Law states that in an AC circuit, $V = I\,(R + j\omega L - j/\omega X_C)$, you can state that,

$$F = U\left(R_M + j\omega M_M - \frac{j}{\omega C_M}\right),$$

where F = force in newtons, U = volume velocity of the air in cubic meters per second, $\omega = 2\pi f$, R_M = mechanical resistance in newton × seconds per meter, M_M = mechanical mass in kilograms, and C_M = mechanical compliance in meters per newton. Thus you have the equation shown below.

If you cancel the units algebraically and remember that a newton is a kilogram × meter/second, you'll see that this equation makes sense‡.

Modeling mechanical systems in this way allows considerable simplification in analysis. As complex as equation (8) looks, it is nevertheless an algebraic equation. To describe the analogous mechanical circuit without using an electrical analogy would require a fourth-order nonhomogeneous differential equation.

Next month we will apply this filter theory to the better design of speakers.❖

*Stiffness is the number of newtons of force required to compress or expand a springy object by 1m.

†Inverse capacitance is $1/C$. In order to be modeled as a capacitance, compliance, the inverse of stiffness, is used. Compliance is measured in meters per newton.

‡Frequency, of course, is cycles/second, but a cycle is not really a unit, so the units of frequency are 1/second.

$$\text{newtons} = \frac{\text{meters}^3}{\text{second}}\left(\frac{\text{newton}\times\text{seconds}}{\text{meter}} + j\frac{2\pi}{\text{seconds}}\times\text{kilograms} - \frac{j}{\dfrac{2\pi}{\text{seconds}}\times\dfrac{\text{meters}}{\text{newton}}}\right).$$

An Unorthodox Two-Way, Part 1

This easy-to-build two-way speaker features an 8″ mid-woofer. But before you begin construction, take a look at some of the special considerations associated with this design. By Jon Mark Hancock

PHOTO 1: Finished speaker on stand.

This article describes the fruits of my efforts to build a moderate-size speaker, which might be slightly above average in performance for a basic two-way system. I desired a speaker like this both for my own use in secondary systems and for some friends, who required fairly full-range response (i.e., "it's gotta have good bass"). Also, though I have access to a good wood shop, many potential DIY enthusiasts may not, so I hoped to use a modified off-the-shelf enclosure, rather than a build-from-scratch design. This makes construction simpler, especially with regard to having a nice-looking finished speaker, even if you don't have previous wood-finishing experience.

Why an 8″ two-way, which is a fairly unconventional approach? Most two-way systems are typically based on a 5¼″, 6½″, or 7″ mid-woofer, often in an MTM configuration, and are usually configured with a crossover at 2.5 to 3kHz to one of the popular European soft dome tweeters. But, if that's what you're looking for, you won't find it here!

A good 8″ mid-woofer will have some marked advantages in swept area compared with the smaller drivers, and usually a higher X_{MAX}. These qualities extend the dynamic range and low-frequency reach compared with smaller drivers. An 8″ two-way does pose some special challenges; particularly if your goal is to maintain relatively uniform directivity over the critical midrange to high-frequency octaves. This, in fact, required new thinking in the crossover design, and resulted in an approach which adapts ideas I used years ago in the output filter of a Class D amplifier[1].

FOURTH TIME'S A CHARM

This article describes the design and construction of the fourth version of this speaker (*Photo 1*). Why did this project go through so many iterations? The MkI version (*Photo 2*) was fairly successful at meeting many of my initial goals, but the drivers (Focal 8V4211 and MB Quart MBTTR1) were discontinued for DIY availability as I finished the speaker.

In the MkII version (*Photo 3*) I decided to toss my initial cost constraints out the window in the selection of drivers, going with the Eton 8-800 and Accuton C23-6. Unforeseen shortcomings in the performance of these drivers left me dissatisfied with the cost/performance trade-offs. The Eton 8-800 has a woofer cone mode at about 1400Hz visible in the (not published) impedance curve, but not directly obvious in the 1m axial response (its big peak is at 3.2kHz). The Accuton C23-6 did not measure as smooth and extended in response as published curves would have me believe.

The MkIII and MkIV are similar. Both use the HiVi Research M8a woofer, with rear-firing ports. They differ in the front-panel layout and choice of the tweeter. The MkIII uses the Focal Tc120dx2 (another discontinued part), while the MkIV features the Vifa XT25 ring radiator. In this article I focus on the design and construction of the MkIV.

I included some overview about each speaker's evolution in design, as well as the details of constructing the final design. It's been a voyage of discovery about making choices, as well as trying to blend a combination of experience and theory. So I'll describe what I think worked well and what could be improved, and maybe inspire you to build further on this design.

THE BASIC CONCEPT

Two-way loudspeaker systems typically require a series of compromises, judiciously made to reflect the needs of the application and values of the designer. Constraints are imposed by limitations of size, complexity, and expense. First, I'll explain the needs and values I brought to this project.

For a moderate-size enclosure—no more than 40 ltr or so—I wanted reasonable bass extension and power handling, and good tonality and pitch definition. Also, since the majority of music is in the midrange, particularly classical, jazz, and vocals, excellent performance in this area is very important to me. This requires both smooth axial response and fairly flat total room power response, where two-way systems with larger drivers often fall short. This implies trying to keep as constant directivity as possible through the operating range of the mid-woofer and tweeter— not an easy task in most cases.

Transparency (resolution of inner detail) is a characteristic that is sometimes hard to define in simple terms. In this regard, I wanted to minimize "additive" errors due to driver or enclosure resonance. This requirement caused me to seek out drivers developed for pistonic behavior over a wide frequency range.

I lean towards the composite or metal cone constructions over the various plastics that are often employed in loudspeaker drivers. The latter are intended to softly decouple portions of the cone with increasing frequency. Some, such as the SEAS P17RCY, do this fairly admirably, with nary a glitch in the impedance curve, yet they still seem to have a characteristic cone coloration regardless of how the measured frequency response is tailored.

There are other issues to consider about the ultimate "voicing" of the speaker:

- Low frequency voicing—positioning it from adjacent boundaries and the degree of baffle step compensation to use
- Directivity—axial response versus forward power response
- Presence region voicing—to BBC dip or not to BBC dip?

Modern loudspeaker drivers, with their extended response and wide dispersion, tend to provide flatter power response than many components available ten to 20 years ago. And modern recordings tend to benefit from extended wide-range response, assuming a very "clean" playback chain. However, many drivers and speakers have probably received a bum rap from time to time for being too revealing of flaws elsewhere in the system, including source components, electronics, and even the room. That doesn't mean that a revealing speaker is flawed; it means that you must consider the overall system configuration to get the most musical results.

To realize the optimal performance, you must take into account the expected room placement while you develop the design. This means considering speaker placement relative to adjacent boundaries such as the rear or side walls and, of course, the floor. While audiophile conventions acknowledge the desirability of locating the speaker

PHOTO 2: The first version, using Focal woofer and MB titanium dome.

PHOTO 3: The second version with Eton woofer and Accuton C23-6 tweeter.

so that comb filtering from early reflections and its destructive effects on frequency response and imaging are avoided[2], this doesn't reflect where most bookshelf speakers are located in the home.

A more subtle issue is that of the "audiophile dip" (also known as the "BBC dip"), which provides a slight reduction in the response in the presence region. In theory it renders classical and other recordings with close microphone distances with a musical perspective resembling "mid hall." This enhances the apparent sound-stage depth and reduces the hardness that may otherwise result from the interaction of common recording techniques and a lively playback environment. This is often perceived as more pleasant, but may also be less transparent with recordings considered to be audiophile-grade. But how much of your music collection falls into the audiophile category, versus the portion which you simply love to listen to?

That these goals might be met—at least to a significant degree—was reaffirmed for me when listening to a pair of Avalon Eclipses at some friends' last summer. Not coincidentally, I was inspired to mimic aspects of the basic configuration of the Eclipse, which is also an 8″ two-way speaker. Conversations with the Eclipse's original designer revealed two key points to making this work. First, a low crossover frequency is required for the mid-woofer to stay out of its cone modes. Second, the tweeter choice and crossover are critical to integrating the two acoustically while maintaining adequate power handling.

PICKING THE WOOFER

Why use a single eight instead of the more typical dual 6½″ or 7″ drivers? A good 6½″ or 7″ driver will typically cost almost as much as an 8″, but with only about half the S_D, or working piston area. Also, if you desire low distortion and higher output in the low frequencies, X_{MAX} and efficiency trade-offs are issues. A few 8″ drivers are in the 6mm X_{MAX} range; so, eliminating the MTM configuration from my choices provided either a very significant cost reduction or, conversely, the opportunity to use better quality drivers, but still led to a difficult search for a suitable mid-woofer.

In selecting the woofer, my goal was to find a driver with an exceptionally clean midrange up to 2kHz, while having a Q_{TS} in the range suitable for either a ported enclosure or critically damped sealed enclosure (Q of 0.5). I also desired a reasonable X_{MAX} (5–6mm or better) and a tolerable price, preferably under $100. I looked at many drivers, comparing their published response and impedance curves (watch for wrinkles in the impedance curve indicative of cone breakup modes) and evaluating the possible LF alignments using Unibox[3], a very nice freeware LF modeling spreadsheet for Excel developed by Kristian Ougaard.

I initially selected the Focal 8V4412, based on these parameters and its cost under $100. I had prior experience with some other 8″ woofers, including SEAS and Scanspeak, but I thought the Focal was a well-balanced choice. I tried two more drivers before "finalizing" the mid-woofer choice used in the final versions. Naturally, when I picked these drivers, I didn't anticipate that by the time I finished building the first set of cabinets, Focal would discontinue the woofer and replace it with one with very different T/S parameters.

THE MAGIC OF METAL?

Among 7″ and 8″ drivers, SEAS Excel magnesium cone drivers are very well regarded and maintain pistonic operation to a fairly high frequency. I was somewhat dissuaded, though, by their high prices, and also by the large variation in response above 1kHz, from –6dB to +12dB in the range from 1500Hz to 5kHz, for the W22. This presented a challenge with the crossover design that I wasn't sure I was up to confronting.

Another 8″ metal cone driver, the Hi-Vi M8a, with an aluminum/magnesium cone looked like a reasonable candidate, due to its more moderate cost (under $100) and a response characteristic that appeared easier to manage. The response rise at the first breakup mode at about 2.5kHz (*Fig. 1*) was not quite as ferocious as some metal woofers, and I hoped it would require less extreme crossover measures to suppress.

The first tweeter I selected is one I've used frequently over the years, the MBTT1 from MB Quart. This 1″ titani-

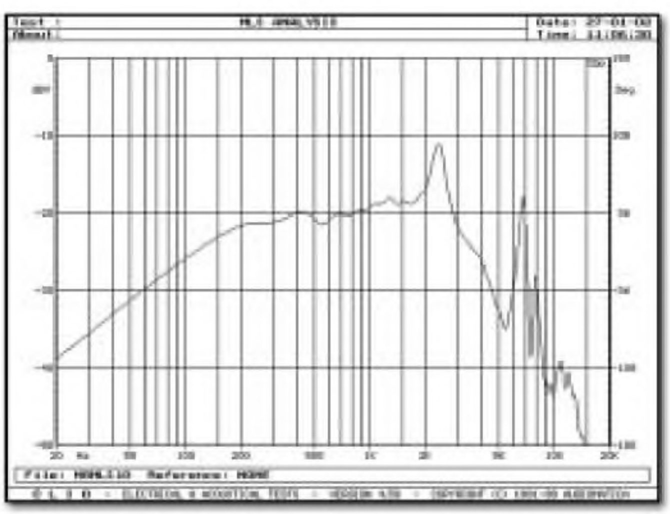


FIGURE 1: MLS plot of HiVi M8a response (gating window rolls off below 200Hz). B-2080-1

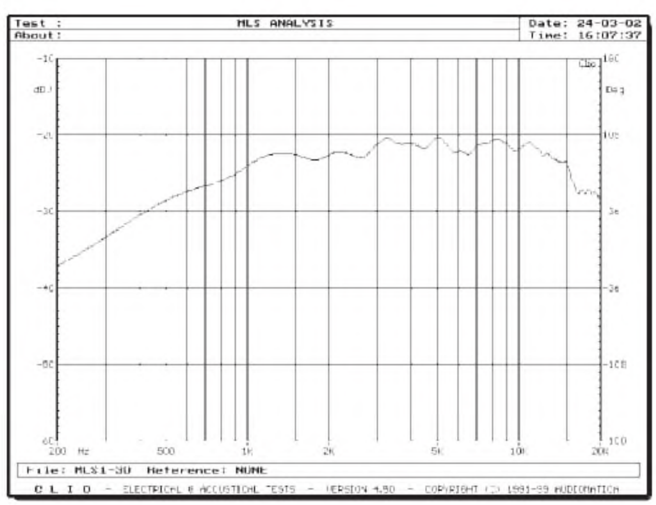


FIGURE 2: MLS plot of Vifa XT25 15° off-axis. B-2080-2

um dome tweeter has a very compliant surround, high efficiency due to a dual magnet design, and detailed, extended high-frequency response. Unfortunately, with change in ownership of MB, their tweeters appear to be no longer available to the DIY market.

