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Speaker Builder®

THE LOUDSPEAKER JOURNAL

20th YEAR

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Joe D'Appolito

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Phil Abbate

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John Mattern

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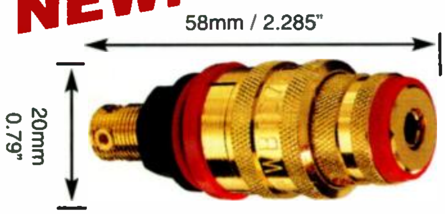
Don't be stupid!

Highest quality sound from the front of your speaker project requires high quality signal to the back of your speakers. Be smarter, use WBT!

Binding Posts (crimp technique)

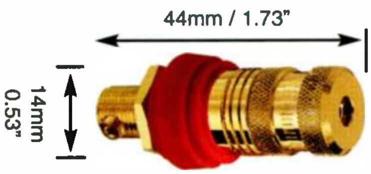
NEW!



WBT-0701 Topline — each $50.00
WBT-0701-KIT 2 posts, 1 Torx™ key, boxed — kit $104.00

If you have this much room (20mm) for performance...buy it!

400 amperes continuous current. The ultimate pole terminal for demanding applications. Robust design with special elements and features. Patented clamping mechanism for high compression connection of spade connectors and crimped wire ends up to 16mm² (6AWG). Patented collet chucking device for banana plugs. New milled groove on both sides of compression plates for high-strength connection of spades. Includes built-in insulation. Color code: red or white. Can also be soldered.



WBT-0730 Topline — each $30.00
WBT-0730-KIT 4 posts, 2 spacer blocks, 1 Torx™ key, boxed — kit $126.00

200 amperes continuous current. Patented clamping mechanism for high compression connection of spade connectors and crimped wire ends up to 10mm² (8AWG). Accepts 4mm banana. Color code: red or white.

Safety Line

WBT-0735 Topline — each $35.00

Highest performance, safety, quality. A bit more and worth it!

Fully insulated safety version of WBT-0730.

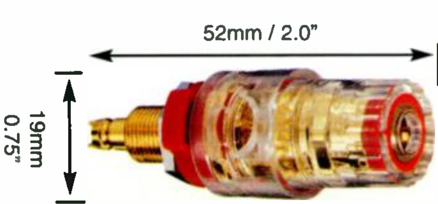
WBT Crimp sleeves	Conductor size *mm²*	*AWG*	*each*
WBT-0430	0.5	#20	$0.50
WBT-0431	0.75	#19	$0.50
WBT-0432	1.0	#17	$0.50
WBT-0433	1.5	#16	$0.50
WBT-0434	2.5	#14	$0.50
WBT-0435	4.0	#12	$0.75
WBT-0436	6.0	#10	$0.75
WBT-0437	10.0	#8	$0.75
WBT-0438	16.0	#6	$1.00

WBT Crimp sleeves with insulated collar	Conductor size *mm²*	*AWG*	*each*
WBT-0441	1.5	#16	$1.00
WBT-0442	2.5	#14	$1.00
WBT-0443	4.0	#12	$1.00
WBT-0444	6.0	#10	$1.50
WBT-0445	10.0	#8	$1.50
WBT-0446	16.0	#6	$2.00

Binding Posts (solder technique)

WBT-0744 Midline — each $17.50

Single pole terminal suitable for 200 amperes continuous current. Patented clamping mechanism, for high compression connection of spade connectors and crimped wire ends up to 6mm² (10AWG). Accepts 4mm banana plug. Color code: red, white or black



Safety Line

WBT-0745 Midline — each $20.00

Very popular for upgrades!

Fully insulated safety version of WBT-0744.

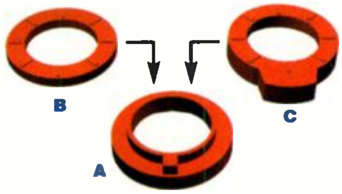
Part A Insulating shoulder washer, fits to the body of the binding post, and can be positioned at 90° increments. The "key", which is visible, fits in a corresponding keyed hole in the panel.

Part B Insulating lock washer is placed on the inside of the panel. The binding post nut tightens directly on this washer completing the assembly.

Part C Insulating cup washer is placed on the inside of the panel. This washer has a cavity that fits over the shoulder and "key" of part A. The binding post nut tightens directly on this washer completing the assembly.

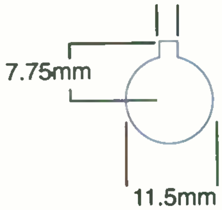
A + B	for panel thickness from 2 - 6mm
A + C	for panel thickness from 0.1 - 3mm

All binding posts are priced with washer set A + B.
Set A + C may be substituted for A + B at time of order.
May be deleted at time of order if not required.



Available in the colors white, red or black

Mounting hole dimensions



WBT-0721 PCB adapter block

Fastens to circuit board with included bolt and lock washer. Mounts horizontally or vertically. Torx™ screw fastens jumper wire terminated with crimp sleeve. Other end of jumper wire connects to binding post.

WBT-0799 safety stick

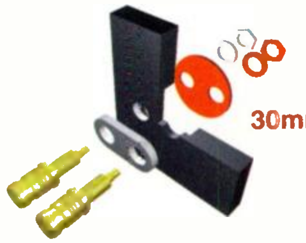
Included with all **Safety Line** binding posts.

Specially shaped plastic peg, inserts in banana plug hole in binding post, required to accomplish CE safety conformity.

May be deleted at time of order if not required.

WBT-0716 thick panel mounting kit

Kit includes a gold-plated brass exterior bracket(1), and a plastic interior bracket(2).



30mm spacing

WBT spacer block (19mm spacing)

Fits all binding posts except WBT-0701.
3/4" plastic insulating spacer block.
Color code: red or white

WBT-0717 holding tool *(not illustrated)*

Use when mounting binding posts in the WBT-0716. Snaps into the two posts, holds them in alignment and prevents the posts from rotating while tightening the nuts.

WBT-0403 Crimping pliers

Crimping pliers with special multiple spike jaw. The only tool designed to crimp all WBT crimp sleeves. Gold-plating protects against corrosion. Comfortable, plastic-coated grip.

Associated parts

WBT-0488	Single Torx™ key (T6)	$4.00
WBT-0716	Thick panel mounting kit	$16.00
WBT-0717	Holding tool	$12.00
WBT-0721	PCB adapter block (works w/crimped cable ends)	$8.00
WBT-0799	Safety stick	$0.50
Part A + B	2-6mm panel	$1.50
Part A + C	0.1-3mm panel	$1.50
WBT - spacer block	two pole spacer block	$2.00
WBT-0403	Crimping pliers.	$90.00

WBT-0488 Torx™ key

T-6 size fits all WBT parts.

WBT silver solder

Highest quality solder with 4% fine silver content. The halogen-free flux is gentle on precious metal surfaces, the melting temperature (178/180°C) is extremely low.

WBT-0800	small spool	0.9mm Dia.	Weight 42 grams	$14.00 each
WBT-0820	medium spool	0.8mm Dia.	Weight 250 grams	$41.00 each
WBT-0840	large spool	1.2mm Dia.	Weight 500 grams	$74.00 each

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The art of connection

In 1985, WBT developed the first RCA type plug machined from a single piece of metal, it also featured an adjustable locking mechanism. Since 1985 WBT has continued to refine the art and science of connection. From a single idea has developed an impressive array of audio connectors. Today, WBT is known worldwide for the best connectors in the industry.

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All products made in Germany! Dealer inquiries welcome! OEM quantity prices available!

12 page color catalog available, including tools, binding posts, RCA type connectors and more.

Reader Service #65

Editorial

PERSONAL ADS ARE being reinstated in three of our periodicals: *Glass Audio*, *Audio Electronics*, and *Speaker Builder*. They will appear under the heading *Yard Sale*. And they are free to subscribers for personal purposes.

We discontinued free classifieds three years ago after much agonizing and soul-searching. I think readers should know why we changed our policy.

As usual, a few very irresponsible and thoughtless readers spoiled the feature for everyone. The submissions were casual, to assign the kindest adjective possible to them. They were often thinly disguised business ads, illegible, or badly spelled—especially the trade names.

Eventually everyone on the staff dreaded dealing with personal classifieds. They required not only patience, but often research to try to figure out what the reader was writing about.

Further, many readers paid no attention to our requests that we could not take such ads over the phone, or that we could not tell callers just when their ad would appear. Readers called regularly to ask us to make additions or deletions to their mailed copy.

The speculations of a few readers about our motives in making all classifieds paid were unwelcome. Those who made the suggestion that we are a bunch of money-grubbers were certainly speculating ahead of their data. There are always a few who delight, it seems, in attributing the worst motives possible to the actions of editors or publishers.

What was especially sad about the situation is that for many, many years readers of our flagship *Audio Electronics* sent along their ads in good form and with minimum fuss. These were a great service to other readers, and an excellent way to trade items with other readers for money or barter. Such ads were good reading and a very useful feature of the magazine.

We are ready once again to try to provide this service to all subscribers to our avocational magazines, hoping for minimum levels of headache. Our list of requirements may seem stiff and demanding. We make no apologies for the details, which are born of years of frustration and hard experience. Every game has rules, and ours have been set to make the game possible for everyone.

I hope all of you who have surplus items, old gear, and even special buys of parts you want to share with other readers, will make good use of this reinstated service. We are regarding this as an *editorial* feature, which will not go through our advertising department. If you have business interests you wish to promote through classifieds, those are still available at stated rates through the advertising department.

In the interest of the feasibility of a valuable service, I ask every reader to take full notice of the stated conditions on this page. If you are submitting an ad, and I hope you will, please have the courtesy to refer to these guidelines before you mail, fax or e-mail your text. These magazines are built on service to the reader. I hope you will help us make this service a rousing success for us all.—E.T.D.

FREE PERSONAL ADS ARE BACK!

To publish your free personal ad in *Speaker Builder*, simply follow these guidelines:

1. For subscribers only. Include your account number with each personal-ad submission.
2. This service is for subscribers to sell or find audio-related personal equipment or supplies. Submissions not related to audio will be discarded. Businesses, organizations, and non-subscribers should contact the advertising department to place their paid ads in the classified advertising section.
3. All personal-ad submissions must be printed out via typewriter or computer. Illegible or questionable submissions will be discarded.
4. We will not be responsible for changing obvious mistakes or misspellings or other errors contained in ads.
5. We will not handle any submissions over the phone.
6. Please do not call to verify acceptance, or inquire about the status, of your submission. We cannot personally acknowledge receipt of submissions.
7. It is entirely up to the magazine's discretion as to when your free ad will appear in ***Speaker Builder.***
8. Each ad submission will be used one time only. It will be discarded after publication.
9. Maximum 50 words (no accompanying diagrams or illustrations or logos will be used). Submissions over 50 words will be discarded. A word is any collection of letters or numbers surrounded by spaces.
10. Each submission must be clearly addressed to "Yard Sale" and the name of the magazine.
11. Submit your ad to Speaker Builder, Yard Sale, PO Box 876, Peterborough, NH 03458. Or send by Fax to 603-924-9467 (Please be advised that smudged, illegible faxes will be discarded.) Or, by E-mail to editorial@audioXpress.com
12. Noncompliance with any of these guidelines will result in your free personal-ad submission being refused for publication.

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Free the artist in you with state-of-the-ART software from HARRISTECH.

Design speaker boxes with...

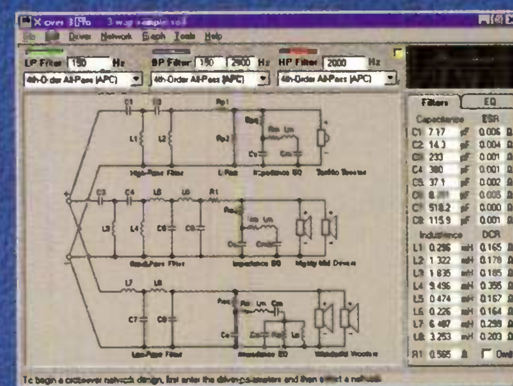
BassBox Pro

Visit us on the web at www.ht-audio.com

Our website includes detailed information about BassBox Pro, X•over Pro and other audio-related stuff. We also provide FAQs (frequently asked questions) and technical support topics. Licensed users can download free updates. And we provide links to the known websites of driver manufacturers.

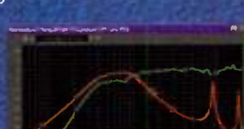
Design crossovers with...

X•over Pro



- BassBox Pro helps you quickly design many different types and shapes of speaker boxes. With its Design Wizard, advanced easy-to-use layout and on-line help it is a joy to use. As many as ten designs can be viewed at the same time.
- Includes a **MONSTER**-size driver database with the parameters of **thousands** of drivers.
- Handles many box types, including closed, vented, vented with active filter (B6), bandpass and passive radiator.

- Accepts designs with multiple drivers, including isobaric designs. Both the mechanical and electrical configuration can be controlled.
- Performance can be evaluated with 9 graphs, including 2 amplitude response graphs, max acoustic power, max electric power, cone displacement, vent velocity, impedance, phase & group delay

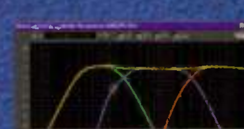


- Acoustic data can be imported for drivers and listening spaces. Both automotive and architectural environments are supported.
- The response of a passive network can be entered or imported from X•over Pro and used to display the system response.
- Includes both driver and passive radiator test procedures.
- Create custom printouts with box drawings & parts lists
- Includes a beautiful, well illustrated 204 page manual. Available on CD-ROM or diskettes.

BassBox Pro: $129.00* *U.S. dollars plus shipping and handling.

Shipping & handling: USA = $7; Canada = $9; All others = $23.

- X•over Pro helps you quickly design passive crossover networks: low-pass, band-pass and high-pass filters, impedance equalization networks, notch filters and L-pads. Its straight-forward layout and on-line help make it very easy to use.
- Includes a **MONSTER**-size driver database with the parameters of **thousands** of drivers, including both open back drivers (woofers) and sealed back drivers (tweeters). Compatible with BassBox Pro.
- Design 2-way and 3-way passive crossovers with a parallel topology. Each filter can be a different type and have a different slope (1st–4th orders).
- The values of individual components can be manually changed and the effects viewed in the graphs.
- Impedance equalization is available for the voice coil inductive reactance and the driver/box resonance peaks.
- Performance can be evaluated with 4 graphs, including amplitude response, impedance, phase & group delay. The net response can be displayed in both the amplitude and impedance graphs.



- The equivalent series resistance (ESR) of capacitors and DC resistance (DCR) of inductors can be estimated or entered.
- Includes additional helpful tools like a Parallel-Series Value Calculator, Color Value Decoder and Notch Filter Designer. The Notch Filter Designer can calculate two types of series notch filters and one type of parallel notch filter.
- Create custom printouts with parts lists & schematics.
- Includes a beautiful, well illustrated 169 page manual. Available on CD-ROM or diskettes.

X•over Pro: $99.00* *U.S. dollars plus shipping and handling. See details under BassBox Pro opposite.

X•over Pro is only $70.00 when purchased with BassBox Pro.

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Reader Service #3

About This Issue

From survey results, we have determined that many of our readers are interested in learning more about horns. For those, we offer **Dick Crawford**'s Dome Horn (p. 10), which is a good project for first-timers to tackle. It's inexpensive and easy, and the result is a speaker–combining the best of dome tweeters with the advantages of horns–with great uniform directionality.

Can't live with factory sound in your new car? Don't be afraid to fix your auto's sound system to suit your tastes. **Bill Fitzmaurice** shows you some modifications for a sub installation you might consider ("The Sensible Sub," p. 14) that still leaves plenty of room in your car for precious cargo.

Sooner or later, you'll come to embrace computers as tools for audio measurements. **Tom Perazella** has, and now he has devised this clever fix that provides protection for your sound boards when using the LAUD Measurement System (p. 16).

Sometimes good things can come out of tragedy, as **Philip E. Abbate** relates with his latest system update ("Afterburner for Aftershock," p. 26). The author's simple crossover project results in a happy home for a home theater setup.

From the design plans and construction details of **John Mattern**'s article ("Another Look at TL Design," p. 28), you can build a stereo pair of transmission line speakers that promise pleasing results for home audio use.

John Badalamenti shows off his skill in assembling a three-way system that looks and, he claims, sounds great ("Tools, Tips & Techniques," p. 58).

For a glimpse at the shape of things to come in speakers, see **Andrew Ware**'s stunning design in this issue ("Showcase," p. 62).

Speaker Builder

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

JOHN STUART MILL

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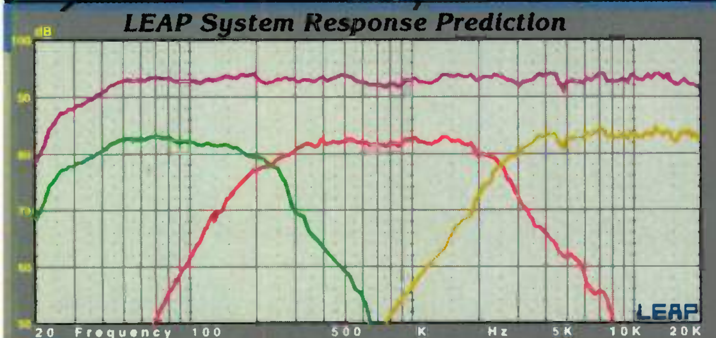
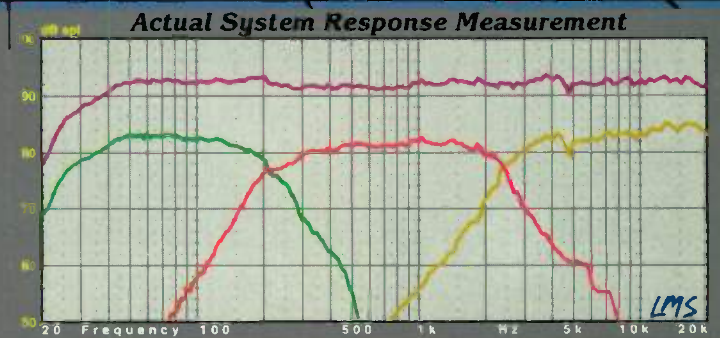
New! LEAP Version 4.5

LEAP

Precision Development Tools for Precision Loudspeaker Designs





The Art and Science... of loudspeaker system development today has become more complex than ever before. Competition is tough, and to compete each design must perform to the best of its ability, and make the most out of every dollar's worth of transducer cost. The simple approach of choosing a combination of seemingly appropriate transducers coupled with ordinary networks and filters, has given way to a painstaking process of meticulously blending selected transducers in combination with carefully devised and matched crossover designs.

■ Seminars/Workshops available, call for details.

LEAP (Loudspeaker Enclosure Analysis Program) is a complete full range analysis package which provides virtually *all* of the tools necessary to develop precision loudspeaker systems, for today's demanding audio markets. Whether your applications are consumer audio, car stereo, professional audio, or custom esoteric marvels, LEAP provides the power, flexibility, and accuracy to investigate every possible design permutation. The open architecture and broad spectrum of features provided will dramatically reduce your development time, while improving the quality of the final result... and demonstrates why LEAP has become the #1 choice of professional loudspeaker designers world-wide!

Advanced Transducer/Enclosure Simulations

- ✔ Sealed,Vented, Bandpass, PR Simulations with multiple speaker/ port capability.
- ✔ Large signal analysis of TempVC, and Non-Linear BL/ Ports/ Compliance.
- ✔ Acoustic Parallel or Acoustic Series (Isobaric) Driver Mounting.
- ✔ Port Standing Wave resonance modeling.
- ✔ Frequency Dependent Revc and Levc modeling.
- ✔ Library storage of 36 transducer parameters, and over 24 enclosure parameters.
- ✔ Generic transducer modeling of electro-dynamic, ribbon, and piezo devices.

Advanced System Analysis Features

- ✔ Use simulation or imported actual measured SPL/Z data.
- ✔ 5-Way crossover system modeling, and more.
- ✔ Time offset between transducers.
- ✔ Active or Passive based crossovers.
- ✔ Hilbert-Bode transform for deriving phase.
- ✔ 22 Passive components per xover section.
- ✔ 16 Active filter blocks per xover section.
- ✔ Passive Network Optimizer for single/system response.
- ✔ Active Filter Optimizer for single/system response.
- ✔ Frequency ranges from 1Hz to 100kHz.

Extensive Documentation

The two volume manual set comprises almost 1,000 pages of documentation which thoroughly covers the operation of the program- and provides numerous examples of how to maximize your use and understanding of the program's many features. The Reference Manual describes all graphs, menus, commands, and their operation. This manual explains the unique and special non-linear speaker and port models, as well as proper use of the optimizers, importing data, and the many other utilities. The Application Manual provides many exciting examples showing how to use the powerful features of the system in a combined manner to perform both simple and complex design tasks. Both novice and experienced users alike will find this information invaluable for exploiting the full power of the system. Additional information is also provided on loudspeaker measurements, design tips, filter calculations, and complete crossover system development for both passive and active based systems.

✔ 502 Page Reference Manual ✔ 436 Page Application Manual

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When you wish to produce a hardcopy output of your finished designs and graphical data, LEAP supports a large number of printer standards, and even supports numerous desktop publishing graphic formats in both black & white and color! Portrait/Landscape orientations in any custom size and aspect ratio are user controllable.

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Reader Service #36

Speaker Builder

THE LOUDSPEAKER JOURNAL

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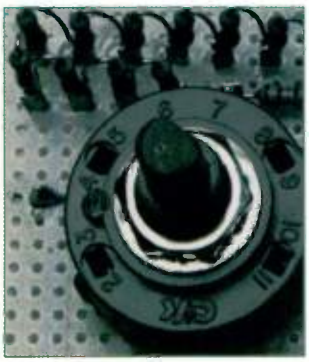

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This article describes a simple, inexpensive way to make horns—for example, a short horn suitable for use with many 1″ or 1⅛″ dome tweeters. The purpose of this horn is to control the directionality of the tweeter to make it similar to that of a 6.5″ woofer.

Dome Horn

By Dick Crawford

Horns have been around a long time. Some of the earliest acoustic phonographs used attractive petal-shaped horns. Their advantage lies in providing better acoustic coupling from the driver (woofer or tweeter) to the air, thus increasing the acoustic efficiency. They also help in controlling the directional characteristics of the sound. On the other hand, they can degrade the sound by having poor transient response (ringing) or a jagged frequency response (peaking). Horns are now used less for hi-fi because the need for efficiency has decreased with the advent of 100W amplifiers.

Horns still, however, have a number of advantages. One is that the increased efficiency they provide means that the tweeters require less power, thereby making them more reliable. Also, directional control has a great deal to do with ambience, which depends largely upon reflected sound bouncing off the walls, floors, ceiling, furniture, and whatever.

The ambience of a recording depends upon the recording venue and the microphone setup. Loudspeakers add it according to their directionality and the acoustics of their environment. Speakers with the same on-axis frequency response can sound differently in the same room, depending upon how much they add. The greater the ratio of reflected sound to direct sound, the more ambience.

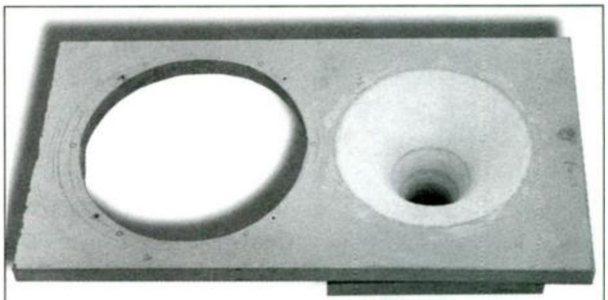


PHOTO 1: Front piece with horn opening and hole for woofer.

Loudspeakers are designed to accommodate listeners' tastes. Planar loudspeakers, both magnetic and electrostatic, tend to be directional, providing a small region (or sweetspot) of optimum listening, fewer reflections, and less ambience. Horn speakers are generally moderately directional, with a medium sweetspot, and moderate ambience. Dome speakers tend to be least directional, giving almost uniform sound over a large region, and a greater amount of reflection and ambience.

DIRECTIONALITY CHANGES

One problem with speakers using dome tweeters is that the directionality changes in the crossover region, going from narrower in the woofer to wider in the tweeter. This is because the woofer is larger than the tweeter, and a larger radiator tends to have more directionality, particularly at higher frequencies. The change in directionality causes the ratio of direct sound to reflected sound to change, thereby altering somewhat the character of the sound, particularly in the crossover region.

This article describes a horn that narrows the directionality of a dome tweeter so that it is about the same as the woofer in the crossover region. The horn achieves this because the flare at its mouth is about the same shape as the woofer cone (*Fig. 1*). The result is a speaker with a more uniform directivity.

This is not a constant-directivity horn, but a combination of woofer and horn with uniform directivity. My Dome Horn is optimized for 6.5″ woofers, but it should also work well for woofers from 5″–8″. The Dome Horn is more directional than a regular dome tweeter, and so will add less ambience, but if you desire more, you can easily add it with today's electronics.

A further advantage of horns has to do with time delay. In some loudspeaker systems it is desirable to move the tweeter farther back so that the sound from the woofer and the tweeter tend to arrive at the listener at the same time. You can use a horn to move the tweeter backwards and still have a flat and verti-

Front piece — 7.25″ × 7.25″ × .75″; 5″ hole
2nd piece — 5.5″ × 5.5″ × .75″; 3″ hole
3rd piece — 5.5″ × 5.5″ × .75″; 2″ hole
4th piece — 5.5″ × 5.5″ × .75″; 1½″ hole
Rear piece — 2″ outer diameter and 1¼″ inner diameter; .25″ thick
Gasket — 2″ outer diameter and 1¼″ inner diameter; .125″ thick

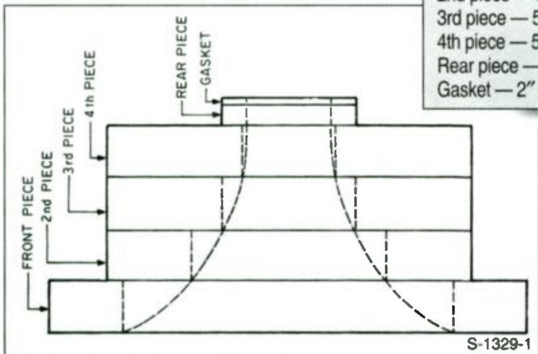


FIGURE 1: Cross-section view of dome horn.

cal front panel. To accomplish this, many monitors now use a waveguide, which is basically a very short horn. A waveguide about 1″ deep can double the sound output of the tweeter in the 2–5kHz region.