In the case of the MkIV version, a friend kindly loaned me a pair of Vifa XT25s to look at. I hadn't seriously considered them because of some "theoretical" questions I had about their design, as well as a possible issue with the published response curve. But the actual tweeters test well and sound even better—especially the off-axis response, which I found to be smoother than on-axis.

Note that the response graph shown in *Fig. 2* is 15° off-axis! I've found better correlation in general with my perception of the sonic quality of a tweeter with its off-axis measured performance, in the range of 15 to 30°, than for the on-axis results. If this perception is correct, it's probably because of the impact on the total power response in the listening area.

LF DESIGN AND SYSTEM VOICING

Defining and voicing the low-frequency design must be based on the intended room placement. Room gain due to adjacent boundaries is a fact of life. Some speakers utilize it very explicitly—good examples are the designs from Roy Allison in the 70s[4] and professional studio monitors intended to be built-in flush to a front wall. This type of mounting controls the radiation load of all the drivers, generally resulting in a nominal one-half pi to one-quarter pi space, which increases the output by between 3 and 6dB in most cases, compared with anechoic radiation into a full "sphere."

Most home speakers (except built-in wall speakers) are not mounted or used in this way, so the design process becomes a bit more complicated. It's possible to calculate this effect and the impact on response, which depends on the distance from boundaries. For years I've used a straightforward document in MathCAD—concepts expounded by Roy Allison—to calculate the boundary reinforcement and the interaction which occurs with various spacing ratios. You can approximate the response in an average rectangular room just by including the effects of three adjacent boundaries (front wall, floor, and side wall, for example). Numerous commercial programs such as SoundEasy, LspCAD, and RPG Room Optimizer, offer this facility in conjunction with other functions.

Consider what might happen if you use the HiVi M8a driver in a sealed box of 32–34 ltr. This will result in a Q_{TC} of about 0.585 (just slightly over a Bessel alignment of 0.577), with an f_B of 47Hz and an f_3 (where the response is down 3dB) of 59Hz. How does that fit in with the room boundary gain?

SPEAKER PLACEMENT

First, look at what happens when you loan the hypothetical speakers you just "virtually" built to my daughter. She takes them into her bedroom and will

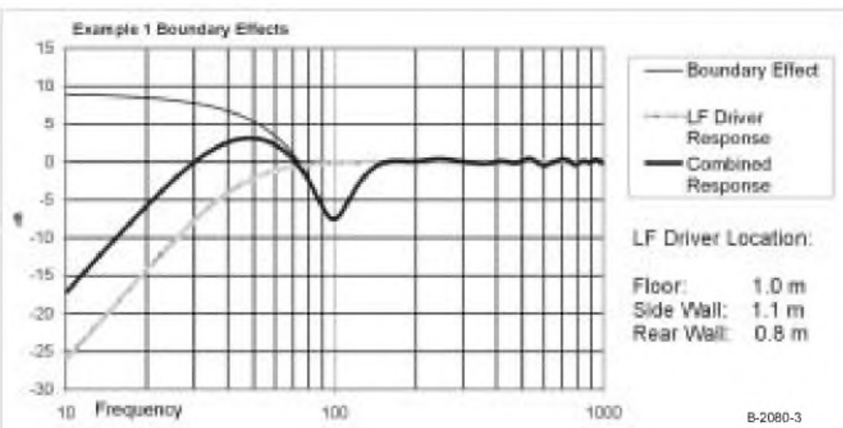


FIGURE 3: MathCAD sealed box Q_{TC} 0.585 analysis first location.

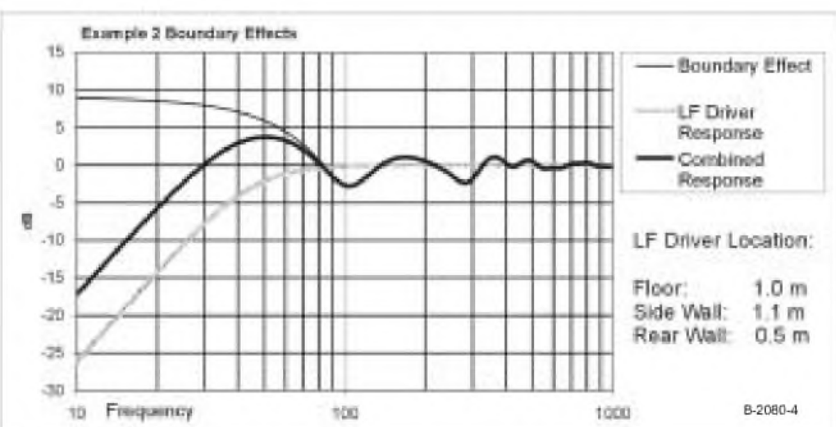


FIGURE 4: MathCAD sealed box Q_{TC} 0.585 analysis second location.

likely put them on the bookshelf by her bed; the woofer for the left one, for example, is 1m from the floor, 0.8m from the back wall, and 1.1 from the side wall. Let's look at how this placement affects the radiation resistance and the frequency response into the room—ignoring for the moment issues such as room modal response.

I've calculated this and shown it in *Fig. 3*. Not a pretty picture, is it?—not only a big hump in the 30–50Hz area (which she liked with her music), but also some unexpected dips in the response, with the upper mid bass suffering a 7dB dip. It would be easy to imagine a casual "audiophile" listener accusing this system of one-note bass and poor pitch definition.

Next, move the speaker around a little bit (*Fig. 4*) closer to the rear wall (actually, more typical), and while the dip at 100Hz is not so great, you have more of a ripple, and a new problem in the lower mid at 300Hz. No wonder there's not much pitch definition on the bass guitar or standup bass. Yet, this speaker has a theoretically well-damped bass alignment.

If you want to develop synergy between the speaker and room, you must avoid placements that introduce excessive nulls and peaks, and make good use of the room to extend the low-end response of the speaker. The placement and LF transfer function should be matched to each other.

Actually, that's not as complicated as it seems, because I've found there are two main factors to consider. First, there is a very simple ratio (the golden mean, 1:1.618) that you can use for the boundary spacing conditions. You can further investigate the application of the golden mean in the reference material on the Cardas website[2]. Using this ratio between the dimensions distributes the peaks and nulls and smoothes the response.

The other factor to consider is the absolute dimensioning so that the LF reinforcement is complementary to the speaker system rolloff. In *Fig. 5*, I show suggested distances for an LF system with damped low-frequency extension similar to my design goals for this speaker. I obtain a nice response by repositioning the speakers so that the acoustic origin (woofer) is at the following distances: 1.96m from the rear wall, 0.75m from the floor, and 1.21m from the side wall.

The implication of this is that for optimizing a box speaker with a given LF profile, there really is such a thing as too small a room . . . unless, of course, you design for placement at the boundary— in which case there's no room gain to mess up the low end or to reinforce it compared with the anechoic response.

We designed this same system—in a sealed box with a critically damped Q—for a friend. You can place it in a similar manner, but then the bass is a bit "light" (*Fig. 6*). It still goes fairly deep, and if you desire response closer to flat in the 50 to 90Hz region, reducing all the distances in proportion will move the room boundary support up in frequency.

I used Unibox for both evaluating drivers and selecting the box volume and alignment. Because I wanted low-frequency extension below 40Hz and adequate power handling for playback peaks around 100dB, a ported enclosure seemed a good choice—I've noticed some similar-size sealed systems have a tendency to develop a bit of congestion at playback levels over 90dB, and I hoped to avoid that through driver selection and enclosure design. The mid-woofer I ultimately selected, the HiVi M8a, has a reasonable combination of X_{MAX} (6mm) and Q_{TS} (0.43), though as a result it gives up a little in sensitivity in order to gain LF extension.

ENCLOSURE CONCEPT

I have access to a fairly complete woodworking shop, but I found after design calculations that I could construct the enclosure by using a modified Woodstyle W123REV enclosure. This is a reverse aspect ratio version of the original Woodstyle 123 box, which has an initial internal volume of 48 ltr. This enclosure is available from several sources at reasonable prices, so I

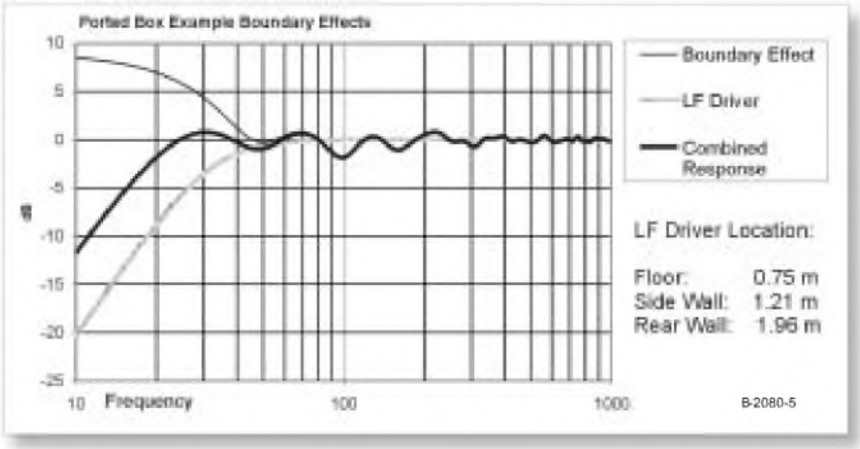


FIGURE 5: MathCAD analysis for ported "optimal" location.

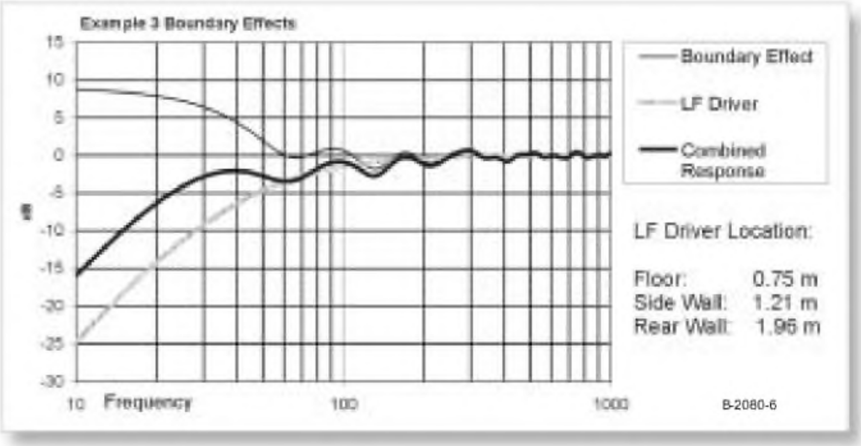


FIGURE 6: MathCAD analysis for critically damped sealed cabinet in same "optimal" location.

proceeded using that path, since you could build this with as little woodworking equipment as a saber saw and drill. Note that considerable enclosure volume of the W123 cabinet includes bracing, the driver volume, and of course, the port.

Figure 7 shows the concept developed for this enclosure and the planned driver positioning of the MkIV. This perspective shows the extensive bracing used to make the enclosure more rigid and better damped. Less obvious, perhaps, is the ¼″ sub-panel added to the front of the woofer and port area—also to thicken and strengthen the area of the cabinet with the largest holes—and the extra rear panel.

While I usually prefer a port exiting towards the rear of the enclosure, the recipient of the first pair (a longtime friend) planned to place these in a way that made this a poor choice. For the MkIII and MkIV, I implemented a rear-exiting port (*Fig. 7*) and substantial changes to the bracing.

While building the first pair I frequently heard the question, "Why so many braces in a cabinet this size?" Though ¾″ MDF is a good starting point for cabinet construction, it takes little in the way of a "knuckle test" to reveal its shortcomings— you'll find fancy instrumentation isn't necessary! Mark Wheeler wrote an interesting series in 1999 *Speaker Builder*[5], in which he discussed the effects of different wall and bracing constructions, including the effects of materials and choices of adhesives. His experience mirrors mine, in that dense hardwood bracing is more effective than most other types, and that two-part hard-setting adhesives, such as epoxy, produce the best subjective clarity.