DESIGN

Long horns tend to give more acoustic gain and narrower directionality, but also more ringing and peaking. The Dome Horn is slightly over 3″ in length, which is much shorter than the 9″ maximum recommended by some authors. The horn's flare is intended to be exponential, but you can modify this. The Dome Horn would qualify as a short, symmetric horn with a moderate angle.

Measurements show that it is effective down to about 1kHz, which is low enough to be useful for dome tweeters. You can use this horn with almost any of the 1″ or 1⅛″ dome tweeters on the market today. I say almost, because some already come with waveguides or short horns as part of their front surface, and it is difficult to get these to mount well to this horn.

CONSTRUCTION

Over the years I have been interested in making horns, I've used cardboard, balsa wood, clay, and other materials. Lately I've achieved better results with Sculptamold, made by American Art Modeling Clay Co. This material seems to be a combination of plaster and papier-mâché. It has papier-mâché's ease of use, yet you can sand and shape it.

The horn is constructed of pieces of ¾″ wood (*Fig. 1*). The front piece of the tweeter horn is normally a 7¼″ square, but, as *Photo 1* shows, I've extended the front piece to a height of 14″ and mounted a 6½″ woofer to this panel. The front piece of the tweeter thereby becomes the front panel of the speaker enclosure. I used an MDF board 7¼″ wide for the front piece, and for the three middle pieces a 5½″-wide board cut into 5½″ squares. The rear piece is ¼″-thick Masonite, in the form of a washer with an inside diameter of 1¼″ and an outside diameter of 2″.

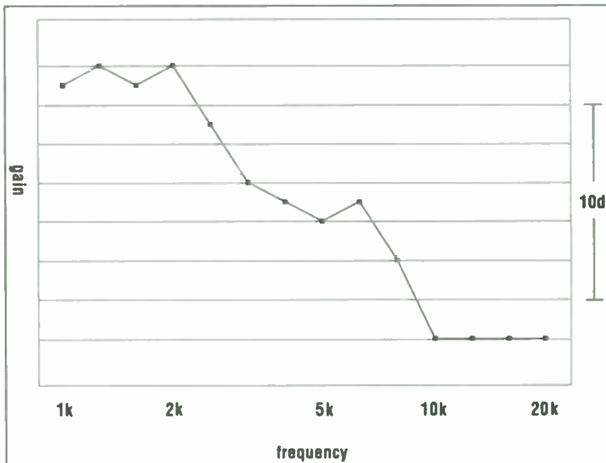


FIGURE 2: Dome Horn acoustic gain. S-1329-2

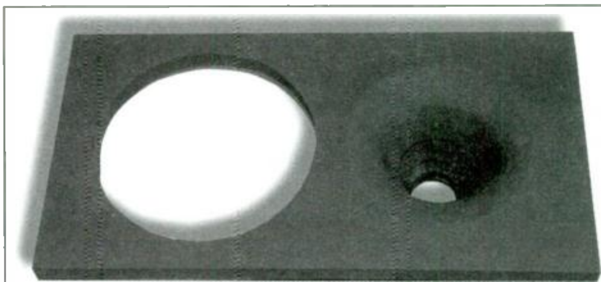


PHOTO 2: Finished horn.

To keep the construction concentric, drill a 3⁄16″ hole in the center of the four MDF pieces. Put a large nail through all four pieces to align them. Drill 3⁄16″ holes about ½″ in from each of the four corners of the three middle sections. Then insert 2.5″ drywall screws through these corner holes and into the front piece. Mark the top of each piece.

Disassemble the sections and cut the holes in the center of each. A hole saw is useful for the smaller holes, but you can use a saber saw for the holes with diameters larger than 2″. Reassemble the sections, gluing and screwing them together. Use only glue for the rear piece. You now have a very sturdy form for the horn, with the center holes guiding its final shape.

Mix up the Sculptamold modeling compound, about two cups worth, and apply it. Press it into the form, trying to achieve a smooth transition between the sections. Neatness is not necessary. Use more than you need, because you can sand it and shape it after it is dry. Sculptamold does not seem to shrink as it dries, but it does take a couple of days to dry, especially if you use too much water.

After the modeling compound is dry, you can sand it into shape. Actually, it is easier to shape with a wood rasp when still only partially dry. Shape it until the center holes show as circles in the horn, and you will have the correct horn shape. If you wish a different shape, cut different-sized holes. Sand it smooth, and then paint it if desired. *Photo 2* shows my finished horn.

Many dome tweeters have screws protruding above the front surface, and some have uneven front surfaces near the dome. It is desirable to reduce air leaks between the horn and the tweeter, so I used a gasket made of ⅛″-thick neoprene rubber between the tweeter's front plate and the horn's rear piece. Cork, felt, and foam rubber are other possible choices for the gasket, which I made with the same inner and outer diameters as the rear piece.

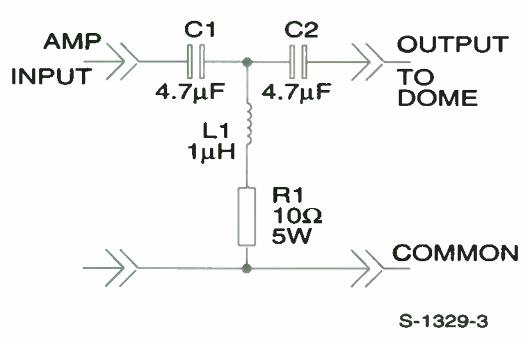


FIGURE 3: Dome Horn crossover schematic.

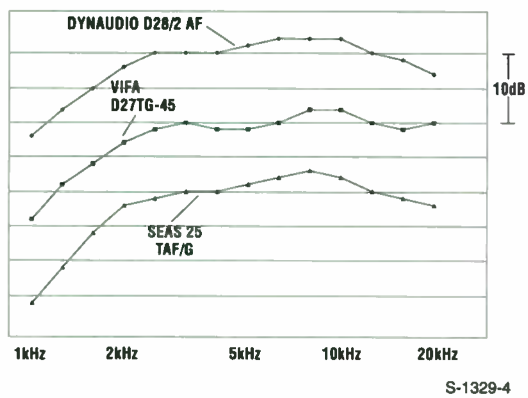


FIGURE 4: Frequency response of crossover with several dome tweeters.

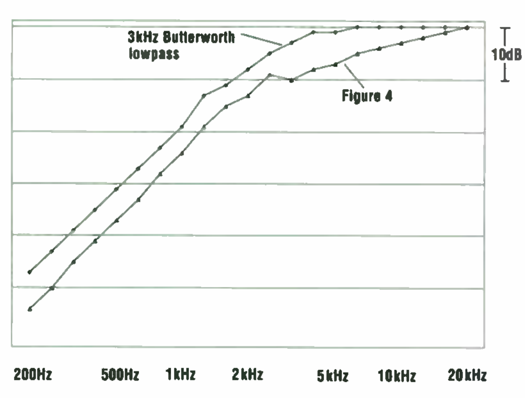


FIGURE 5: Crossover frequency response compared with a Butterworth low-pass filter.

MEASUREMENTS

I measured the acoustical gain of the horn using a 1⅛″ dome tweeter (Dynaudio D28/2 AF). *Figure 2* is the difference in frequency response of the dome at 1m with and without the horn. The horn makes little difference above 10kHz because the dome's diameter is larger than a wavelength at these frequencies, so the Dome Horn offers little increase in loading or directivity.

Between 1–10kHz the horn gives significant gain, since it is about 10dB at 2kHz. This means that for a given acoustic output at 2kHz, the horn tweeter needs only one-tenth the electrical power. The Dome Horn consequently increases the power-handling capability of the dome tweeter. At the same acoustic levels, the Dome Horn should be more reliable than the dome tweeter by itself.

CROSSOVER

The acoustic gain of the tweeter means that the tweeter crossover networks need to be revised. *Figure 3* is the schematic of a high-pass crossover, and *Fig. 4* is the frequency response of this crossover with several dome tweeters. The combination of tweeter, crossover, and horn works well for the Dynaudio, Vifa, and Seas domes. You can use others, but the crossover would need to change.

Notice that this crossover gives an acoustic crossover of about 1.6kHz for the Dome Horn. This is a low crossover frequency, and if you wish a higher one, you must modify it. In particular, for a higher crossover frequency, you would need to decrease L1 and R1.

One concern is to avoid frying the tweeters. Most 1″ dome tweeters are rated to perform reliably with 12dB-per-octave high-pass filters of 2.5kHz or higher. *Figure 5* shows the frequency response of the crossover in comparison with a 3kHz Butterworth low-pass of 12dB per octave. This crossover always delivers less power to the tweeter than the 3kHz low-pass, so it should be safer for them.

CONCLUSION

I am currently designing and testing a speaker system using this horn and a 6.5″ woofer (Dynaudio 17WLQ4) that has a 3″ voice coil. The horn is designed so that if the woofer were modified, a horn of this shape could fit through the middle of the woofer, with the final flare of the horn being the woofer's cone. This would be a modern version of the famed Tannoy loudspeaker, but using modern dome tweeters and in a smaller size. This should make for an excellent monitor loudspeaker with the ability to use the best dome tweeters. If any woofer manufacturer is interested in this idea, send me a letter.

In summary, I've discussed why you may want a horn, described how to build it, and shown results with several dome tweeters. The horn construction is inexpensive, easy, rugged, and uses normal tools.

Shown here is Morel's coaxial speaker, just one of Morel's high quality products.

You don't need to sacrifice valuable cargo space in your car for quality sound. The author shows how to achieve this with a few simple modifications to your present system.

The Sensible Sub

By Bill Fitzmaurice

You've heard them. You've felt them. Long before you can see the vehicles in which they reside, your ears are assaulted by the "Boom-Boom-Boom" of the monster subwoofers, most often $4ft^3$ reflex boxes nestled in the rear of rusting hatchbacks.

It is only a matter of time before the vibrations either shake the car to pieces or reduce the remnants of the brain of the (usually) 16-year-old driver to the consistency of tapioca pudding. As Andy Rooney might ask, "Why is it that most $1,000 auto stereos seem to be housed in $100 cars?"

PHOTO 1: Tweeters mounted on car doors.

Well, my new '99 Cougar cost me a bit more than a hundred bucks, and while I had no interest in occupying half its cargo space with an oversized speaker cabinet capable of causing a cattle stampede two counties away, neither could I tolerate "factory sound." However, by making a few fairly simple and reasonably cheap improvements to the OEM system, I've achieved good sound without giving up much usable cargo space.

FIRST MOD

The factory FM/tape player actually sounded fairly decent; I was able to live with it almost a week before I made the first mod, the addition of tweeters. The stock "premium speakers" were 5" × 7" whizzer cone jobs with what looked like 5-oz magnets, though they actually covered the 100Hz-5kHz range fairly well. To get the high end up to snuff, I mounted four piezo tweeters, two up front and two in the rear.

The units I bought are marketed by Dayton Loudspeakers, although I found the identical drivers at Radio Shack as well (stock #40-1391) for $29.99 a pair. You can surface-mount these mini-drivers just about anywhere; they need only a small hole for the wire. I mounted the front units on the car doors (*Photo 1*), and the others on the rear pillars, as close to ear level as possible for imaging. These drivers have very high efficiency, require no crossover, and do a creditable job within the confines of auto sound.

With the high end now fixed, the lack of deep bass became even more obvious, so I set about seeking a good driver for a sub. I desired an eight-incher with good specs and dual coils; I found it in Madisound's model 81524DVC. With specs of f_S=32Hz, Q_{TS} = 0.31, V_{AS} = 39 ltr, and an 80W rating, it will work well to 40Hz in a small box, with the 4Ω coils well suited to autosound. At $37, the price was right, too.

I decided to mount the driver in a cabinet located in an area that had no value for cargo carrying—the space behind the rear wheelwell (*Photo 2*). I removed the carpet lining the fender and used expanding foam to seal all the airways and chambers in the sheetmetal. I then cut cardboard templates and used them to trace onto ½" plywood the parts to be cut out with a sabre saw. I assembled these plywood sections into a three-sided box to fit behind the wheelwell (*Photo 3*). The holes I cut were for the driver and a 2" PVC duct.

The fender and trunk floor formed the remaining sides of the cabinet, which I attached to the floor with hot-melt glue. After the glue had set, I injected expand-

PHOTO 2: The space for the driver behind the rear wheelwell.

ing foam into the gaps between the cabinet and the sheetmetal. It was necessary to do this in a number of steps, allowing about an hour of setting time between applications, since many of the gaps were too large to be filled in one shot (*Photo 4*). The excess foam that splattered around was impossible to clean up when fresh, but when it hardened, I easily chipped it off. When the foam was completely cured, I used a long carpet-installer's razor knife to trim it flush to the contours of the sheet metal.

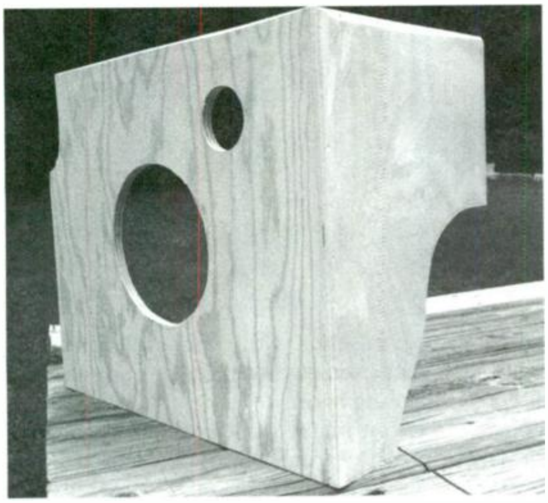


PHOTO 3: Three-sided enclosure shaped to fit behind the wheelwell.

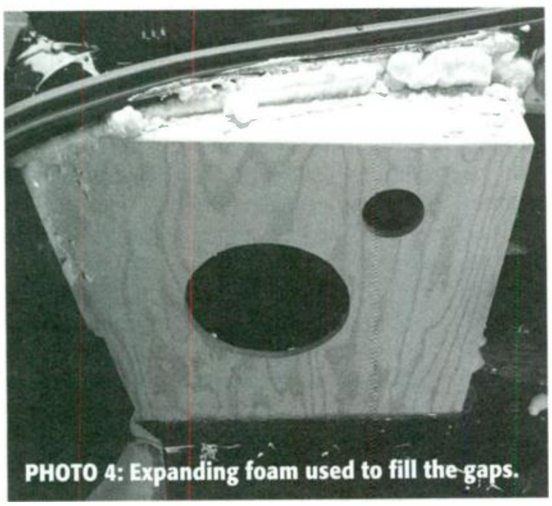


PHOTO 4: Expanding foam used to fill the gaps.

PHOTO 5: The finished sub.

Attaching carpet to the finished cabinet with spray glue and installing a steel grille and decorative port flange resulted in an appearance you could easily mistake for original equipment (*Photo 5*). I completed the entire job in six hours, spending the foam-curing time in installing and wiring the crossover and power amp for driving the subwoofer.

TIPS AND HINTS

Should you attempt a similar installation in your vehicle, here are a few tips. First, the volume of the finished cabinet is almost impossible to estimate (in my car I think it is about 20 ltr). This makes deciding on a duct size very difficult without impedance-testing equipment. If you have the test equipment, a box frequency of 30–35Hz is the target; if you don't, you'd be better off by using a closed box.

Next, use a separate amp to drive the cabinet, since it takes more power than a dash unit provides to drive this speaker effectively. Perhaps most important is to use an active crossover. I decided on a Radio Shack model 12-2012, which, while only $29.99, still has all the necessary features. Its high-level inputs make it possible to tap your existing speakers for signal, three high-pass frequency choices (I set mine at 50Hz), and bass-boost in addition, so you can easily compensate for whatever sonic deficiencies the sub cabinet may possess.

A good location for installing the amp and crossover is in the area behind the wheelwell opposite the cabinet. Also take into account the access to the taillights for bulb replacement. On my car this requires removing the driver; make sure you leave a way to access them in your car.

Finally, when you tap the factory speakers for signal, tap both sides. While most modern recordings have all material below 100Hz shared on both channels, many older recordings have the bass or drums hard-panned to one side. If you tap only one channel and feed that to both sides of the power amp for the sake of convenience, you might regret it later.

When you first fire up the system, check the phasing, trying the input connections both ways to see which gives more bass. At first, wire only one coil of the woofer, making sure it is in phase with the rest of the system, and then wire the second coil to its amp channel, making certain the two woofer coils end up in phase with one another. You'll also need to adjust the volume of the sub amp by trial and error for correct balance, since the bass response in the trunk area can be far too loud to achieve a proper response in the passenger compartment.

Should you decide to compete with the 16-year-olds for "maximum boom," you could build two cabinets, one for each side. Wire the voice coils in parallel, and use a 100W/channel amp capable of driving 2Ω loads (most are). My own experience is that one cabinet is sufficient, and 25W/channel is enough to do the job; the money I saved will cover at least one payment on my car.

High praise for LAUD, a handy tool for audio measurements. But before you begin, check out this easy-to-assemble project that will give you peace of mind when using this computer-based product.

Interface for LAUD Measurement System

By Tom Perazella

"The good old days!" We've all heard how good the "old days" were. Gasoline was 29 cents a gallon, movies were 25 cents, there were no telemarketing calls to intrude on your dinner, people were friendlier, and so forth, ad nauseam. Well, maybe there are a few isolated nuggets of truth in those sayings, but most of them are based on remembering the good things in a better light than they really were, and conveniently forgetting the bad.

When it comes to audio measurements, however, make no mistake. The good old days are right now. There has never before been a time when so many tools and techniques were available to make simple, fast, and accurate measurements of audio parameters, whether electrical or acoustic. A tool I have found extremely useful is the LAUD measurement system from Liberty Instruments.

LAUD SYSTEM

The LAUD system is based on a software suite that allows you to use a personal computer to measure, display, print, archive, and transfer various audio parameters. The software works in conjunction with PC sound boards that have specific digital signal processors, such as the Fiji or Pinnacle boards from Turtle Beach. The software controls both the generation of test signals on the sound board and the measurement of the return signals by the board.

(The range of measurements LAUD provides is too great to detail here. You can find more information and a demo on the Liberty Instruments Website, www.libinst.com. The postal address is 6572 Gretel Drive, Middletown, OH 45004.)

When using a computer-based tool such as LAUD, there are two areas of concern: setting levels of signal attenuation, and protection of the sound board. Even though the sound cards used have level control, it covers only a limited range. For the best measurement results, you should use external means to control to an acceptable range the level of signals reaching the sound card. Think of it as coarse adjustment, with the sound card making the fine adjustment.

The sound cards also have rather strict limits on the acceptable maximum signal levels they will accept. Violate those levels, even for a very short time, and you will not only have high levels of distortion, but also very likely bad sounds and smoke coming from the computer. The new sound cards are very good, with dynamic ranges of 20 bits. They are also a bit on the costly side, with prices ranging over $200. Turning one of these cards into a crackling, smelly hunk of nonfunctioning silicon and glass epoxy can ruin your whole day.

VOLTAGE LIMITATIONS

Be aware that these boards do not like to see over 5V. Above that and you are in tiger country. Although most cards have some sort of over-voltage protection, that capability is usually limited to sinking currents of around 10mA. Go beyond that level, and, as Bill Murray would say, "You're toast."

The instructions supplied with LAUD repeatedly make it clear that the boards are at risk, and that it is your responsibility to protect them. To assist you in that task, directions are given on how to make plug-style attenuators that you can use in series with your test leads to prevent excessive voltages or currents from causing harm.

You might wonder why you need to provide protection when you are measuring acoustical signals. Most, but not all, microphone preamps will not produce over 5V or more than 10mA of output. However, in the course of many measurements, LAUD uses a reference signal from the source driving the speaker to compensate for any irregularities in response that may be present in that source. So you can achieve a true picture of the speaker under test, even if the source is less than perfect.

A 5V peak AC signal across an 8Ω load produces a power dissipation of just over 1.5W, so it's clear that just about any amplifier you use for speaker testing will generate more than 1.5W, thereby putting your PC at risk. An amplifier that can produce 200W into an 8Ω load will have a peak output voltage capability of 57V. You cannot tolerate that level of voltage at the input of your sound card for even a moment. If you make a mistake, permanent damage will occur before you realize you have erred.

SAFETY DEVICE

Being basically lazy and prone to errors, I decided to build a device that would provide safe and repeatable controls of the input signals to my Fiji board. The goals were to have two channels (one for the measurement, and one for reference), switch-selectable attenuation levels, and protection from all but direct connection to the main substation of the local power company. The protection feature should be foolproof; that is, forgetting to perform a function would produce an inac-

curate reading, but not any damage to the sound card.

The first step was to design an attenuator that would enable me to adjust the input levels with a high degree of accuracy and precision. Throughout the following discussion, I will describe a single channel, but the second one is identical except for its designations. I used a two-pole, five-position rotary switch to select among the five different operating conditions shown in *Table 1*.

TABLE 1
ROTARY-SWITCH FUNCTIONS

SWITCH POSITION	RESULTING ACTION
1. Off	Disconnects the input signal from the sound card and shorts the output of the interface to ground.
2. Spare (future X100)	At the present time, disconnects the input and output jacks of the interface.
3. X10	Passes the input signal through a resistive divider that will result in a tenfold attenuation with the LAUD program.
4. X1	Passes the signal through a resistive divider that will result in a onefold factor with the Laud program.
5. Direct	Passes the input signal directly through the interface to the output jacks.

For a schematic representation of the switch functions, see *Fig. 1*. For the dividers, I used 1% ¼W metal-film resistors. The LAUD manual gives resistor values for both the 1X and 10X probes, assuming the nominal Fiji input impedance of 10kΩ. The manual specifies a two-resistor divider for both the 1X and 10X configurations. For the 1X probe, the high-side resistor is 7.68k and the low side is 2.87k. The 10X uses a 9.76k on the high side and a 232Ω on the low side. However, doing a little math on the 10X divider showed that a single resistor on its high side could result in excessive power dissipation, and possibly failure of that resistor.

Instead, I chose to use ten resistors in series on the high side of that divider to achieve the desired value. That combination resulted in a dissipation capability of 2.5W, which was more than adequate. As an added benefit, nine of the resistors were selected as a standard 1k value, and the tenth as 845Ω, the total giving a division ratio closer to ten times that of the 1k probe values. Realize also that the input impedance of the Fiji card is in parallel with the low side of the divider and becomes part of the equation.

The 5V limit for the sound card assumes that the PC is on at the time. When power is not applied to ICs, they are more susceptible to voltage damage. Leaving the switch in the off position until you actually make measurements provides another level of protection for the sound card, since the input to the card is kept at ground.

That explains the part of the design that appeals to my lazy side, because the switch allows easy changes of the attenuation levels to LAUD, and also provides a quick way to disconnect the sound card from the measurement probes without needing to physically remove them from the device under test.

PROTECTION METHODS

For the protection part of the interface, I considered several methods. Two potential circuit configurations I rejected were the use of zener diodes from the line to ground, and a sensing circuit to trigger clamping devices or open a relay.

The zener-diode approach is not good in this application, since zeners have nonlinear reverse characteristics across most of their operating range. Even well below the knee point, there is significant nonlinear current flow. Zeners also exhibit noise, which can be detrimental to the measurements you're making. Using a capacitor across the zener to reduce this noise would introduce a significant shunt path to ground for high-frequency signals. Essentially, it would add a first-order low-pass filter to the measuring circuit.

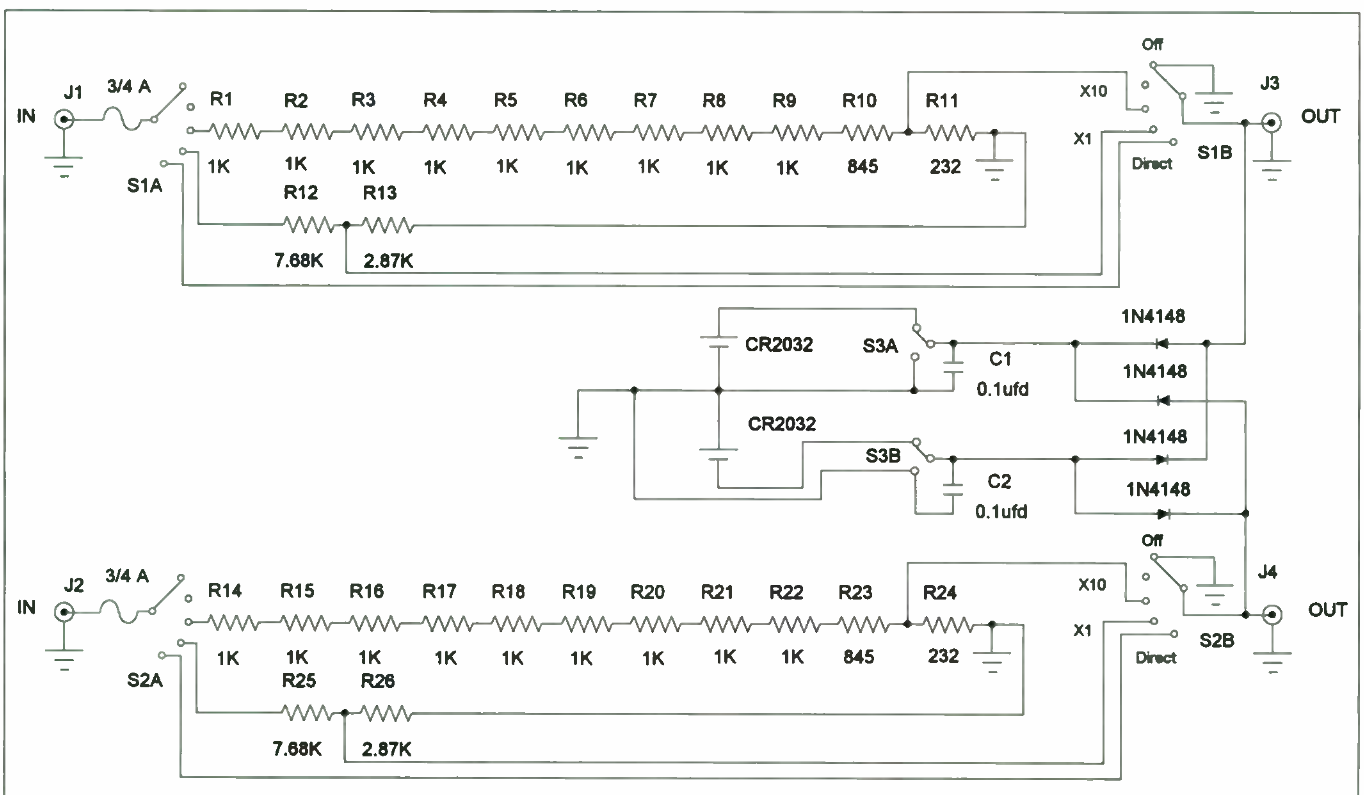


FIGURE 1: Schematic of interface circuit for LAUD.