In the first two versions I used edge-on braces, but in the later versions, I essentially lined the enclosure interior. Here are some simple guidelines:

- Use 1 × 3 or better oak reinforcement for all large holes, such as the woofer.
- Use sub-panels behind the baffle to acoustically isolate the tweeter from the woofer back pressure.
- Reinforce the front panel with a hardboard (HDF) ¼″ sub-panel, which lifts the driver plane closer to the flush front of the grille cloth frame.
- In the MkIV version, I used vertical 1 × 6 panels to reinforce the sides (and similar panels on the top and bottom).
- Brace the long walls by joining with a cross brace.
- Damp the remaining wall vibration with acoustical treatment.

For bracing, I used red oak, which is readily available from home improvement stores at moderate cost. Its stiffness is over twice that of MDF. I attach the braces using slow-setting two-part epoxy. Besides its strength and working time, another advantage of the epoxy is the ability to make fairly good joints even when full clamping isn't possible.

Using extensive bracing in this manner is time consuming, and undoubtedly impractical for a commercial product. But, for the DIY constructor, who is typically using premium quality parts, the extra investment of time and modest cost pays real dividends.

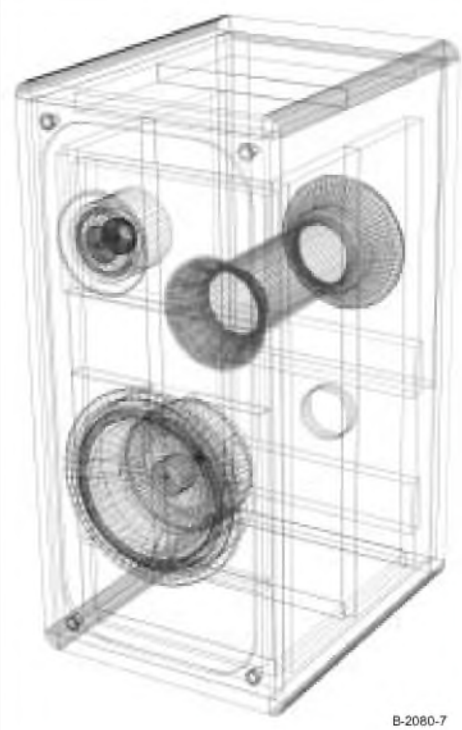


FIGURE 7: Preliminary cabinet concept.

There's a huge difference in the "knuckle test" between the original Woodstyle cabinet and the modified version, especially for the front panels, which I think are critical because they are the "launching point" for the music.

CROSSOVER CONCEPT

My initial plan for the crossover, as described previously, was for a fourth-order Linkwitz-Riley network. This topology has many desirable characteristics, well suited toward the goals for a wide-range two-way speaker:

- Individual driver responses are –6dB at the crossover frequency, which lessens the driver power compared with B3, B2, or L-R2.
- The complementary phase characteristic of LF and HP through the crossover region maximizes constructive interference.
- The complementary phase characteristic is less sensitive to driver offset, which can easily compensate.

All these characteristics work well in the context of a two-way system, which generally must stretch the performance potential of two drivers to span the wide range of frequencies that must be reproduced for many kinds of music. The first two benefits, especially, lessen the workload on the drivers compared with other choices, such as the second- or third-order Butterworth crossover topologies, since the latter is down only 3dB at the crossover frequency, compared with 6dB nominal for the L-R topologies.

I chose a crossover frequency below 2kHz for two key reasons:

- To avoid upper-range breakup modes for the mid-woofer, even when partially suppressed by cone damping and cone taper design.
- To keep within the range that avoids "beaming" as frequency increases—which is necessary if you desire correlation of on-axis response and 20–30° off-axis response.

The latter is especially desirable because I wanted to avoid the problems in timbre and tonality which occur when, as in most two-way systems, you go from a portion of the frequency range where the tweeter has a wide flare in the response to a relatively "beamy" mid-woofer. Still, I was concerned about whether my goals for uniform response on- and off-axis could be met in conjunction with reasonable power handling and distortion for the tweeter, even with a fourth-order L-R crossover.

Some initial design study with mea-

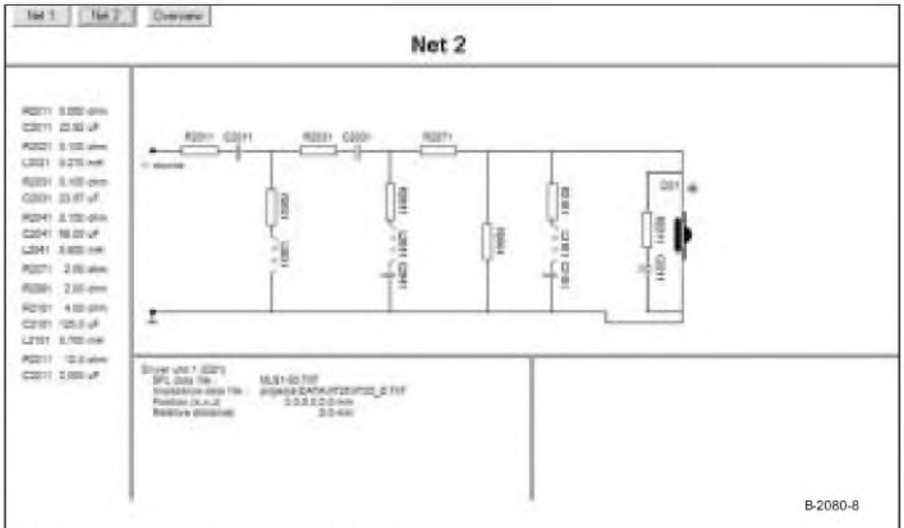


FIGURE 8: High-pass elliptic filter topology and implementation.

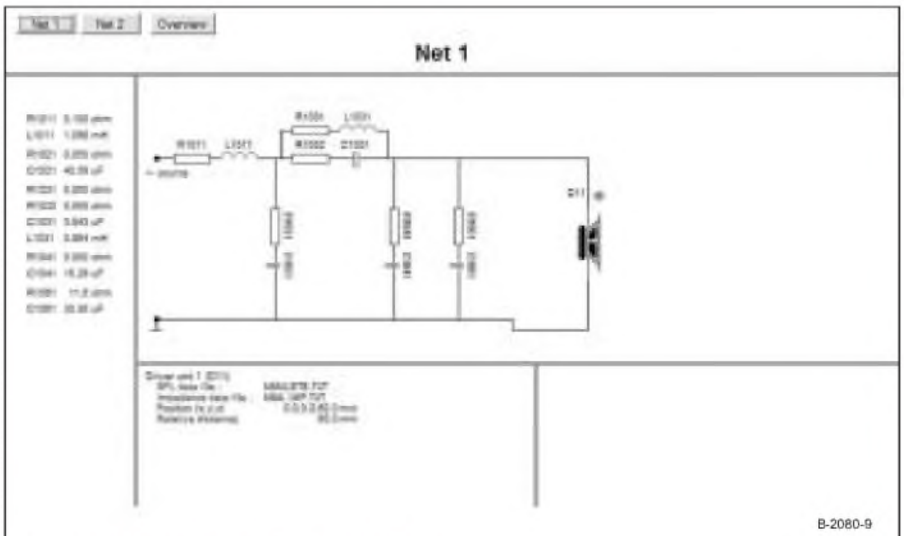


FIGURE 9: Low-pass elliptic filter topology.

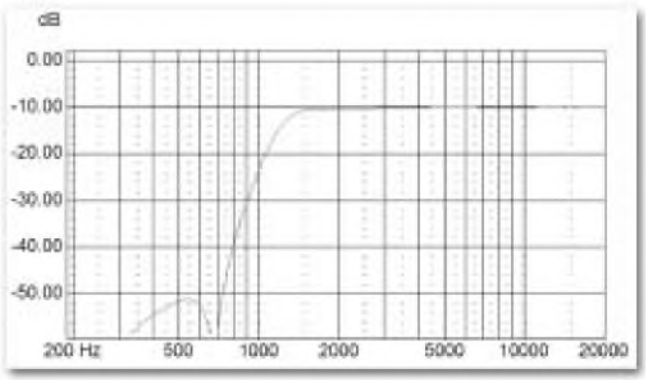
sured driver data imported to LspCAD showed how easy it was to hit a fourth-order acoustic target in principle—and the predicted amplitude response looked fairly good. But despite achieving the acoustic target of fourth-order slopes, the actual electrical high-pass slope was only about 12dB/octave—which saved money on the crossover parts, but raised big questions in my mind about power handling, even for the robust MB tweeter I first considered. While the tweeter acoustic output is down about 15dB at 1kHz, the electrical power rolloff is about half that value—not even 10dB at 1kHz. I thought that a steeper slope crossover was necessary to really make this work, but I didn't want to use a lot of components. Fourth-order L-R requires many parts, as it is.

I considered some filter techniques I used in the late '80s and early '90s with Class-D amplifiers. I used modified Elliptical-Cauer filters to implement the output filter of a Class D amplifier (*Figs. 8* and *9*). Elliptical-Cauer filters belong to a class of filters referred to as finite zero filters.[6] In standard implementations, they have equal ripple in both the passband and the stopband. The group delay characteristic is not very different from standard "audio" filters, and for the same number of components, they have very narrow transition bands compared with other filter types.

Furthermore, they have a null in the response near the transition frequency which can be tuned somewhat relative to the nominal design for the passband; it was my intuition that this could be helpful for certain crossover filter problems.

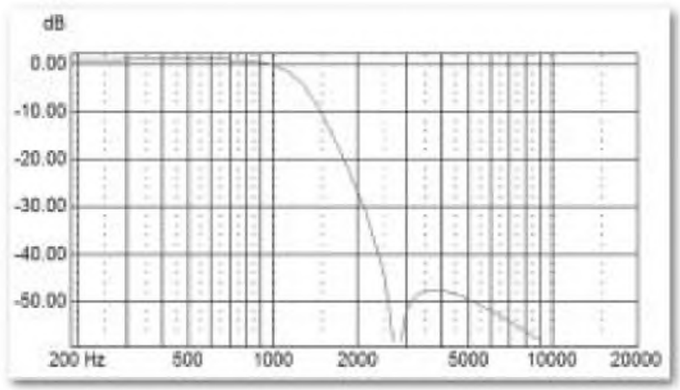
The key to making them work for audio is adapting the filter alignment to the requirements of the application. In the Class-D amplifier, I tuned the transfer function null to the primary switching frequency of the Class-D half bridge and adjusted the filter elements to achieve a suitable Q at the corner frequency within a reasonable variation of loudspeaker load impedance.

These thoughts led to some experimentation with an elliptic network, based on a fourth-order L-R topology, by adding a capacitor to both the woofer network and tweeter networks to provide the finite zero element. With adjustment of the filter coefficients, I found it was possible to emulate various filter orders. I investigated emulating sixth-order L-R and eighth-order L-R networks in the passband and corner frequency.



B-2080-10

FIGURE 10: High-pass electrical response with 10dB pad.



B-2080-11

FIGURE 11: Low-pass electrical response without baffle step compensation.

With the latter it was feasible to achieve 48dB/octave slopes and minimize the peak of the "bounce back" after the zero null to below the –40dB level. This can be seen in *Figs. 10* and *11*, which show the response profiles for the "ideal" transfer function into a resistive load. Some modification of the actual transfer function is employed to achieve any necessary frequency contouring such as baffle step compensation.

Note that in these figures I've expanded the typical vertical dynamic range by covering 10dB/div extended down to –60dB! For good fill behavior, the net acoustical response should track the target filter function at least down to –20 to –24dB.

This filter topology appears to have several interesting properties when applied to loudspeaker networks.

- The null in the transfer function notch can be tuned to driver resonance—for example, to the upper range breakup frequency of a metal or Kevlar cone driver, or the fundamental resonance of a tweeter—possibly eliminating the need for an additional notch filter.
- Trade-offs in corner frequency rolloff rate and null frequency can be adjusted with some independence, allowing some variation for adjusting group delay and compensating for driver offset on the listening axis (i.e., woofer acoustic origin behind tweeter).
- The overlap frequency range between drivers is very low, which facilitates minimizing destructive interference.
- Variation in frequency response above and below the primary listening axis is reduced, which results in a more nearly flat room power response.
- Out-of-band loading on drivers is significantly lower, even compared with fourth-order L-R crossovers.
- Increase in component count is minimal compared with that to implement a true sixth- or eighth-order L-R network.

Initially I suspected there might be issues with higher group delay in the crossover region—which begged the question, would it be audible, and would it be significantly deleterious in comparison to the benefits this approach seems to offer? As it turned out, the predicted group delay for the filter alignments I used didn't differ greatly from conventional fourth-order L-R filters. As for subjective qualities, there's really only one way to find out, of course: build it and listen.