Using a control circuit to monitor the signal circuit would not only add unnecessary complexity to the circuit, but would also require some sort of power supply. In addition, shunt devices such as silicon-controlled rectifiers (SCRs) have a significant amount of junction capacitance, resulting in the same type of roll-off problem as previously mentioned. Timing is an issue. The reaction time of a triggered clamp circuit can be a problem, and would certainly be so if you were to use a relay to open the signal circuit. Most ICs will be history long before a relay can react to open the path to a fast-rising overload.

I finally decided to use an old method for clamping signal voltages: clamp diodes connected to the input circuit that are reverse-biased at an appropriate level. *Figure 2* shows a basic implementation of that circuit for a positive limit function. The circuit operates on the principle that when an ideal diode is reverse-biased, it is essentially an open circuit; that is, no current can flow through the reversed junction.

When the signal voltage on the anode side of the diode reaches that of the cathode side plus the junction forward-voltage drop, current can begin to flow. The forward drop is not linear, and hence does not increase in proportion to the current flowing through the diode. Therefore, assuming the voltage source used to bias the diode has a low internal impedance, the current from the signal circuit will be shunted through that source.

The only further increase in voltage in the signal path will be that due to the change in diode forward-voltage drop with increasing current, plus the voltage drop across the impedance of the voltage source. Both should be relatively low compared to the impedance of most signal sources. Remember, except for the direct-connection path in the interface, resistive dividers add impedance to the input circuit.

MIRROR-PICTURE PROTECTION

In practice, the mirror picture of the *Fig. 2* circuit—referenced to a negative voltage source—also exists to provide protection against excessive negative voltages. The schematic of *Fig. 1* shows the final implementation of the protective circuit for both channels. For reference voltages, I chose two lithium batteries of the type used in PCs for CMOS back-up power because of form factor, long stable life, and easily available cells and cell holders. Type CR2032 has a button form factor, so it fits readily on a circuit board.

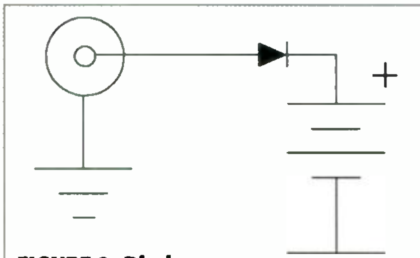


FIGURE 2: Diode clamping circuit.

Although the type of diode I used in this circuit, 1N4148, has very low leakage in the reverse direction, to disconnect the batteries from the circuit when it was inactive, I chose to use a DPDT switch, with one pole for each section of the battery circuit. As shown in the schematic, one side of each pole is connected to the ungrounded side of the batteries, with the other side connected to ground. The arms of the switch are connected to the junction points of the diodes from each channel.

A small capacitor from each summing point to ground provides a lower impedance path than the batteries for high frequencies. The capacitors in this configuration do not produce the low-pass effect mentioned previously, since they are kept high by the batteries and are essentially invisible to the circuit because of the reversed diodes.

When the switch is in the off position, the batteries are floating to prolong their life. The diodes' summing points are then connected directly to ground, so any input signals will be shunted to ground through the diodes. This is the fail-safe mechanism. If you forget to turn the circuit on, the diodes will not allow signal voltage to exceed their forward voltage, thus providing protection for any devices in the sound card.

BUILDING THE INTERFACE

The factors I considered most important in selecting a housing for the interface included electrical shielding, ease of machining, strength, and compact size.

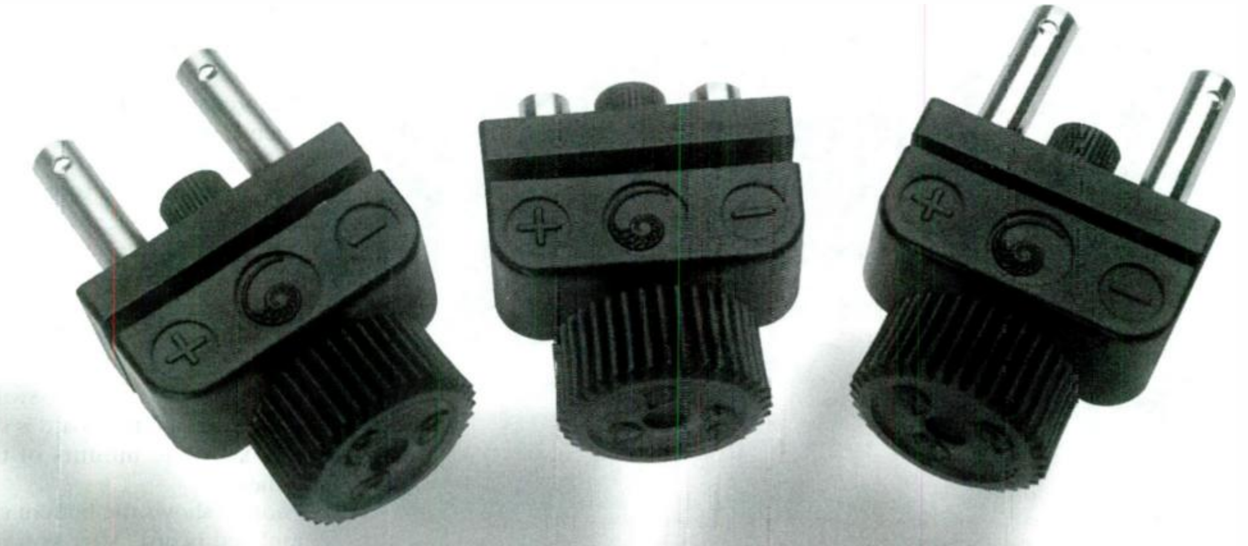
For many of my projects, I have found that using die-cast aluminum boxes is a good solution to these needs. Some people use plastic boxes, but I find them lacking in two areas—shielding and ease of machining. The shielding issue is clear. You might think that machining a plastic box would be easier, but this is really not the case. The aluminum alloy used in the cast boxes is very easy to cut or drill, especially when using a cutting fluid meant for aluminum. The resulting holes are very clean, with few if any burrs, and hole sizes tend to be more predictable.

Plastic, on the other hand, can be a real problem if you are not using tools designed especially for it. The material is rather soft, and using a standard drill bit, for example, can result in the plastic grabbing the bit, resulting in a very messy hole at best. In some cases, the bit will bind in the plastic, turning the box, if not

TABLE 2
PARTS LIST

QUANTITY	ITEM NUMBER	DESCRIPTION	PRICE TOTAL
FROM DIGI-KEY:			
2	CKC7003	Rotary Switch—two pole, six position	$10.26
1	HM159	Die-cast aluminum box	$10.90
2	10201	Black aluminum knob	$7.24
5	7.68KXBK	7.68kΩ ¼W 1% resistor	$0.54
5	2.87KXBK	2.87kΩ ¼W 1% resistor	$0.54
5	232XBK	232Ω ¼W 1% resistor	$0.54
5	845XBK	845Ω ¼W 1% resistor	$0.54
20	1.00KXBK	1.00kΩ ¼W 1% resistor	$2.16
10	1N4148DICT	1N4148 switching diode	$0.64
5	283-2364	Fast acting fuse 750mA	$3.75
2	E1104	0.1μF polyester capacitor	$0.58
1	V1119	Vector perf board 6.5″ × 4.5″	$5.43
1	V1069	Vector Micro-Klips pkg of 100	$5.32
FROM RADIO SHACK:			
2	270-430	Lithium-battery holder	$3.98
2	23-162	CR2032 lithium batteries	$5.58
1	275-663	DPDT 6A mini toggle switch	$3.59
FROM SOUND CONNECTIONS:			
2	CM7F	Sets of two RCA jacks	$19.00
			$80.59

Note: Quantities of some parts, such as resistors and diodes, are in excess of what you need for the project, but represent minimum purchase quantities. Paint and transfer type are not listed, since they were already available in my existing stocks.



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- *Conductors are soldered inside the contact posts for an extremely short signal path.*
- *Most metal spades are covered by the clamp, so it's CE friendly in Europe.*
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- *A single bolt locks the binding post base to the chassis or cabinet.*
- *Raised bosses on the back fit standard mounting holes so base cannot rotate.*
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- *Posts available in bare metal or Rhodium, Gold, Silver, or Nickel plating.*
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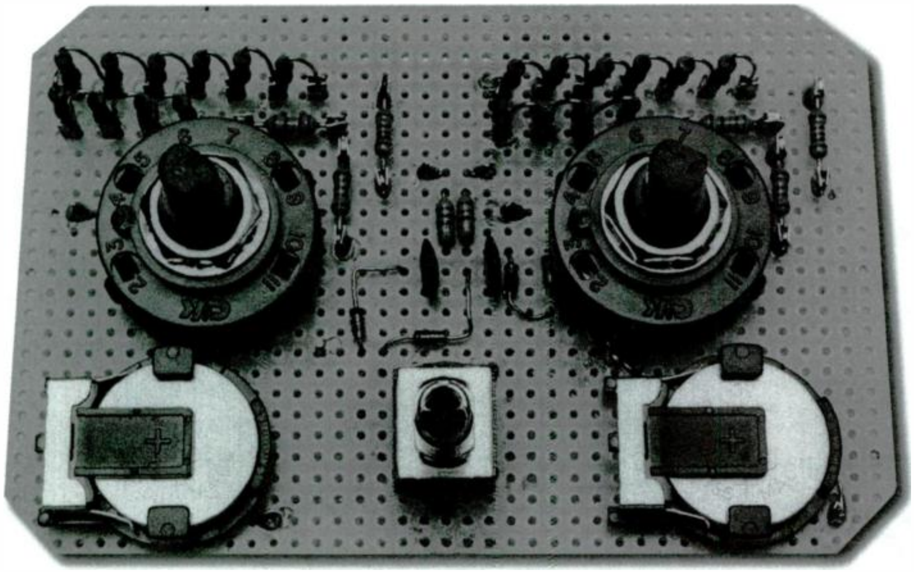

PHOTO 1: Top view of completed circuit board.

securely fastened, into an out-of-balance spinning projectile that tends to smack everything in its path, including hands.

As with any project, the first job is to make a parts list (*Table 2*) and purchase what is necessary. Since this was a one-time project, I decided to use perfboard and terminals to construct the component board. This allows you to do your layout at the same time you're actually building the board. If you make a layout mistake, it's a lot easier to fix with this type of construction than with a PC board.

A quick layout of parts similar to those I would use in the project indicated they would fit into a standard box about 4¾″ × 3⅛″ × 2¼″. I then either ordered the parts or dug them out of my home supply.

The first stage was to build the board itself. With perfboard, you can either use terminals or mount the parts directly to the board. Most of the time I use terminals, but in tight quarters, or when mounting IC sockets, I generally skip them and use the direct approach. I used both techniques on this board. *Photo 1* shows the component layout on the top side of the completed board.

COMPONENTS

The major components are the two selector switches, the two lithium batteries in their holders, and the on/off switch, which I hesitate to call a power switch, since it provides no power to anything, but simply connects the measuring circuits to the protective networks. "On" refers to the condition where the protective diodes are connected to the batteries. When it's in this state, you can measure normal voltages with the umbrella of the protective circuit to guard against accidental overloads.

"Off" indicates that the protective diodes are connected to ground, clamping any voltage that may appear at the output jacks of the interface to the forward voltage drop of the diodes. Initially, I mounted the switch to the board instead of the box, since it was easier to make the electrical connections on the board, and then secure the switch to the box later. In fact, the board is secured to the box by the mounts of the three switches.

Photo 2 shows the bottom side of the completed board. After you've located the main components on the board, add the resistors, diodes, and capacitors. Make the connections with #24 solid-copper hook-up wire. The two larger insulated leads are the connections from the on/off switch to the clamp diodes, and using insulated wire for these makes the layout easier.

As with any construction project, it is advisable to mark and drill all the holes in the box before painting. Once the board is completed, you can precisely measure the switch positions and drill the holes in the box. Then insert the board into the box and check that it fits. When it is in position, mark the locations of the RCA connectors on the box, allowing sufficient clearance to avoid interference with the board. Then remove the board and drill the holes for the jacks.

FINISHING

Before painting, lightly sand the box to provide a better surface for the primer, and then clean with solvent. After priming bake it in the oven at around 175° for about two hours. I have found that baking standard spray-can finishes results in a much harder, more permanent finish. Let the box cool to room temperature and then spray it with a matte black finish, followed again by the baking routine.

For lettering, I used white press-on transfer type. Art stores usually have a reasonably good selection. The advantage of this type is that if you want to make a change, you can carefully remove it without damaging the box. Its disadvantage is that repeated touching can wear it off. You can protect the type by spraying it with a clear acrylic or urethane, but the final finish will not be as flat as the basic matte black paint. For

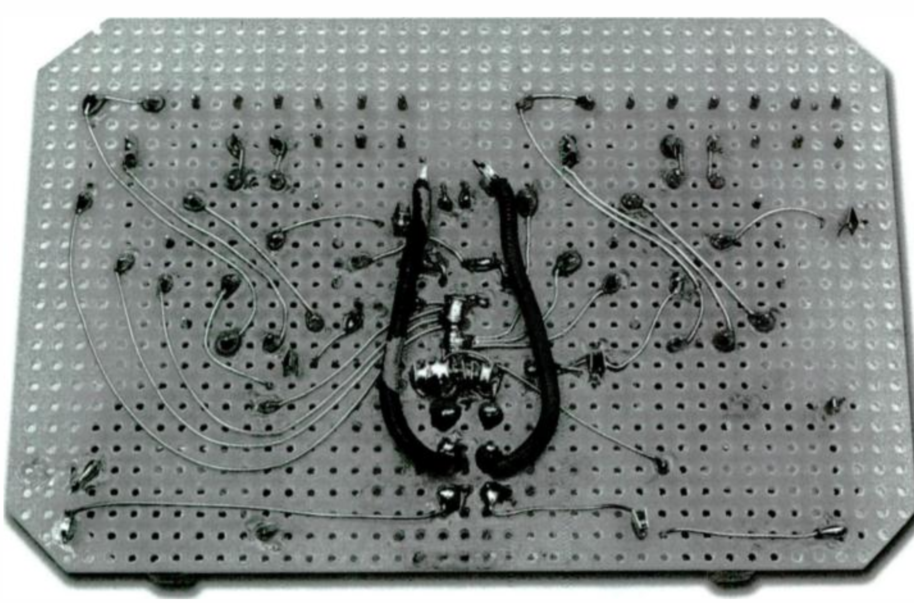

PHOTO 2: Bottom view of completed circuit board.

scan-speak All your senses are related.

Type	Ohm	dB 2.83V /M	Fs (Hz)	Mms (g)	VC Ø mm	Qms	Qes	Qts	Vas ltrs	Power W,kHz,dB/Oct	Special Parts	Mounting Flange/Cutout Ø mm	X-max Lin. / Max. +/- mm
Tweeters													
D2008/8512	5.7	90	800		19					150 / 4.0 / 12	Ferro fluid	92 / 69	
D2010/8513	5.7	90	800		19					150 / 4.0 / 12	Ferro fluid	98 / 69	
D2905/9300	4.7	90	650		28					150 / 2.5 / 12	Ferro fluid	104.5 / 80	
D2905/9500	4.7	90	550		28					150 / 2.5 / 12	Ferro fluid, NRC	104.5 / 80	
D2905/9700	4.7	89.5	500		28					225 / 2.8 / 12	SD-2, LCD, NRC	104.5 / 80	
D2905/9900	4.7	91	500		28					225 / 2.8 / 12	SD-2, LCD, NRC, CDD	130 / 80	
D3806/8200	5.7	90	450		38					100 / 1.0 / 12	SD	125.5 / 100	
Midranges													
13M/4535	3.0	89	60	5.8	38	3.3	0.3	0.28	5	35 Full range	Paper, SD, R, G	130 / 103	2.5 / 6
13M/8636	5.8	86.5	77	4.6	38	2.8	0.36	0.32	3	100 / 300 / 12	Kevlar, SD, R	130 / 103	1.5 / 5
13M/8640	5.8	87.5	64	4.1	38	2.7	0.27	0.24	5	100 / 300 / 12	Paper, SD, R2	130 / 103	1.5 / 5
Mid / Woofers													
18S/8535	5.8	86.5	26	15.5	38	2.5	0.47	0.40	72	70	CP, Shielded, SD-1, R3	177 / 158	5.0 / 10
18W/8535	5.8	86.5	26	15.5	38	2.5	0.45	0.38	72	70	CP, SD-1, R3	177 / 158	5.0 / 10
18W/8542	5.5	89	30	15	42	1.7	0.26	0.22	49	70	Paper, SD, Foam	177 / 158	6.5 / 10
18W/8543	5.5	88.5	30	15	42	1.4	0.26	0.22	49	80	Polypropylene, SD, R	177 / 158	6.5 / 10
18W/8545	5.5	88	28	20	42	2.3	0.30	0.27	48	100	CP, SD-1, R3	177 / 158	6.5 / 10
18W/8545K	5.5	87.5	28	20.5	42	5.2	0.30	0.28	48	100	CP, SD-1, R3, K	177 / 158	6.5 / 10
18W/8546	5.5	88	22	18.5	42	1.7	0.22	0.19	84	100	Kevlar, SD-1, R3	177 / 158	6.5 / 10
Woofers													
21W/8554	5.5	90	23	20.5	42	1.7	0.25	0.22	160	110	Kevlar, SD, Foam	222 / 194	6.5 / 10
21W/8555	5.5	87	20	32	42	4.5	0.33	0.31	136	100	HP, SD-1, R3	222 / 194	6.5 / 12
21W/8555-01	5.5	87.5	19	36	42	4.8	0.27	0.26	136	100	HP, SD-1, R3	222 / 194	6.5 / 12
25W/8565	5.5	88	20	43	42	5.4	0.44	0.41	229	100	HP, SD-1, R3	255 / 228	6.5 / 12
25W/8565-01	5.5	88	19	47	42	5.6	0.36	0.34	229	100	HP, SD-1, R3	255 / 228	6.5 / 12

SD = symmetric drive
SD-1 = optimized symmetric drive with transient soft clipping
SD-2 = optimized high frequency transient response
LCD = low compression design
NRC = non resonate chamber
CDD = constant directivity design
CP = carbon paper cone
K = Kevlar voice coil former
R = high loss rubber
R2 = low loss rubber
R3 = non resonant low loss rubber suspension
HP = hard paper
G = metal grill

The Scan-Speak product philosophy is based on the goal to produce the highest quality "true to life" loudspeaker drivers possible, capable of being used in the finest audio systems available anywhere in the world, regardless of cost.

We define "true to life" sound quality as reproducing a piece of music that is so detailed, dynamic and engaging that you believe you are at the original performance. This experience could include hearing such subtle details as the whispers of approval from the audience in a small jazz club. Or it could involve the overwhelming impact of a Kettle drum played during a symphony in one of Europe's finest concert halls; Or even the charismatic, heart-felt voice of a soloist performing in the Choir of a large church.

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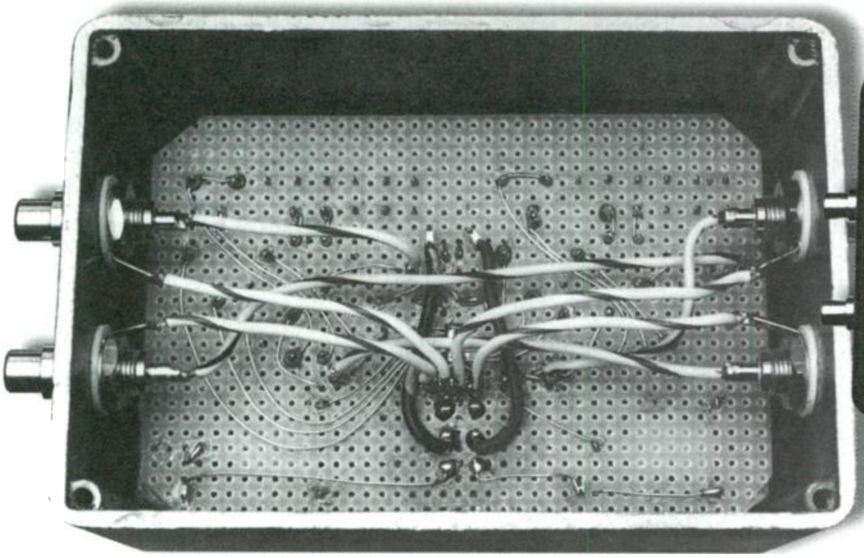

PHOTO 3: Connections to the circuit board inside the interface.

PHOTO 4: Top view of completed interface.

this project, I chose not to clear coat, but the choice is up to you.

Once the box is painted and lettered, the next step is to install the board. For easier assembly, solder to the board the leads and fuses that will connect it to the input and output jacks. As previously mentioned, the board is held in place by the two selector switches and the toggle switch. When tightening the nuts on these switches, take care to protect the painted surface and lettering from marring.

After the board is in place, attach the signal jacks to the housing. The best RCA jacks I have found for the money are Vampire from Sound Connections International. Signal jacks represent a relatively small portion of most construction-project costs, so it pays not to skimp here. I've used jacks from Radio Shack, Switchcraft, and others, but the Vampires have consistently provided the most reliable connections.

Now connect the lead wires and fuses to the appropriate jacks. *Photo 3*, taken before the input fuses were in place, shows the jacks and the connections to the board. In the final version, I connected the input fuses between the center terminals of the two input jacks and the board.

The last step of the assembly is to attach the bottom of the housing, put on four self-stick feet, and attach the knobs to the selector switches. *Photo 4* shows the top of the completed interface.

TESTING

Ideally, this interface should provide the degree of protection desired without affecting the quality of the signal, as long as it is within the levels allowed. In other words, it should act as a straight wire with attenuation rather than gain, and also as an "electronic axe" to chop off the tops of any signal levels if the output goes higher than desired.

The first test was to determine whether the circuit would limit signals to a safe level. I connected a signal source to the interface, which was set to the X1 position, and an oscilloscope—set for a vertical deflection of 1V/cm—to the output jack. For this test, I used a 1kHz sine wave, increasing the signal level to a point where the peak amplitude was 3V. As you can see in *Photo 5*, the signal shows no visible deformation.

I then increased the signal level so that the peaks approached 4V. *Photo 6* shows that the tops of the traces are limited to around 3.5V, the sum of the battery voltage and the forward drop of the clamping diodes. This is the clamping action I was looking for. Next, I increased the input signal level to produce peaks well beyond 5V. *Photo 7* shows the limiting action in effect, with peaks clamped around 3.6V.

With the first goal of peak-signal limiting achieved, the next test was to determine whether any signal degradation occurred. To do this, I set up a true straight-wire test as my reference. I connected the line out from the Fiji card to the line in, using a double female RCA coupler. I then ran a harmonic-distortion test at 0dB, with the results shown in *Fig. 3*. What appears below 0.01% is probably noise contamination from the wicked electrical environment that surrounds the Fiji board within the PC.

To test the interface, I removed the

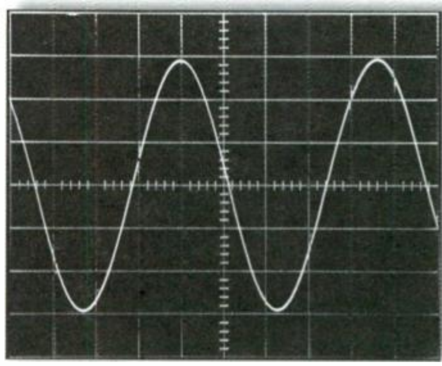

PHOTO 5: Test signal below limiting point.

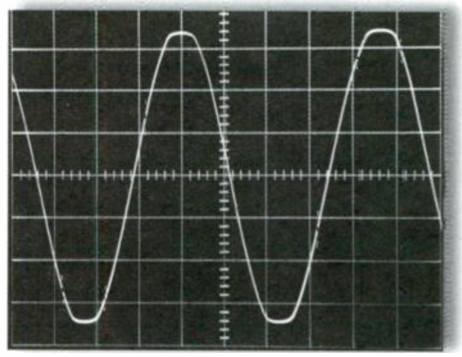

PHOTO 6: Test signal at the onset of limiting.

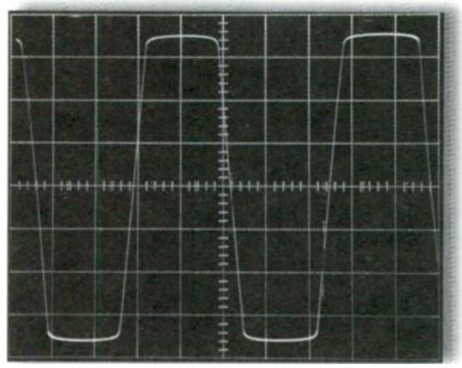

PHOTO 7: Test signal heavily into limiting.