In fact, one of the tests I usually perform on a new design is to listen to the low-pass section by itself, without the tweeter or midrange connected. If there is edge or grunge in the upper end of the woofer passband that isn't sufficiently attenuated by the crossover network, it may be masked (to a degree) by the upper range driver when connected, but it will reveal itself quickly when listening to the low-pass alone. I found that this unconventional crossover combined with a metal cone woofer gave excellent results in this regard, sounding like a steep active filter, without audible artifacts that would only be partially masked by the tweeter. I think this contributes greatly to both the transparency and ease of the finished speaker sound.

Figure 11 shows the resulting low-pass network schematic after optimization to an eighth-order L-R transfer function in the range of 200Hz to 2.5kHz, for a nominal corner frequency of 1.25kHz. C1031 is the capacitor that converts the output filter section into an elliptic filter. The resistors associated with the reactive elements only model the parasitic loss of the component.

The high-pass filter for the tweeter is shown in *Fig. 8*. C2041 in series with L2041 converts the conventional fourth-order topology into an elliptic filter in the output section. The tweeter padding network is formed by R2071 and R2081, setting the gain of the L-pad.

The series resonance of the tweeter is compensated by an LCR Zobel formed from the network of R2041, L2041, and R2041. This network is critical to maintaining a nearly resistive impedance in the area of tweeter resonance. At one point, I'd hoped that by tuning the null of the HP section to the tweeter resonance that I could eliminate this section, but it turned out that the component sensitivity of the elliptic, (which is high) and the impedance interaction made it impossible to get the correct corner and slope without the Zobel network.

Part 2 details cabinet design and construction. ❖

TEST SYSTEM AND DESIGN SOFTWARE

Measurement System

Measurements were made using the PC-based CLIO system from Audiomatica, with a B&K 4133 external microphone and preamplifier. Distortion measurements are made with an HP8903A, and component measurements with an HP4192.

Design Software

LF box and driver modeling was done using Unibox 2.33 and 3.0, while crossover modeling and optimization and acoustic simulations were done using LspCAD.

REFERENCES

1. J. Hancock, "A Class D Amplifier Using MOSFETs with Reduced Minority Carrier Lifetime," *Journal* of the Audio Engineering Society, Sept. 1991.

2. George Cardas, Loudspeaker placement guidelines under "Insights," http://www.cardas.com.

3. Christian Ougaard, Unibox 3,4; website: http://www.danbbs.dk/~ko/ubmodel.htm.

4. Roy F. Allison; "The Influence of Room Boundaries on a Loudspeaker Power Output," *Journal* of the Audio Engineering Society, vol. 22, pp. 314–320 (June 1974).

5. Mark Wheeler, "Navigating Speaker Design, Parts 1, 2, and 3," *SB* 6/99, p. 38; *SB* 7/99, p. 34; *SB* 8/99, p. 32.

6. Richard C. Dorf, *The Engineering Handbook*, IEEE Press, pp. 1204–1212.

Phase Inverter Design

The split-load phase inverter has been around for half a century, and with applications like this, there's no reason it shouldn't be in use 50 years from now. **By Adriaan Hammer**

During the past 30 years I have built and tested many designs; I find that the split-load inverter is superior with respect to simplicity and performance. It's therefore not surprising that this design has been favored by manufacturers and amateurs alike for half a century!

A typical version that I have successfully used in various projects is shown in *Fig. 1.* This circuit consists of two tri-

PHOTO 1: Modified version of the original Williamson using the inverter circuits mentioned. Output tubes are Electro-Harmonix EL34s, but you could also use 6L6s. The transformers are homemade and frequency response is flat up to 100kHz (without overall negative feedback). Power output is 11W per channel. This year I will remove the power transformer and construct an external power supply using TV damper tubes as rectifiers, large chokes, and a DC, slow rise, filament supply.

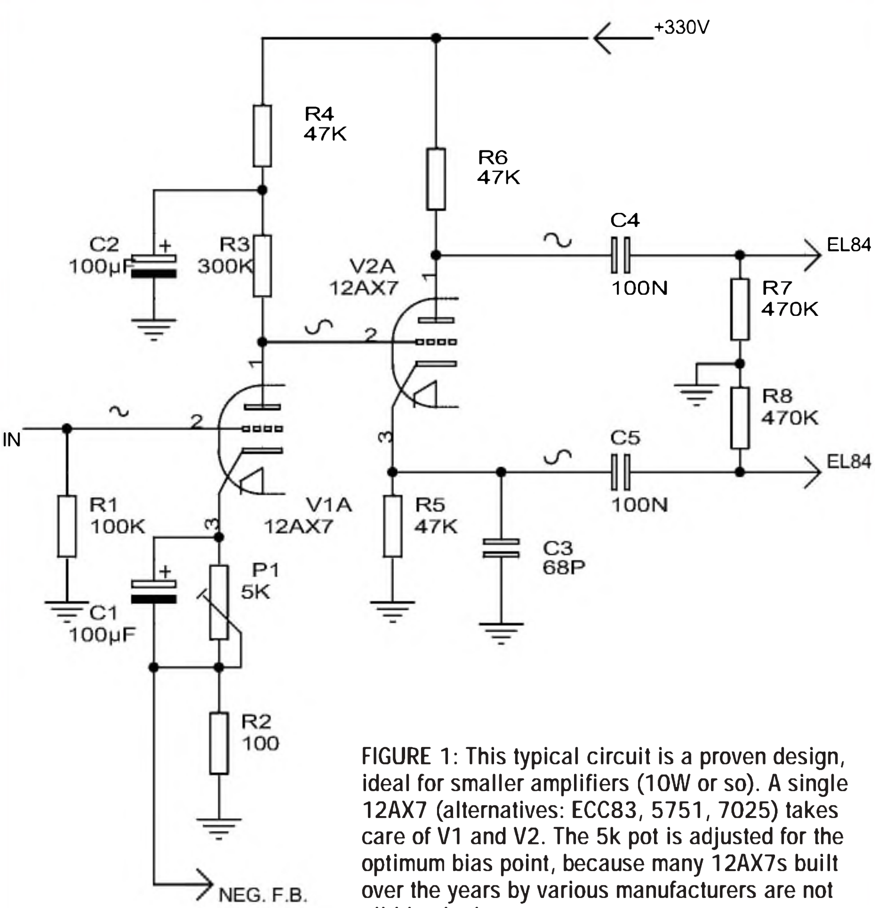


FIGURE 1: This typical circuit is a proven design, ideal for smaller amplifiers (10W or so). A single 12AX7 (alternatives: ECC83, 5751, 7025) takes care of V1 and V2. The 5k pot is adjusted for the optimum bias point, because many 12AX7s built over the years by various manufacturers are not all identical.

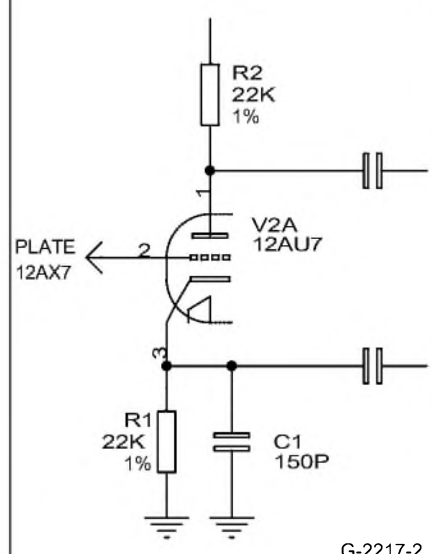


FIGURE 2: By replacing the phase inverter V2 with a 12AU7 (alternatives: ECC82, 5814, 6189), a lower impedance triode, the circuit now can function properly in larger amplifiers (20 to 30W). The 12AX7 is still desired for V1, for its large amplification factor. You want all the gain you can get, assuming you'll be including some negative feedback.

PHOTO 2: A professional version of my amplifier using Hammond transformers. In spite of inexpensive transformers, it is a very good performer.

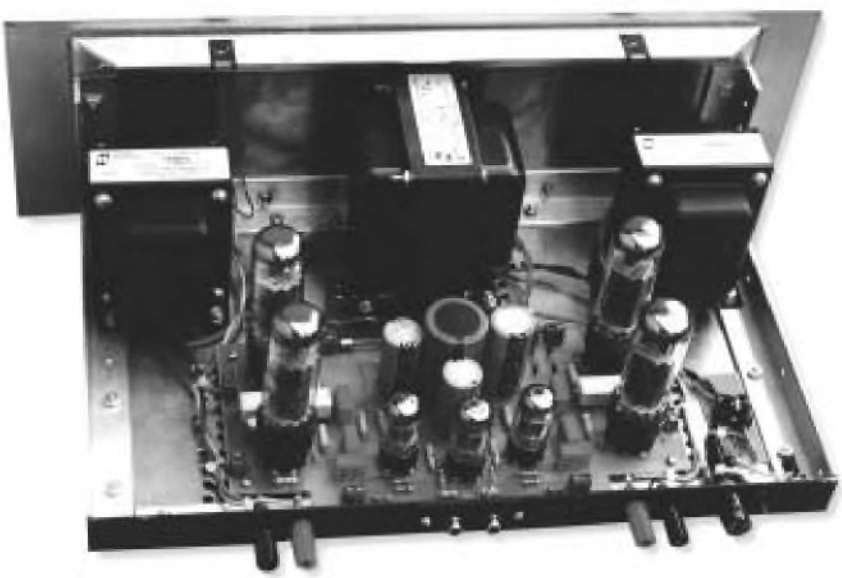


PHOTO 3: Interior of amplifier.

ode stages—a voltage amplifier (V1) followed by the phase inverter (V2). You can use direct coupling between these two stages to improve performance, but the circuit must be carefully biased for proper operation.

The 5k (10-turn) pot serves this purpose; it is adjusted for approximately 90V at the plate of V1. This will place the operating point of V1 in a very linear region and V2 will be biased properly as well. A single 12AX7 (dual triode) is frequently employed for its high amplification factor. V1 provides a gain of about 80; V2 doesn't amplify but divides the signal voltage into two equal halves 180° out of phase as required for driving the push-pull output stage. The small capacitor at the cathode ensures a flat frequency response and perfect balance up to 100kHz.

This inverter is the ideal companion for a pair of 6BQ5/EL84 output tubes operating in ultralinear with no more than 9dB of negative feedback. The resulting amplifier can be driven directly by a CD player, eliminating the need for a preamp.

ABOUT THE AUTHOR

Born in the Netherlands, Adriaan Hammer emigrated to Canada, where he was educated and later worked for 22 years as an aerospace technician. He has been building amplifiers and speaker systems since the age of 12, and a ten-year stay in France enriched his audiophile interests. He likes classical music and jazz, enjoys playing the clarinet, and is active in black and white photography. He lives with his wife in Louisiana.

Output tubes such as the 6V6 and 6AC7/EL34 need larger drive voltages. In this case for V2 use a 12AU7, whose much lower internal resistance will provide double the drive that the 12AX7 is capable of. You will need to adapt some of the component values as shown in *Fig. 2.*

There are specially built dual triodes (12DW7/7247) that combine one half 12AX7 with one half 12AU7 in one envelope. These are ideal for this circuit, but they may be difficult to find and are quite expensive. One alternative is the 6GW8/ECL86; the triode section is equivalent to a 12AX7 and would do perfectly for V1, while the pentode section (configured as a triode) would be ideal for V2.

PHOTO 4: A previous project—phono and line stage preamp, based on the well-known RCA tube manual phono stage with the simple triode stages replaced with SRPP topology. None of my designs use negative feedback. This unit has relay input switching, a slow turn-on DC filament supply and high voltage filtering using a choke and large capacitor. The power transformer is a toroid type to which I added a high voltage winding. (There is a simple technique to accomplish this without too much difficulty.) Tubes used are 12AX7, 12AT7 and 12AU7. All this fits neatly in a small chassis 6 × 15 × 1½″. I'm extremely pleased with this project as well.