RCA coupler from the setup, and inserted the interface in its place. I set the interface to the direct position so there would

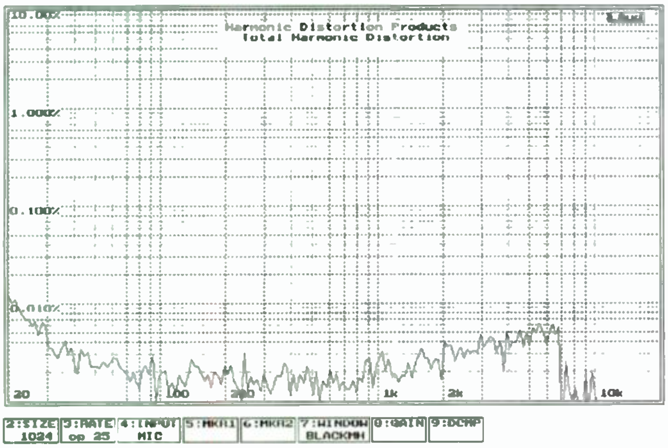

FIGURE 3: Harmonic distortion of direct-connection test signal.

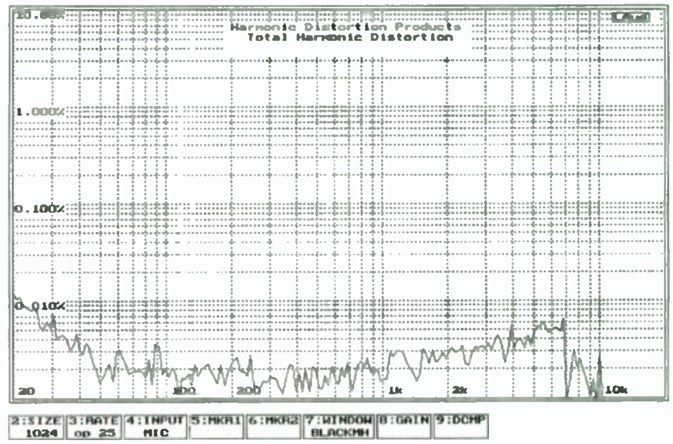

FIGURE 4: Harmonic distortion of test signal passed through interface.

be no attenuation of the signal, but the protection circuit was in place. I then repeated the harmonic-distortion test, with the results shown in *Fig. 4*. Again what appears is residual system noise. The interface does perform as a straight wire, introducing no distortions greater than background noise. The testing confirmed that the design goals were met.

SOURCES
Digi-Key 800-344-4539
Sound Connections International 813-948-2707
Radio Shack Local store

USING THE INTERFACE

In the months since completing this project, the interface has performed without problems. In a typical measurement setup, it sits right next to my PC, enabling me to make changes without needing to alter any equipment. There is no confusion factor, since all conditions are clearly indicated by the switch positions visible on the interface. And before you ask, I'll confess that I was clumsy in a setup one day, and the interface did save my sound card when I inadvertently pulled an input to an amplifier while the power was on. So my Fiji is still with me—a tribute to the effectiveness of the interface.

Like the Phoenix rising from the ashes, this author finds new life for his amp with this power supply/crossover fix. A small electronic excursion for speaker enthusiasts who like active crossovers.

Afterburner for Aftershock

By Philip E. Abbate

When I was in engineering school, I learned a little trick to play on the other engineers at work. Come into work early and stick a 1k 1W resistor in the electrical outlet on the bench of your favorite victim, and inform the rest of the early birds. When the laggard turns on his bench, we wait for the smell of burning electronic parts and the panic reaction from the victim. It is hard to keep from laughing as we watch the victim turn off the power and inspect his latest project for a failure, only to repeat the exercise until the resistor burns open or he sees the smoke.

NO JOKE NOW

Sixteen years later, when I came home and encountered that familiar smell, it was no practical joke. A 200W-per-channel Quatre gain cell power amp went up in smoke, taking one of my woofers with it. Seems my 14-year-old son and a couple of his audio buddies were running a police-response time test, with Rage Against the Machine playing at 110dB+. No problem; I had more amps and was a regular at Eddie's Speaker Recording in East LA.

One of the channels was burnt beyond repair, but the other one and the power supply were fine. I put the gain cell amp in the attic. My plan was to use the power supply and heatsinks for a ZEN amp or something someday. That day came when Aftershock (*SB* 6/96) went back into my home theater with the Aluma Pro Alusion driver (*SB* 6/98).

I needed a single-channel amp with an electronic crossover that would sum my Pro Logic receiver's left and right channels. The f_3 of my LE 14 woofers is a disappointing 50Hz since I plugged its ports, so I decided on a 50Hz crossover point for Aftershock, hoping it would integrate properly.

The power supply in the Quatre amp was a little too high in voltage to regulate down to ±15V DC, but I had a ±12V AC transformer and a pair of National Semiconductor three terminal voltage regulators handy, so I built a power supply for the crossover and attached it to the amp side of the fuse in the amplifier.

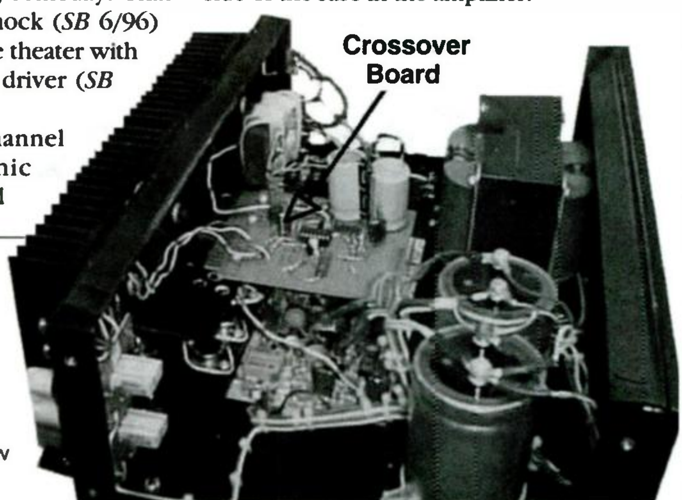

PHOTO 1: Innards of Afterburner.

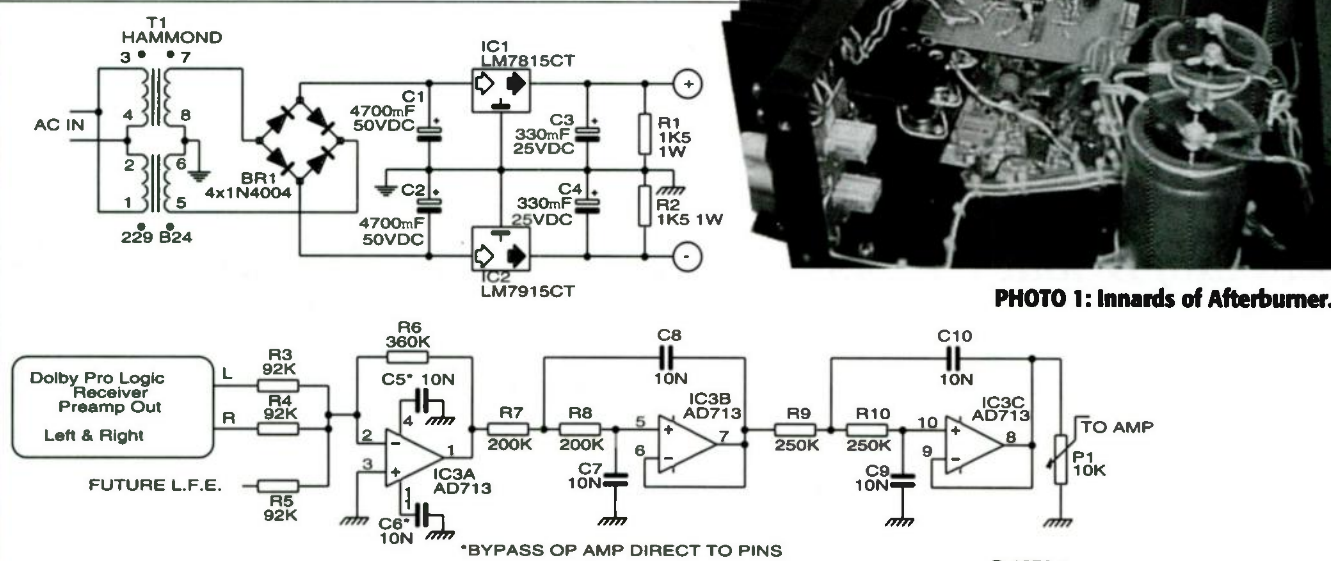

FIGURE 1: Crossover schematic.

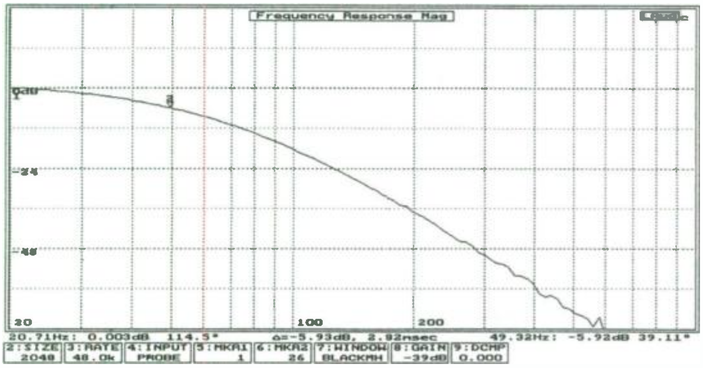

FIGURE 2: Frequency response of crossover and gain-cell amp.

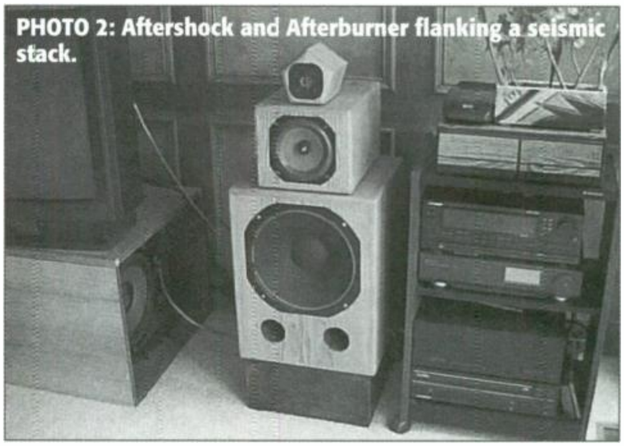
PHOTO 2: Aftershock and Afterburner flanking a seismic stack.

I also had a spare Analog Devices AD713 quad op amp and enough resistors and capacitors to make a fourth-order filter from spare parts. This project took one afternoon. I removed the burnt-out channel and replaced it with the power supply and active crossover filter. *Photo 1* shows the innards of the completed Afterburner.

POWERFUL SOUND

I desired plenty of wallop for my action movies, so I designed a summing amplifier with 12dB of gain. The summer has three inputs, two for my current Dolby Pro Logic system and one low-frequency-effects (LFE) input for a Dolby Digital System I hope to buy some day.

Resistor and capacitor values were determined by what I could dig out of my junk box. I used polypropylene caps for the active filters and ceramic for the power-supply bypass. I mounted all the regulator and filter circuitry on a Radio Shack prototype board designed to plug into an edge connector. All wiring was point to point. *Figure 1* shows the schematic of the project.

Since the resistors and the capacitors in each filter stage are the same value, you can calculate the break point of each stage as the reciprocal of $2\pi \times$ resistor $\times$ capacitor. The break point for the first filter stage (op amp IC3B) is the reciprocal of $2\pi \times 200k \times .00000001F$, or 73.22Hz. The break point for the second filter stage (op amp IC3C) is the reciprocal of $2\pi \times 250 \times .00000001F$, or 58.58Hz. *Figure 2* shows the composite electrical response of the two filters at the amplifier terminals. The –6dB point is about 50Hz, as planned. The –6dB point will be equivalent to the –3dB point for power. The octave from 200Hz–400Hz shows a rolloff of nearly 24dB per octave deep in the reject band.

I was a little worried about turn-on

to page 53

This speaker-building veteran offers a pair of transmission-line speakers with a detailed, musical sound well-suited for CD play, FM broadcast, VCR movies, and TV.

Another Look at TL Design

By John Mattern

The spark that started my current transmission-line (TL) project came from a neighbor's Bose Wave Radio. The acoustic-labyrinth concept was not new to me, but I was intrigued by the simplicity and wished to know how it worked. So I set out to build one on my obsolete but perfectly useful computer.

I was able to undertake a computer simulation because of my experience with both computer programming and electrical transmission lines. My knowledge of the latter was especially helpful because of their mathematical similarity to acoustic transmission lines. But the simulation was far from easy, since I still had much to learn about acoustics.

I built the first speaker using a 4″ woofer with a folded transmission line made from 3″ plastic pipe elbows. I also built a second device with straight pipes that I used to measure the acoustic-absorption characteristics of various materials. Those tested included samples of various foams, thermal insulation, and Radio Shack acoustic insulation. I chose to use the Radio Shack insulation on the basis of these tests and its availability. I concluded from this first effort that I could not accomplish my goal with this simple speaker arrangement.

STARTING OVER

I went back to the drawing board, in this case my books and one in particular written by Weems.[1] In it he described a transmission-line design that connected a 6.5″ woofer at such a point that it did not excite the line's third overtone. He also tapered the line, which looked as though it would increase the load on the back side of the driver cone.