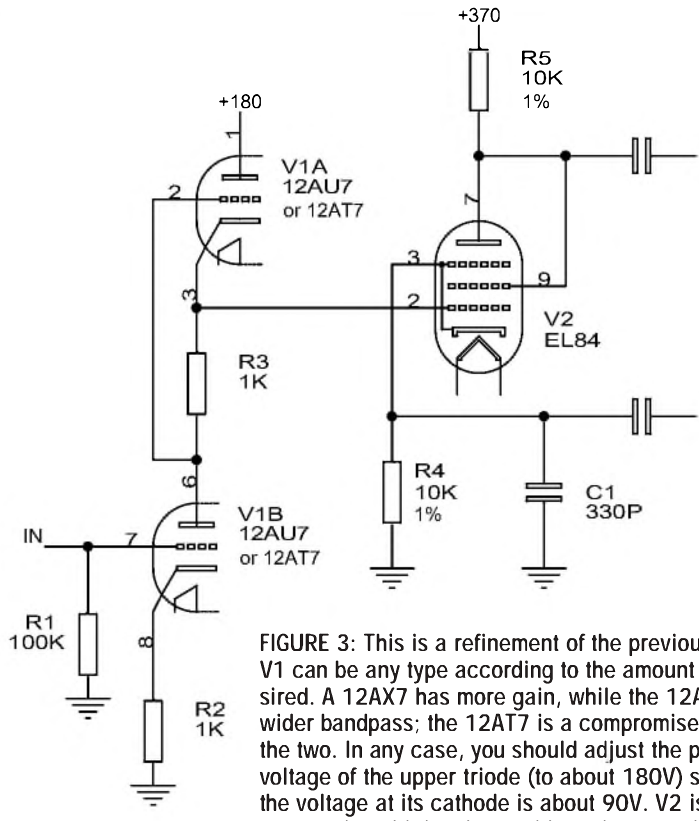


FIGURE 3: This is a refinement of the previous circuits. V1 can be any type according to the amount of gain desired. A 12AX7 has more gain, while the 12AU7 has wider bandpass; the 12AT7 is a compromise between the two. In any case, you should adjust the plate-supply voltage of the upper triode (to about 180V) such that the voltage at its cathode is about 90V. V2 is a 6BQ5 power tube with low internal impedance and is very linear in triode mode.

G-2217-3

My latest version of this phase inverter is now in use as a modification and simplification of the original Williamson amplifier. For V1 I'm using a dual triode in the SRPP configuration providing lower distortion and higher output drive. For V2, I used a 6BQ5/EL84 in a triode configuration; it has low internal impedance and also happens to be very linear and good sounding.

This combination directly drives the output stages of the Williamson, eliminating one amplifying stage. This is possible because the 6CA7/EL34 tubes need less drive than the original KT66s. Also, 20dB of overall negative feedback is dispensed with, reducing the gain requirements ten-fold. The circuit is thus greatly simplified, and the result is a great improvement over an already excellent design. The resulting sound is very refined, detailed, and breathtaking. ❖

Xpress Mail

PRODUCT REVIEW

Your review of the Behringer DSP8024 in the February issue (p. 52) was really a timely and welcome piece. The further I get into speaker building, the more I realize the value of controlling the signal going to the speakers electronically. In the not-too-distant past, using electronic signal control with anywhere near the capabilities of the DSP8024 would have required large sums of money and much greater rack space. This product certainly typifies the huge value increases available to the audiophile in today's electronics.

A few months ago I purchased a DSP8024 for use with my system. The combination of the graphic and parametric controls provides a huge degree of flexibility that is both easy to use and reproducible. The ability to instantly switch between a modified setting and flat assists in determining small changes to the EQ values. Ditto for the ability to switch between two different EQs. I won't go into other details, which were covered in the review.

However, I did run into one problem with the DSP8024 that does not detract from its sonic performance. Rather, it is an inconvenience that I hope will be remedied soon.

The problem is an apparent incompatibility between Windows XP and the Behringer control software that allows you to control the DSP8024 through your computer's MIDI port. I loaded the software and made a connection using MIDI cables. The software did not recognize the DSP8024. Settings were confirmed with tech support at Behringer USA, but to no avail. Behringer in Germany confirmed that they are investigating the situation, but I have not yet received a solution.

Although the DSP8024 represents a great value, Behringer has recently outdone itself with the introduction of a new product, the Ultra-Drive Pro DCX2496. While visiting their website recently, I saw the spec sheet for this new product and was absolutely awestruck. I have never before seen a single product that offered so much in the way of speaker control as the DCX2496.

The list of features is huge, and the performance specs are equally impressive. It offers three balanced analog inputs. One of these connectors can be switched to accept AES/EBU input, which means that other devices such as the DSP8024 can be connected digitally. Another of the inputs can be switched for microphone input including phantom power.

Analog input signals are converted to digital at 24/96, leaving nothing to be desired in the quality area. If digital signals are input, it has an integral sample rate converter to accept signals from 32 to 96kHz. The digital signals are then available for processing in many flavors before being converted back to six balanced analog outputs.

Processing includes adjustable gain; time delay; parametric eq in low-pass, bandpass, and high-pass modes; and crossovers in each output channel in Butterworth, Bessel, and Linkwitz-Riley configurations having slopes from 6dB/octave to 48dB per octave, depending on the type used. Polarity can be inverted independently on each channel, and phase adjustment is possible to assure exact phase matching. If desired, limiters can be used as well as dynamic eq that alters the boost or cut according to signal level.

Now there is no reason why an amateur cannot build a world-class system. Need to determine the best crossover type, slope, and points? No problem.

Don't use passive crossovers. Use the DCX2496 and just enter the different settings you want to test.

You say you don't have input level controls on the amps you are using to power each driver? No problem. Adjust the output gain to each driver in the DCX2496.

Have an arrival time problem from the drivers whose acoustic centers are displaced from each other? Just enter the appropriate time delay into the DCX2496 and you are home free. No need to build those difficult offset driver mounting panels.

Need to correct that small frequency bump in the midrange? Forget the Zobels. You have parametric EQ at your command.

I'm waiting for the DCX2496 to hit the streets because I plan to use it for my system, which consists of an eight driver subwoofer (*Speaker Builder* 5/96), a planar magnetic mid/high frequency speaker (*audioXpress* Jan. and Feb. '01), and recently finished dipole midbass using twelve 10″ woofers. I hope to write up that project for *audioXpress* in the next few months. The DCX2496 will allow me to remove several other electronic components currently in use while giving me much greater flexibility.

So, don't wait. Check out the DCX2496 for yourself at www.behringer.com. Look at the spec sheet, and if you want more info, download the instruction manual. If you get the manual, make sure you're sitting down when you read it, because it will knock your socks off. It will open your eyes to the range of control you have easily available when putting together a killer system.

Tom Perazella
Sleepy Hollow, Ill.

REVIEW RESPONSE

I thought your review of the Adire Audio (AA) HE10.1 in the February 2003 issue (p. 42) was incomplete. The measurement portion from Mr. D'Appolito was very informative, as always. However, a person with Joe's experience should have picked up the XO null on the second sample as the classic out-of-phase tweeter hook-up. It was not apparent that Joe had picked this up in the article and measured the second sample at its face value.

Even though AA had mismarked the tweeter, the null is readily apparent and the reversed phase tweeter is easily fixed. Your readers would be better informed had this error been fixed prior to your actual audition of the speakers themselves. Because of this, there is a lot left to be desired from the "critique" portion of the article. I believe that a follow-up audition is definitely in order.

Also, the listening test did not include pertinent information such as: associated equipment and their familiarity thereof, length of audition (I assume a brief session since the audition by Mr. Colin was at Joe's place), and whether the speakers were broken in. This information is needed to give your readers a complete picture to evaluate the article and the product under test. I am not affiliated with AA but I am interested in building the HE10.1 or something similar to it.

Toan Duong
TDUONG@ladpw.org

As a long-time reader of your magazine, having purchased every issue since its inception, I must take issue with the Adire speaker review. Please let me state that I have no experience with Adire or any of the company's products or personnel. I have accessed the website after reading the current review. My only perception is that they seem to have simple products at a fair price, but I cannot attest to the truth of that perception. My note concerns the conduct of the review.

After reading what amounted to a somewhat negative review, I read the manufacturer's follow-up. It seems that your reviewers conducted the entire review with the tweeter polarity reversed. Please explain how a respected speaker designer with untold hours logged in at the test bench could fail to detect such an obvious anomaly. Please don't tell me that they must review the kit as it comes because that is no help to the reader. Do we have to re-read every review in case such an event occurred previously?

Everyone makes mistakes; however, Mr. D'Appolito's glib and disingenuous response amounted to an insult to all readers. He is the purported expert, and after reading countless frequency-response graphs he should have recognized the error and rectified it. What he wrote amounted to character assassination, and he should re-test the speaker and submit an apology so that the readers know that while he is not infallible, he is, in fact, a gentleman.

I must admit that I am disappointed to the extent I am moved to write this letter, and while I respect most of your endeavors I'm afraid I must take Mr. D'Appolto's reviews with a grain of salt.

John Remson
Massapequa, N.Y.

Joseph D'Appolito responds:

Well, this test report and review has ruffled some feathers. Replying to Toan first, fixing the polarity error was neither simple nor obvious. The HE10.1 as assembled was not biwired. Checking the tweeter connections required that I take the speaker apart. I did not do this in the initial testing as I do not consider it part of the process.

Part of evaluating a kit is to assess the average builder's chance for success. After Adire's comments arrived, I received permission to remove the driver and check the tweeter polarity. This is when I found the tweeter polarity mismarked. I consider this a serious flaw. The poor upper-range response of the HE10.1 was sufficient to mask the polarity error. The average builder would not uncover the polarity error by listening alone.

This leads to the second part of the letter. In an earlier review we did uncover a polarity error in the laboratory testing. The reversed tweeter caused a response dip of some 12dB deep over an octave wide. In this case the polarity reversal was easy to find because the speaker had provisions for bi-wiring.

Incidentally, the tests were run after the listening review, and the reviewer did not pick up the gross response error. Dips in response—even large dips—are much harder to hear than peaks. Listening in stereo with only one tweeter reversed makes it just that much more difficult to detect the error.

Although I did not report on it, auditioning these speakers again after correcting the polarity error did not change my opinion of their sound. The sound is dominated by the 10dB peak-to-peak response variations above 1kHz. Correcting the tweeter polarity did not cure these variations.

Turning to Mr. Remson, I believe his characterization of some of my comments as "glib and disingenuous" is itself glib and disingenuous. There really is no need for that kind of language. You can state your opinion without being insulting. It will give it more weight and better elicit a thoughtful response.

I suggest Mr. Remson reread my final comment. I cited both the strengths and weaknesses I found in testing the HE10.1. There was no "character assassination." It *was a balanced critique. I believe this is what I owe the reader. Incidentally, in a previous issue we tested and reviewed another Adire product much more favorably.*

MYTHS AND INVENTIONS

I have several comments regarding the March '03 issue that I would like you to consider:

1. In the article by Thomas Perazella, "Beyond Thiele/Small" (p. 50), the author incorrectly identifies "Mr. David Clark, [as] the inventor of DUMAX" in the caption of Photo 11. Unless the "invention" is the actual word "DUMAX," which Mr. Clark did coin, the credit for pneumatically actuating a loudspeaker in order to measure its nonlinear parameters must go to myself and my team of engineers working in Madison, Wis., in the early '90s. We developed this technique to quantify the nonlinear parameters of the drivers that we were using in our active noise control project.

   This work was published in the AES preprints "Efficient Loudspeaker Linear and Nonlinear Parameter Estimation" presented at the 91st AES Convention in October 1991, and a follow-up paper, "Efficient Loudspeaker Linear and Nonlinear Parameter Estimation: An Extension," presented at the 93rd AES Convention in October 1992. Mr. Clark was fully aware of both these articles and borrowed extensively from them in developing his DUMAX system.
2. In the article by Dick Pierce, "Bass in Small Spaces" (p. 32), I believe that the author is incorrect in asserting that his topic is the subject matter of "A common and pervasive myth . . . ." This could only be true for someone who has not done a literature review of this topic.

   In "The Localized Sound Power Measurement Technique," *Journal* Audio Eng. Soc., Vol. 34, No. 3, March 1986, and its companion publication "The Equalized Sound Power (ESP) Automotive Sound System," Henry Blind and myself discuss virtually everything that Mr. Pierce discusses. We showed how to "utilize this low-frequency loudspeaker and "room" pressurization behavior to effectively extend the response of the system to very low frequencies . . . ," a technique that has been used in OEM automotive systems produced by numerous manufacturers ever since.

   I would also like to point out that this effect relies on a sealed enclosure to reach its theoretical potential, which can never be even remotely valid for any practical enclosure. Humans need to breathe and as such the exchange of air into and out of any habitable enclosure accounts for a substantial leak at DC (static pressure). This makes the effect mute for all but the very smallest enclosures—cars, and even then it is not found to be very effective for the all-too-common SUVs and vans.

In commenting on these two articles I would simply state how disappointed I am with the background knowledge shown by the authors. In both cases these errors could have been avoided by even a cursory literature search. I cannot comprehend how someone can write an article without finding out what others have done and said on the same subject. To come before a thesis defense committee and be caught not having done a proper literature review is paramount to rejection.

I would also like to invite your readers to participate in an Internet study on the Perception of Distortion being conducted by my wife and myself. This can be done by visiting our website www.GedLee.com, which might also be a good place to start a literature review for their next publication.

Dr. Earl R. Geddes
GedLee LLC
Novi, Mich.
www.gedlee.com

Thomas Perazella responds:

Being criticized by a person of the stature of Dr. Geddes certainly gets your attention. Dr. Geddes has contributed greatly to the audio community for many years and has used a sound scientific approach in his work. Personally, I have benefited from his activities on several occasions.

While he lived in Illinois, I was invited to audition the acoustically isolated listening room he had built in the basement of his house. The results were truly amazing. In addition, he was gracious enough to conduct a presentation at my house for the Prairie State Audio Construction Society on his Speak software and even offered members a discount, which was much appreciated.

However, I disagree with Dr. Geddes when he relegates the work of David Clark and his very talented associates to the level of wordsmithing the term DUMAX. Nothing could be further from the truth. The last time I checked Webster, the definition of invention included "a new device or process developed from study and experimentation." From my discussions with Mr. Clark and others involved in the DUMAX project, it is clear that he and his associates did, in fact, follow a course of study and experimentation to produce the product/process known as DUMAX.

It is a very rare invention, indeed, that relies totally upon the work of one individual in isolation. If that were the case, as someone once said early in the twentieth century, there would be no more inventions as everything of value had already been invented. Rather, most inventors—either directly or indirectly—benefit from the work of those before them. Likewise, just because someone invented the internal combustion engine does not mean he also invented the airplane, which relies on that prior work.

The development of DUMAX is certainly a situation that follows the Webster description of an invention. Reading the paper by Mr. Clark that I cited in the article will show that not only did Mr. Clark acknowledge the work of others in this project, including a reference to Dr. Geddes' paper, but even went so far as to issue in section 11 of his paper a special acknowledgment to Dr. Geddes and others that personally helped him.

That being said, there are facets of the DUMAX system that rely on both hardware and software that are unique. The expertise and sweat of the individuals who worked on that project contributed to a unique product/process that is, in fact, a working reality today. In any field, there are always people who have ideas and pave the way for new discoveries. However, that does not make them the inventors of every subsequent product/process that originates from their ideas. The real importance in many cases to the people who ultimately benefit from new ideas is whether or not a practical output from all the ideas together produces a tangible result that makes their lives better. In the case of DUMAX, that is true.

As a contributor to audioXpress, it is my *responsibility to provide the readers—primarily amateurs who design and build their own equipment—with information they can use in their decision-making process as they build such equipment. DUMAX is a product that does currently directly provide them with such a benefit. Looking at advertisements from vendors such as Parts Express and Adire will show drivers that have been DUMAX tested, and the numerical results from those tests. This information is very useful when determining what drivers will form the basis of a new design. I don't recall those vendors listing any drivers as having been Geddes tested.*

Regarding the comment by Dr. Geddes about the lack of background knowledge I have on the subject and the lack of research involved, I can only state that although I am not in the same league as Dr. Geddes when it comes to research, I do have more than a cursory knowledge of this subject that he can verify by speaking to Mr. Clark or Mr. Klippel. I have spent considerable time either in person or through correspondence with both gentlemen on this matter. I have also read the papers of Mr. Clark and Mr. Klippel, including those not listed in my article as references. My article also states that the reader should learn as much as possible from the papers of Clark and Klippel as well as the reference materials of others listed, which would include Dr. Geddes.

If, however, Dr. Geddes is implying that my lack of background knowledge can be inferred by my not knowing of his prior work and thus crediting him as the inventor, he is wrong. I did know of his prior, important contributions, but did not and do not consider him the inventor of DUMAX.

The reference Dr. Geddes makes to a work in audioXpress *passing before the same judgment process as a thesis defense committee shows that he is out of touch with the purpose of the magazine. The last time I checked, Ed Dell had not presented me with any advanced degrees for my work.* audioXpress *does not have the same purpose as the AES* Journal, *nor for that matter other publications whose standards are even stricter than the* Journal. audioXpress *provides an extremely valuable service to the audio amateur by encouraging a free exchange of ideas among persons who generally don't make audio their primary means of support, but rather, very often their primary means of enjoyment. This does not mean that information that is published in the magazine goes unchallenged.*

If Dr. Geddes read the magazine on a regular basis, he would see the number of both challenges and reinforcements to published articles that occur. The fact that his letter and this response have both been published is a tribute to the philosophy that Mr. Dell has had throughout the years. The quote by John Stuart Mill that appears in every issue of the magazine sums it up perfectly. If Dr. Geddes does not see the value that a magazine such as audioXpress *provides, then I would suggest he read the quote by Mill carefully.*

Where I wholeheartedly agree with Dr. Geddes is that the readers of audioXpress *would be well served by participating in the work he is currently doing on his website. I also suggest to Dr. Geddes that both he and the audio community would be better served if he concentrated his efforts on continuing to do the fine work he does and not trying to minimize the also very valuable contributions of others. His work has been of such a high caliber that knee-jerk reactions such as expressed in his letter are not consistent with his capabilities.*

Dick Pierce responds.

I thank Dr. Geddes for his remarks, and welcome his inclusions of the technical references. Certainly, should he feel that these references should have been included in my article, it is a sin of omission on my part.

However, when Dr. Geddes states, regarding my assertion about the topic being a common myth, "this could only be true for someone who has not done a literature review on the topic," I would answer that, while his statement is correct, he is addressing the wrong person. This feeling, that bass is not possible in small enclosures, is a quite common myth among the vast audience in the audio community that have not done the research. I have never encountered this myth amongst those professionals such as Dr. Geddes. I chose to direct the article to a different audience altogether, thus its appearance in the current publication, and not in the Journal *of the Audio Engineering Society.*

That it is a common myth in some circles is evidenced by the number of times you see it proffered as fact on various Usenet groups, Internet audio discussion sites, and in the popular printed literature. You can find any number of examples in the non-professional realm where people have made such statements, also saying things such as "you can't hear bass near a speaker because the wave has not had a chance to develop," and many more such audio "urban legends."

I would ask Dr. Geddes to consider the present forum, where we find possibly the broadest spectrum of readers imaginable—from true experts and luminaries such as Dr. Geddes himself to the most uninitiated neophyte. Allow me to give a recent, concrete example of the problem at hand.

I am sure Dr. Geddes would agree that the principle of linear superposition is sufficiently well-established as to be considered axiomatic, that a system that is capable of supporting two arbitrary continuous functions, one we will designate as F(x), the other as G(x), that system, by the principle of linear superposition, will support the combined function F(x)+ G(x). And I am confident that Dr. Geddes would probably agree that to submit an article on the very foundation of this principle to an esteemed publication such as the Journal *of the Audio Engineering Society would be, at best, inappropriate. I certainly would agree with Dr. Geddes if he were to state that this basic principle should not only be familiar to anyone who has done a literature search, but should be very familiar to such people.*

What, then, would be the appropriate course of action when you are confronted with a question such as the following.

"How do my Martin Logan Aerius speaker panels produce many frequencies simultaneously? In other words, how does the panel vibrate physically at 200 vibrations (cycles) per second and 300 and 1,000, and so on, simultaneously so that the air is set to vibrating at these frequencies so that our ear can hear them?

"If the panel is physically vibrating at 200 cycles to produce this tone, how can it also be vibrating at all other necessary frequencies too?"

This question is seen time and again on any one of the many audio discussion forums on Usenet. Clearly, had this person done a literature search, as Dr. Geddes, many others, and myself have, he would not have asked such a question. Thus, is it appropriate that we do not answer such a question, because the answer is in the literature, and it's there to be found? I counter with the fact that the person asking this question, indeed, is doing a literature search. And, thanks to myself and others who have taken the time from our schedules, he hopefully now understands a little better the answer to his question.

I have been discussing several potential articles with Mr. Dell, including articles about the principles of resonance and the periodic exchange of energy between energy-storing reactances in such systems, why the speed of sound is what it is and other topics, and a number of others directed towards the most basic principles of sound. Because these topics have been covered in great detail in the literature elsewhere (in some cases, dating back to the days of Helmholz and Rayleigh), and while the topics are certainly familiar to one class of potential readers, are they therefore unsuitable for a publication like audioXpress *because they are unfamiliar to "someone who has not done a literature review on the topic?"*

Indeed, it is precisely the group of people "who have not done a literature review on the topic" that I humbly directed this and other such articles. Outside of the circles of the Audio Engineering Society, the Acoustical Society of America, and similar scholarly organizations, there exists a vast audience of those who, by definition, "have not done a literature review on the topic." Among these people—many of them eager to learn—circulates, unfortunately, an active and deeply-rooted tradition of myth, black magic, and flim-flam. You need only review, with frustration, the nonsense in the popular press surrounding the whole topic of sampling and digitization as yet another existence proof of the problem.

Again, I thank Dr. Geddes for his comments, and look forward to his continued participation in providing a solid foundation in physical principles that will help bust these myths and legends and more widely assist in raising the scientific literacy of those not yet privileged to have navigated the literature.

ROTEL UPDATE

I looked at Fig. 1 [RCD 970BX analog output stage] of "Upgrade Rotel's 970BX CD Player," (Feb. '03 *aX*, p. 26) and noticed a couple of weaknesses. Then I saw that you were going to fix that circuit, and looked to see how you would do so.

OK, you put a good mid-value cap across the 100µF. But you didn't do anything about the 4k70 on the + input! I don't know that much about the OPA604AB, so I don't know about its Ib.

But in general, the R at the + input of an op amp, if desired to cancel the op amp's Ib, should not be RF or Rin, but

RF//Rin. In this case, 3.3k//4.7k = 1.93k. A 2k 2% resistor would be fine, and it would make less noise. To cut noise even more, put a 0.001 or 0.01 ceramic disc across the _2k.

Yes we all know that audio freaks *hate* ceramics. But in a simple case like this where there's no DC and no signal across it, 0.01μF ceramic disc would be perfectly fine. Go ahead and tack a 0.01μF disc and a 0.01μF poly to ground. Touch the other end of one or the other cap to the + input of the op amp. Can you hear any difference? I doubt it. Of course not.

[Have you ever considered swapping the AD711 and the OPA604 in and out? Could you really hear any difference? Very doubtful!]

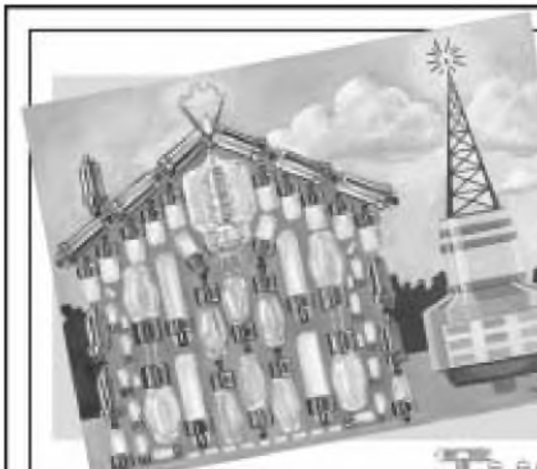
R. A. Pease
National Semiconductor Corporation
Santa Clara, Calif.

P.S. I don't know exactly what load has to be driven by the analog output. But since there is probably neither DC nor any significant AC voltage across your 0.47μF poly cap, it could be replaced with 0.1 or 0.2 or 0.47 or 1.0μF ceramic disc. And I bet you couldn't hear any difference. And, without going to microwave frequencies, you probably couldn't measure it. A ceramic disc will tend to have lower hi-frequency Z (less L) than a 0.42 poly.

So if you just want to make a lower impedance, a ceramic disc might do (infinitesimally) lower Z than the poly. Less resonance at low frequencies (200kHz to 5MHz—maybe a bit more at 50MHz).

Charles Hansen responds:

Thanks for your input. You certainly have me on the 4k70 input bias balance resistor. I did not even take notice of its value! I don't really know why Rotel, who has very talented and practical audio design engineers on staff, even used it. But since they did it should have the value you stated. The TDA1305 DAC is a single ended type that, after the current-voltage converter op amp stage, requires an output coupling because of the DC offset. As far as shunting the balance resistor with a ceramic cap, I don't see any technical reason why not, given the minuscule voltage across it. Others may disagree.



There are some sound technical reasons for not using ceramic caps (other than NPO or CGO types, anyway) in the signal path. The capacitance of general-purpose X7R or Z5U types change with everything: temperature, DC voltage, AC voltage, frequency, dielectric absorption; and this nonlinear capacitance variation introduces distortion. They are also sensitive to shock and vibration. Film caps are just better behaved.

Someone else's critical listening might prove me wrong, but there are certain rules of thumb we all use that save design time and produce known good results. There isn't even a cost incentive to use a GP ceramic. A Xicon 0.1µF 50V Z5V costs $0.26, an X7R Mil CK05 costs $0.39, and the Xicon Polyester of the same value costs $0.24. If you really want to splurge and get a better dielectric, a Xicon Polypropylene costs $0.53.

In the case of the Rotel CD player, the coupling cap has a few volts DC from the DAC across it in addition to the audio signal, and I didn't think a GP ceramic was appropriate there and used the poly. (I had an exchange of e-mails with Bob, and he has agreed with the need for a film coupling cap rather than a ceramic.) Depending on the preamp into which the CD player is fed, the output load could be as low as a couple kΩ. This is why Rotel used that large 100µF coupling cap.

For those readers interested in the investigation of capacitor types as they apply to audio, I recommend the following.

- *In Feb/March of 1980, Walt Jung and Dick Marsh wrote a two-part article for* Audio *Magazine called "Selection of Capacitors for Optimum Performance—Part 1" and "Picking Capacitors—Part 2."*
- *In the 4/85 issue of* The Audio Amateur *(TAA, forerunner to aX), Walt and John Curl developed the "Real-Time Signal Test for Capacitor Quality."*

The choice of OPA604 came about after consulting with some of the people whose judgment I trust. One of the applications engineers at Burr-Brown (now part of TI) suggested the OPA134. The Classé CDP-5 CD player (a Canadian product) had used the OPA134 but recently switched to the OPA604. TI has an app note entitled "Output Spectrum and Post-LPF Design of the PCM1710" that uses the OPA604 as its output stage. Gary Galo recommended the INA134 differential audio line receiver for bipolar DACs and the OPA604 for single-ended DACs.

After all this research and considered opinions of the IC experts, I went with the OPA604. Walt Jung said he had excellent results with the AD743/AD745, but it wouldn't be available in the DIP package in the future.

MARCH MUSTER

As I've been critical of various articles from earlier issues, it's a great pleasure to compliment you on the overall excellence of March's. Dick Pierce's accurate and well-written "Bass in Small Places" is particularly outstanding. I do wonder why Charles Hansen didn't scale the impedances in his filter for the HP 339A down a bit, with NP0 caps being readily obtainable in values as high as 15nF, but this is a minor cavil.

Brad Wood
Bwood@harman.com

TUNING BOXES

This letter is in response to Marc Bacon's excellent article in the February '03 *audioXpress* on optimizing frequency response in ported and closed enclosures (p. 36). My comments have to do with Mr. Bacon's statements in regard to ported enclosures.

Mr. Bacon shows formulae for determining the appropriate size and box tuning frequency for a ported enclosure. He shows a formula for determining the size of a ported box stated as follows:

$$V_B = Q_{TS}^{2.87} \times V_{AS}.$$

He attributes the formula to David Weems, a remarkably creative speaker designer whose work I have followed for as long as I have been building speaker systems, which is almost 40 years. In fact, I have built two of Mr. Weems' projects. One was a distributed ducted port bass reflex enclosure which used a University M12D 12″ driver, and the other was the so-called "Club Sandwich" bass reflex system that used an Electro-Voice 12TRX-B 12″ driver in a big ported box made of layered drywall, insulation board, and plywood over a frame of 1 by 3 lumber. Both were published in *Electronics Illustrated* long before the advent of Thiele and Small, and both were really good, particularly when used with the small amplifiers of the day. Weems made the theory easy enough to understand so that a high school boy like me (then) could understand it. But, I digress.

The formula, though used by Mr. Weems in several books, does not belong to him. In his book, *Advanced Speaker Systems* (Master Publishing, Inc., Richardson, Tex., 1995), Ray Alden correctly credits what I believe to be the same formula to D.B. Keele. Mr. Alden also correctly states the formula. It is properly written as follows:

$$V_B = 15 V_{AS} (Q_{TS})^{2.87}.$$

I believe that the formula Mr. Bacon is citing is the Keele formula because of the unusual use of the 2.87 power in the formula for box volume, and also the use of the .9 power in the formula for the box frequency. If Mr. Bacon is using the formula as stated in the article, he certainly will have a result that is severely affected by small variations in Q_{TS}. He will also get a box that is way too small, about one-fifteenth the size that it should be. If he is using a low Q driver, he will get a box so small that he may not be able to fit the speaker in it. I guarantee that if he is using the formula as stated in his article, his results will be somewhere between really bad and truly godawful. Perhaps this explains why he is getting boomy bass, since he certainly will be using a box that is way too small.

If he is using the correct formula, Mr. Bacon's comments about the result to be obtained from the formula reflect his desire to have a different quality to the bass response than some others might desire. He is, of course, entitled to his preference, and I do not disagree with him totally. I have always preferred a detailed bass response, but I like it loud enough to balance the top end. My suggestion is to leave enough room in the box so that if you don't like the way it sounds when you put it together, you can lengthen the duct and see whether it tightens up the sound to suit you.

Weems always said that it doesn't matter whether you have the exact correct numbers in your alignment if the sound is good. After all, this is why we build these things, so we will get the sound we want. Ultimately, if we are sensitive, and do not deviate too far from the theoretical ideal, we get something that is far more satisfactory than the manufactured product, and the creative and artistic satisfaction of having achieved it ourselves.

There is another use for Mr. Bacon's method, though. When you are required to put a driver in a bass reflex box that is too small for it, Alden uses the following formula to adjust the box frequency to take into account the smaller than optimum box. Alden's formula for adjusting the box frequency to take into account reduced volume is:

$$f_B = \left(\frac{V_{AS}}{V_B}\right)^{0.32} F_S$$

When you use this formula to pick the tuning frequency for the smaller box, you can get a colossal hump in the frequency response at some frequency that will muffle the vocal reproduction. Alden gives formulae for the height and frequency of the peak, but they don't matter here. If the tuning frequency is lowered, as suggested by Bacon, the frequency response flattens out, and the quality of the bass and the lower mids will be much improved.

I just finished a project where I used Alden's Keele numbers to build a ported system to revise a set of 950 in^3 cabinets for a friend. I was using woofers that had Q_{TS} of .38, V_{AS} of 22.23 liters, and f_s of 37Hz, measured in LMS. The "ideal" box for that driver was in the neighborhood of 1300 in^3. The official Alden-Keele alignment for that speaker in the 950 in^3 box gave a box frequency of 39.69Hz.

Modeling the system in LEAP gave a peak of about 2dB at the box frequency, and sounded tubby when I built it. When I lowered the box frequency to 35Hz (not a lot), the peak vanished in LEAP, and the tubbiness went away when I played it. Frankly, I was surprised that I got such a major change with such a slight alteration of the box frequency.

In the old days—before the work of Thiele and Small—you built the box big enough so that it raised the resonant frequency as little as possible. The so-called optimum tuning frequency for a bass reflex cabinet was the cone resonant frequency, or f_s. The theory was that the speaker naturally wanted to have its maximum excursion at that frequency, and that tuning to that frequency would cause the maximum excursion to be reduced. The ideal box volume for a given driver was a volume that gave a port area the same as the cone area at the cone resonant frequency. This gave maximum port radiation at f_s.

Back in the '60s, I built a couple of systems like this, with stiff old 12″ woofers that resonated in the 60Hz area. The speakers I built this way tended to behave in the way Mr. Bacon described, in that they would have a relatively tight low end and would roll off more gradually than speakers adjusted to Thiele/Small parameters.

This suggests that the practical limit for lowering the box frequency is probably not much below f_s. Also, the port area should not be greater than the cone area, since this will unbaffle the speaker. However, you are not likely to be building a box that big or tuning to a frequency that high. Nobody does that any more.

The point is that I verified experimentally what Mr. Bacon is talking about, and he is right. I also think that lowering the box frequency as he suggests can be used to eliminate some of the evils that arise when you are forced by considerations of décor to use a smaller than optimum bass reflex box. Thanks for a really good article.

Joseph A. Zannieri
Norwalk, Ohio

Marc Bacon responds.

Thanks for a good letter and sharp eyes. I proofread the article several times, but missed the "15" in the formula. The correct formula is therefore $V_B = 15Q^{2.87}V_{AS}$, as you stated.

The balance of the article works, just as you've discovered experimentally. Three reasons for lower box tuning than "optimal," other than simply compensating for small boxes, are:

1. *The rolloff begins earlier and is less steep. While this does not give the theoretical bass extension on paper, it does provide for better transients and is a better match to room boundaries in most listening rooms.*
2. *The box is much less sensitive to parameter variations. You can prove this by experimenting or by observing the change in response when you vary the speaker parameters with an "optimally" tuned box versus one tuned per the article's recommendations.*
3. *At high signal levels, drivers tend to behave as though they had a higher Q than their measured small-signal T/S parameters. The lower tuning can accommodate this better than an "optimally" tuned box.*

Tuning to lower frequencies should be done by lengthening the port, rather than reducing the diameter. In this way, nonlinearities and air noises are reduced. If you cannot tune a small box to a low enough frequency without reducing port diameter, try:

1. *A larger box*
2. *A passive radiator*
3. *A different driver*

Thanks again for the encouraging remarks and for finding the error in the formula. ❖

Book Review

Pro Audio Reference

Reviewed by Bill Fitzmaurice

Pro Audio Reference, *Rane Corporation, 10802 47th Avenue West, Mukilteo, WA 98275. $34.95.*

Got an encyclopedia on your bookshelves? How about a dictionary? Probably. How about an electronics reference? You probably have at least two. But chances are you don't have a really complete encyclopedia devoted to modern audio electronics. If you're overdue to rectify that omission, Dennis Bohn's *Pro Audio Reference* is a worthy contender for a spot on your bookshelves.

CONTENTS

I've never seen a more complete nor accurate reference work when it comes to audio. I couldn't think of a term not covered, and found quite a few I was unfamiliar with, not to mention a few terms not commonly associated with either audio or electronics, such as "bumbershoot" and "orange." Author Bohn regularly deviates from the path you might expect of a reference book with welcome injections of humor. This is a much more interesting read than Funk and Wagnall's.

But this book is far more than an audio/electronics encyclopedia. In fact, that portion of the book takes up only about a third of its pages. The remainder is one of the best works I've yet come across as an audio professional's handbook. You see, the publisher of this book, Rane, is also one of the most respected manufacturers of high-end pro audio signal processing gear, and author Bohn is one of Rane's more illustrious engineers. The bulk of the book has chapters written by Bohn and other Rane engineers that cover pretty much every aspect of pro audio signal processing.

Anyone considering a career in pro sound would do well to read this book in its entirety. Unfortunately, most sound professionals—be they musicians, roadies, sound engineers, or record producers—have very little formal education in their respective fields. That's not unusual, considering that pro sound isn't a subject found in the average high school or even most colleges. Audio pros generally learn by doing, because that's their only option. Reading this work is the equivalent of a full semester at Berklee.

Starting with the fundamentals of signal processing, Bohn and his cohorts move on through chapters on audio specs, mike preamps, home cinema, impedance matching, sound system level control, line drivers, and constant-voltage distribution. Three chapters are devoted to the complexities of grounding. Most of the tools of signal processing—equalizers, crossovers, and dynamics processors—are examined in great depth. Rane processing gear is used for examples, as you would expect, but the subjects are so well explained that the knowledge gained is useful for working with other brands as well. It might take years of on-the-job training for you to absorb the information in this volume.

As good as the book is, there's more. The computer CD contains most of the information in the book, and loaded into a laptop, it proves most useful in the field. Additionally, the CD has links to websites for delving even more deeply into every subject. In the long run this could prove more valuable than the book itself, for while the book will not change, technology will. If ten years down the road new technology has rendered much of the printed matter obsolete, the CD's web links will afford you access to whatever the state of the art may have become.

AUDIENCE

You may question whether this work is worthwhile for the non-professional. I believe it is. If your audio system is any more complicated than a simple plug-and-play, a DSP may well be a worthwhile addition—if you know what it can do for you and how to operate it.

If you're into home theater—or want to be—this book answers all your questions. Even if you just like to watch TV through a regular hi-fi, you may find that a compressor/limiter is a worthwhile addition to control the level differences between the programs and the commercials. In short, if you're into audio to the extent that you're reading this review, you would probably find this book a welcome addition to your library. I would. ❖

Showcase

Building Rozenblit's 80W OTL

PHOTO 1: The author's version of the Rozenblit 80W transformerless tube output.

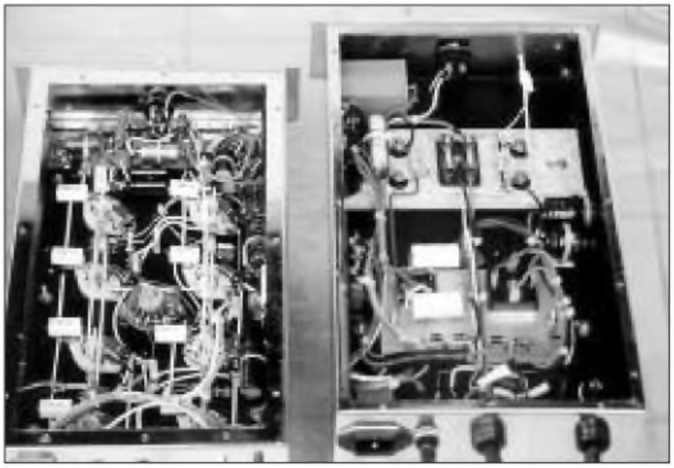


PHOTO 2: A look inside the amp.

I built this amp according to Mr. Rozenblit's design published in his book *Audio Reality* (p. 81, available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, FAX 603-924-9467, e-mail custserv@audioXpress.com). I have made some modifications to increase its performance: I added a choke (20H 30mA) and some additional filter caps for the gain stage and phase inverter stage power supply, which provided very obvious benefit—the hum became almost inaudible. I also increased the capacitance of the power stage power supply filter cap, using two 400V/3900µF "big ponds" to store all the energy needed for the 16 EL504s. This results in a very stable power supply even during signal transients.

I reduced the NFB by deleting R11, which I found was more like "tube sound" than I enjoy. By adding a variable resistor (100k) in each channel for level control, you can directly connect this amp to a CD player without a preamp.

I wound a big single transformer, which included filaments, bias, power stage, and gain stage power source. This transformer and all the filter's parts (cap and choke) are built in one chassis, while the left and right amplifier sections are fixed into two separate chassis.

I used copper composite metal for the chassis, then gold-plated them to form beautiful boxes. Even though all the voltages measured might not be exactly equal to the figure shown in Rozenblit's original diagram, the important thing is to keep "relative voltages" unchanged; i.e., pin 2 and pin 3 of V2 have a 15V potential difference, pin 1 and 3 of V3 have 100V potential difference, and so on.

This amp is not easy to build, but for those who have become bored (DIYers always like to try new things) with ordinary push-pull or single-ended tube amp sound, it is worthwhile to try this new patented design OTL amp. ❖

Ignatius Chen
Bandung, Indonesia

Classifieds

VENDORS

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

Circuit Boards for projects, prototypes, or small quantities. From your artwork. Low Rates. No minimum. **Atlas Circuits** 704-735-3943. **E-Mail:** Atlas@conninc.com

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**.

THORENS

The New Generation TD800 Series

See all the new products at www.thorens.com

Distributed by:
Trian Electronics 608-850-3600 or www.triancorp.com

BILLINGTON EXPORT LIMITED

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

Precision Port™ *Be Different - Sound Better*

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

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

Integrated amp, ALL-FET lineamp
www.borbelyaudio.com
Borbely Audio kits in USA:
www.audiokits.com
Borbely Audio kits in Japan:
http://homepage3.nifty.com/sk-audio/

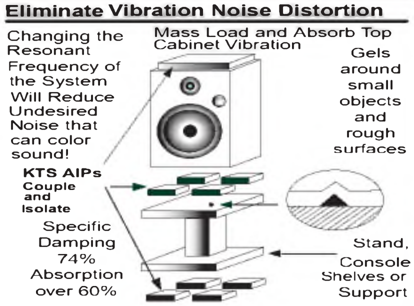
www.classifiedaudio.com
High end audio.
Buy it. Sell it. Revel in it.

CERAMIC CABINET loudspeakers. A sonic revelation! **Visit: www.faradaysound.co.uk** Comprehensive DIY information booklet $15.

Low cost PCB layout and electronics design service **http://inyxsolutions.netfirms.com/home.htm**

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

Eliminate Vibration Noise Distortion



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

FINALLY AFFORDABLE

$10 SET OF 4 PLUS FREE SHIPPING *Continental USA*

SILVER PLATED
SPADE LUGS
All direct plating, no nickel flash. 4/9's pure copper. Designed for 1/4"-5/16" posts and up to 10g wire. Can be crimped and soldered. Also available in gold and bare.
Email orders only please: luminous@cavtel.net

LUMINOUS AUDIO TECHNOLOGY
High-end cables since 92'
www.luminousaudio.com

Creative Sound Solutions
www.creativesound.ca

Piano black 32-liter sub cabinet.
Pre-order price
US $150
(plus shipping).
See website for details.

Repairs and upgrades for audio equipment. Let longtime *aX* contributor and *VTV* senior editor Eric Barbour make your system sound better.
METASONIX, PMB 109, 881 11th St., Lakeport, CA 95453, synth@metasonix.com, (707) 263-5343.

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

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

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

WWW.TESTEQUIPMENTDEPOT.COM
Large supply of new & pre-Owned electronic test equipment—oscilloscopes, power supplies, spectrum analyzers, DMMs, network analyzers, and lots more. Call us at **800-275-6261**

tubes?
www.tubes.it

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

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

gear racks, media drawers and more
Factory direct since 1984
Free brochure (mention audioXpress)
Per Madsen Design (800) 821-4883
www.rackittm.com

KTS www.kleintechsys.com

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

Yard Sale

FOR SALE

New Goldpoint stereo, 10k, 24-position, stepped, ladder-type attenuator. Only two resistors active for each switch position results in lowest possible resistor noise, and best matched channel-to-channel signal level. Considered the best. $199. roy@cottonwoodcreek.com.

Power amps and heatsinks. Two DIY amps for parts or repair—one Williamson 40/40 with regulated power supply, two Hammond 167p60 60VCT 300VA transformers, $100. One 80 WPC with two Hammond 167N70 70VCT 280VA, $100. Large box of assorted heatsinks, medium to large, $50. D. Elderton, 1511 Philip Ave., North Vancouver BC, Canada V7B 2V7, 604-988-6666, elderton@telus.net.

DIY PARTS SALE. Audio grade caps, resistors, chassis, and more! 50% off typical retail. Complete inventory and prices at: http://www.laventure.net/parts4sale/audio.htm.

Pair of Parasound HCA-3500 high quality, John Curl designed 100 lb monster amplifiers. 500W per channel, three years old, like new condition, remaining warranty. $1000 each plus shipping. Several Hafler DH220 amplifier modules, heatsinks, circuit boards, and SJ49/SK134 MOSFET pairs. Call or e-mail for details. Joerg, 561-881-8112, jzschirnt@aol.com.

Pair of AR3as. Woofers (reconed) $125/pair. Cabinets (fair shape, grilles new, no badge) $125/pair. Everything $175. Mids, tweets, xover not included. Prices do not include packing (Mailboxes, etc.) and shipping. Located in northern N.J. Les Winter, joanleswinter@aol.com.

WANTED

Kit construction manual of reprint for HK Citation 12 Power Amp. Will pay all costs. Henry S. Wypa, 3710 N. Beechwood Road, Rose City, MI 48654-9796, 989-685-2429.

Cadillac 16 GM drivers needed for the project in *aX* Jan. 2002. Robert J. Howard, e-mail docrob@starband.net or call 334-566-1999 or 334-372-2909 and leave message.

We Want Your Feedback!

audioXpress encourages reader feedback in the form of letters, queries, and comments.
Send correspondence to:

audioXPRESS

Editorial Dept.
PO Box 876
Peterborough, NH 03458-0876
Phone: (603) 924-9464
Fax: (603) 924-9467

Website: **www.audioXpress.com**
E-mail: editorial@audioXpress.com

audioXpress reserves the right to edit letters for length and clarity. Letters should be brief and to the point.

AD INDEX

ADVERTISER	PAGE
ACO Pacific Inc	44
Adire Audio	52
Alpha Electronics	36
All Electronics	72
Antique Radio Classified	55
Audience	40
Audio Amateur Corp.	
Cliolite	58
Matter of Coincidence	30
Classifieds	70
Old Colony Catalog on-line	32
Testing Loudspeakers	44
World Tube Directory advertising opportunities	54
Audio Consulting	59
Audio Electronic Supply	23
Audio Transformers	65
audioasylum.com	11
Audiogon.com	14,18,37,59,71
Audiomatica	31
Avel Lindberg Inc	47
Classified Audio-Video	47
DH Labs Audio Cables	CV2
Diamond Groove	40
Dynasonic Ltd	42
E-Speakers	53
EIFL	1
Electus Distribution Pty Ltd	33
Exact Power	29
Grennan Audio	42
Hagerman Technology	49
Hammond Manufacturing	17
Harris Technologies	27
Hi Fi Do Inc	39
House of Tubes	65
JENA Labs	51
K&K Audio	43
KAB Electro-Acoustics	59
Kimber Kable/WBT-USA	4, 5
Langrex Supplies	21
LC Audio Technology ApS	9
Linear Integrated Systems	61
Madisound Speakers	28
Marchand Electronics	36
McFeely's	67
Mouser Electronics	63
Origin Live	41
Parts Connexion	13
Parts Express Int'l., Inc.	CV4
Pass Laboratories	69
Plitron Manufacturing	37
Selectronics	43
Sencore	33
Solen, Inc.	CV3
Speaker City USA	45
Swans Speakers	25
Test Equipment Depot	55
The Last Factory	68
Thetubestore.com	57
Usher Audio	7
Vacuum State of the Art Conference	15
Velleman	63
Venus HiFi	61
WBT-USA/Kimber Kable	4, 5
Welborne Labs	19
World Audio Design	35
Zalytron Industries	23
CLASSIFIEDS	
American Science & Surplus	70
Aspen Amplifiers	70
Atlas Circuits	70
Audio Electronic Supply	70
Billington Exports	70
Borbely Audio.	70
Classified Audio-Video	70
Creative Sound Solutions	70
Faraday Sound	70
Inyx Solutions	70
Klein Tech Systems	70
Luminous Technology	70
London Power	70
Metasonix	70
Per Madsen Design	70
Precision Sound Products	70
Rockler Woodworking	70
Sonic Craft	70
TDL Technology	70
Test Equipment Depot	70
Thorens of America	70
Vintage Hi-Fi	70
Welborne Labs	70

New Chips on the Block

By Charles Hansen

TI 1394a-2000 FireWire Controller and TSB43CA43A iceLynx

Pioneer Corporation has chosen TI's integrated (FireWire) link layer controller for the world's first 1394-enabled DVD player and audio/video receiver. The Pioneer Elite DV-47Ai DVD-Audio and Super Audio CD player provides the ability to pass high-quality digitally-recorded audio information via the 1394 interface, enabling one simple connection to other audio/video products. The Pioneer Elite VSX-49TXi audio/video receiver also offers the 1394 interface.

The 1394a-2000 is aimed at providing the highest quality of sound in consumer audio/visual (A/V) equipment and reduces the number of cables needed for component interconnect from six to one.

In a second-generation device from TI's iceLynx family, the TSB43CA43A (also known as iceLynx Micro) transfers high sample-rate digital audio data between A/V receivers and DVD-audio and SACD players. ❖

Texas Instruments Semiconductor Group, 12500 TI Blvd., Dallas, TX 75243, (800) 336-5236, www.ti.com.

Analog Devices AD1896

The AD1896 is a 24-bit, high-performance, single-chip, second-generation asynchronous sample rate converter. Based upon Analog Devices' experience with its first AD1890 asynchronous sample rate converter, the AD1896 offers improved performance and additional features. These include a minimum THD+N of 120dB for all input frequencies and sample rates, 139dB dynamic range, 192kHz sampling frequencies for both input and output sample rates, improved jitter rejection, and 1:8 up-sampling and 7.75:1 down-sampling ratios. Additional features include more serial formats, TDM, daisy chain, a bypass mode, better interfacing to digital signal processors, and a phase matching mode.

The AD1896 is the first sample rate converter to exceed 130dB signal-to-noise ratio and the first device to support the 192kHz specification of the latest Digital Versatile Disc (DVD) standards. The asynchronous conversion ratios—1:8 upsampling and 7.75:1 downsampling—enable it to convert the sample rates for virtually any application, including the matching of sample rates between a 192kHz DVD-A and a 44.1kHz CD, and vice versa. These features markedly increase the chip's versatility without requiring any programming, and permit multiple asynchronous sample-rate converters to be used in conjunction with a digital-signal-processor (DSP) device for enhanced performance, and for producing digital audio effects.

The AD1896 is designed for use in home theater systems, studio digital mixers, automotive audio systems, DVD players, set-top boxes, digital audio effects processors, studio-to-transmitter links, digital audio broadcast equipment, as well as digital tape variable speed applications.

The AD1896 is available in a 28-pin SSOP package, and is priced at $11.30 US in 1000-piece quantities. This device requires a single 3.3V supply but can be made 5V-input tolerant by connecting its VDD_IO supply pin to a 5V source. ❖

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