It made sense, and I had faith that he was on the right track. I therefore modified my simulation by moving the driver out from the throat of the horn and also allowing the horn to taper. This is the

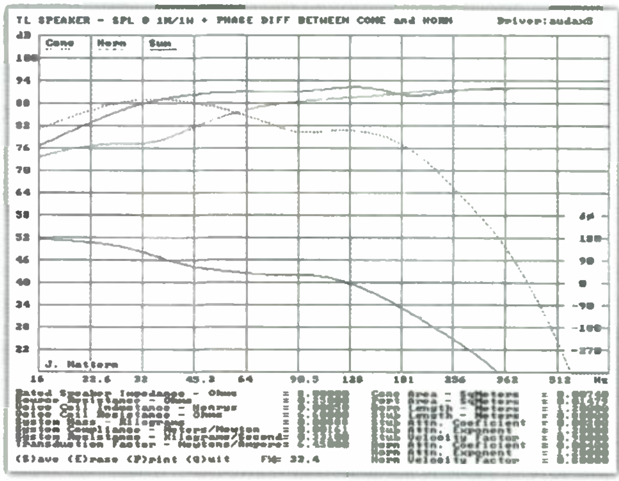

FIGURE 1: Simulated response of TL speaker, showing crossover between port and cone at roughly 60Hz.

PHOTO 1: The completed speakers set alongside the TV receiver. At this distance there is no effect of the unshielded drivers on the display.

ABOUT THE AUTHOR
John Mattern is a retired electrical engineer with many years of signal-processing experience at Westinghouse Electric.

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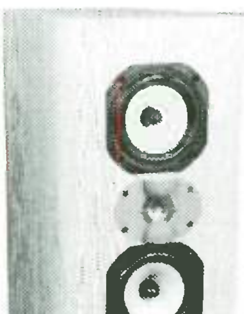

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version of the simulation used to produce *Fig. 1*. This did not happen overnight—by then more than a year had elapsed since I started the project.

Again I built the hardware, but this time I constructed rectangular wooden ducts instead of using plastic pipe. The simulation had narrowed the driver choice to two: the Audax 5.25″ with the TPX cone, and the Morel 5.25″. I chose the Audax because of its greater sensitivity. The new simulation verified the benefit of the new driver position and of the horn taper. Moreover, the simulation allowed me to experiment with the distribution of the acoustic absorber.

The ability to optimize the distribution of absorbent material proved to be very helpful. First, the simulation predicted that the material in the stub (the pipe from the driver to the throat) would have very little effect on the midbass, but considerable effect on the deep bass.

For the simulation, I limited the material in the stub to half of that in the horn, which the simulation indicated should be filled to no more than 66% with fiberglass. So in the latest version of the hardware, I filled only 33% of the stub with fiberglass. There was no noticeable midbass coloration that I could hear, but, as predicted, the deep bass was stronger.

MAJOR ASSUMPTIONS

Figure 1 is an optimized plot showing the response for the case where the front of the speaker enclosure is flush with a large plane and there are no other reflecting surfaces to interfere with the sound pattern.

The simulation allowed me to enter the following parameters in the calculations:

- electrical, mechanical, and magnetic parameters of the driver;
- lengths and areas of the stub and horn;
- the taper of the horn; and
- the absorption characteristics of the stub and the horn.

My object was to design a speaker capable of deep bass in the smallest possible size. To achieve deep bass from a small speaker requires more than just delaying the back wave by 180°. There must also be more acoustic loading on the back side of the cone than on the front side.

Figure 1 shows that the back side radiates 12dB more power at 32Hz than the front. Note that you cannot obtain this power level from the cone of a 5″ speaker without horn loading. The result is a bass-speaker volume of 1.3ft^3. *Photo 1* shows the completed bass and treble speakers with and without grille covers. They are a convenient size for placement next to a television. Although the treble unit is also a TL type, I have made no attempt to simulate its operation.

BASS-CABINET MATERIALS

Figure 2 shows the dimensions of the base enclosure. I made all the pieces from ¾″ birch plywood, but you can choose other materials. I decided on plywood because you can firmly bond it with carpenter's glue. I chose the ¾″ thickness since I hoped it would eliminate the need for internal reinforcement, which in fact it did.

You will find the panels are very stiff and acoustically dead in the sizes used. The Audax speaker has low internal pressures, at least at the frequencies where the plywood is subject to resonance. The bass-cabinet parts list (*Table 1*) shows the pieces and their dimensions. I cut all the 6″-wide cross pieces with the same setting on my tablesaw. You should also cut both 11″-wide side pieces with the same setting.

TABLE 1
BASS-ENCLOSURE PARTS LIST

QTY	DIMENSIONS	MATERIAL
2	7.5″ × 11″ × ¾″	Plywood panel (top & bottom)
1	6.0″ × 23″ × ¾″	Plywood panel (rear)
1	6.0″ × 5.5″ × ¾″	Plywood panel (top front)
1	6.0″ × 13.5″ × ¾″	Plywood panel (bottom front)
1	6.0″ × 20.0″ × ¾″	Plywood panel (inside rear)
1	6.0″ × 14.25″ × ¾″	Plywood panel (inside front)
1	6.0″ × 4.75″ × ¾″	Plywood panel (inside top)
1	6.0″ × 5.0″ × ¾″	Plywood panel (inside bottom)
4	6.0″ × 2.0″ × 2.0″	Corner piece (90°)
1	6.0″ × 2.0″ × 2.0″	Corner piece (85°)

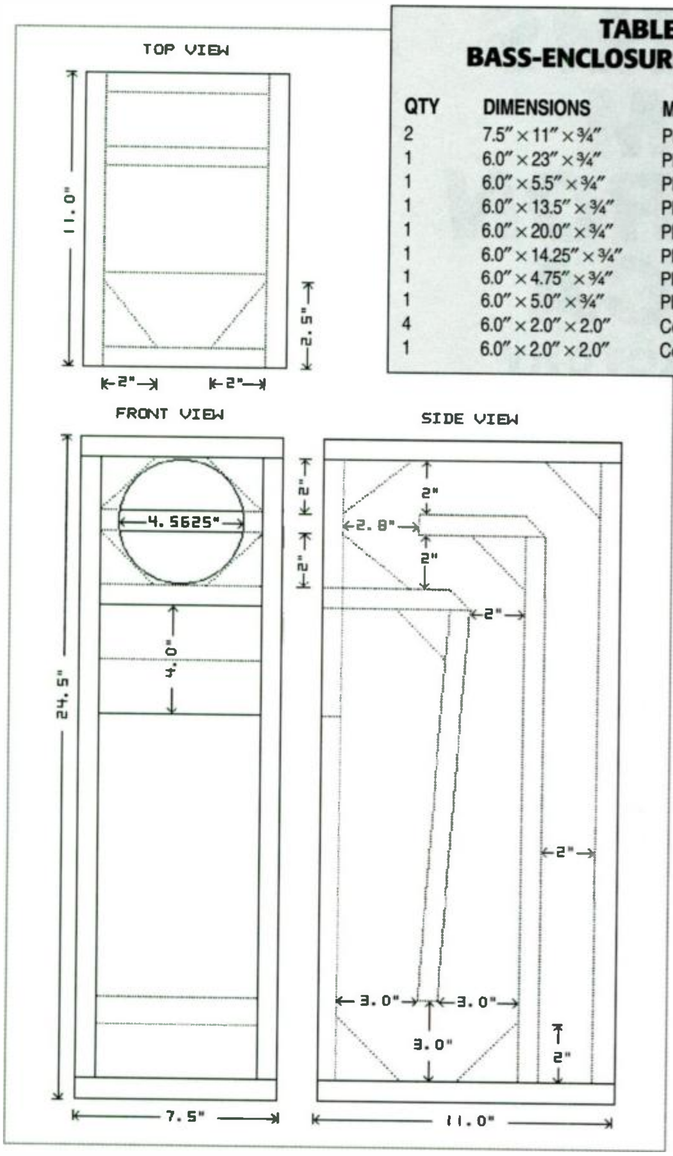

FIGURE 2: Bass speaker enclosure dimensions.

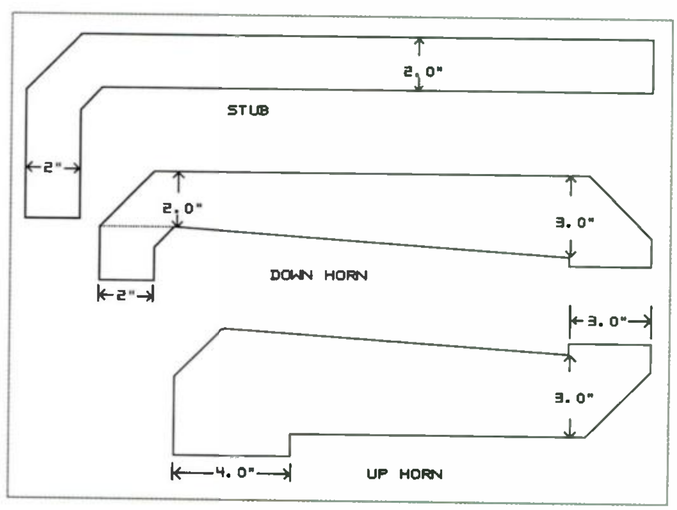

FIGURE 3: Stub and horn templates; see enclosure layout for additional dimensions.

BASS-ENCLOSURE ASSEMBLY

I assembled the bass enclosure in two steps, first without glue, using only nails to hold the pieces in alignment. I assembled the sides to the back, to the two front pieces, and to the internal pieces using 17-gauge finishing nails. You will find 2″ spacers helpful in holding the internal pieces in alignment during assembly, and you will need to do some fitting of those internal parts that are not at right angles. Do not install the corner pieces at this time.

Drill the nail holes for a snug fit with 3″ spacing. You may wish to use only the "end" nails for the temporary assembly. To make the disassembly easier, do not drive the nails all the way in. You can tack on the top and bottom at this time to prevent skewing of the assembly.

When all the pieces are aligned and secure, remove one side to apply glue to the edges of the back, front, and center pieces. After applying the glue, reinstall the side and drive home all the nails. Remove excess glue with a wet cloth, both inside and out.

When the glue has dried, remove the other side and complete the internal construction. Glue and nail the four regular corner pieces, and then fit and glue the fifth. It is quite important that there be no air leaks, so you must make sure a fillet of glue shows along all the glued joints.

You are now ready to glue and reattach the second side, but first be sure to remove any high spots to ensure a tight fit. Again, apply glue to the edges and nail the side as before, wiping off any excess glue before it sets. Reattach the top and bottom while the glue sets, using wax paper to prevent adhesion.

When the glue has dried, you cut the 4 9/16″-diameter hole for the bass driver. I used a 4½″ hole saw and enlarged the hole to size using a powered rotary file. You need to fill the four corners with small tetrahedrons. In my case, these pieces eliminated a strange whistling sound in the midrange. You can make these pieces by cutting corners from a 2 × 4. Do not attach the top or bottom pieces now.

Before you install the sound-absorbent material, you should add a fillet of glue at all joints on the second side. A ¼″ dowel with a few layers of cloth over one end will enable you to reach these joints. You must also make certain the top and bottom will fit tightly. A bench sander is ideal for this, but I used a file and sanding block.

This is a good time to drill a hole for the speaker wire, which should pass through the top of the back and through the corner block, very close to the top board. You should try to keep the wire in contact with the top panel as it goes to the driver. I sealed the gap between the speaker wire and the wall of the hole from the outside with modeling clay.

BASS-ENCLOSURE AND STUB ABSORBERS

At this point the enclosure is complete except for the top and bottom, which you will attach with screws and temporarily seal with weather stripping. You may wish to locate and drill the screw holes at this time. I drilled four countersunk holes so the screws would be anchored into the corner blocks for both the top and bottom pieces. However, you must install all the insulation before attaching these end pieces.

I cut all the acoustic-absorber pieces from the 1″ × 36″ × 36″ sheets sold by Radio Shack. See *Photo 2* for the latest arrangement for the absorbing material.

I made a full-size template out of stiff

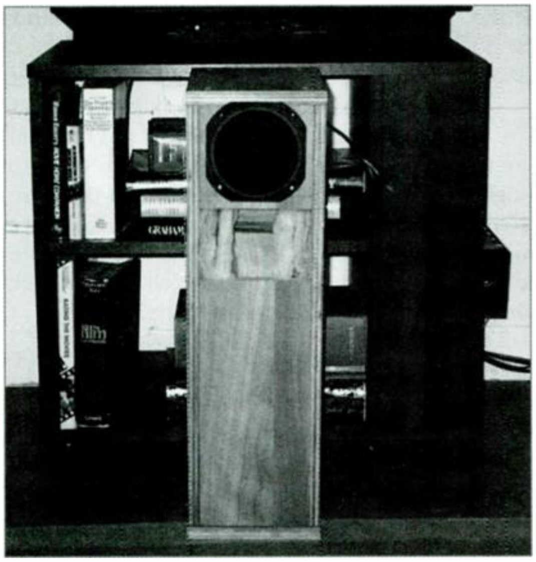

PHOTO 2: A third speaker using the newer fiberglass absorber arrangement. All three speakers have now been upgraded in this same way.

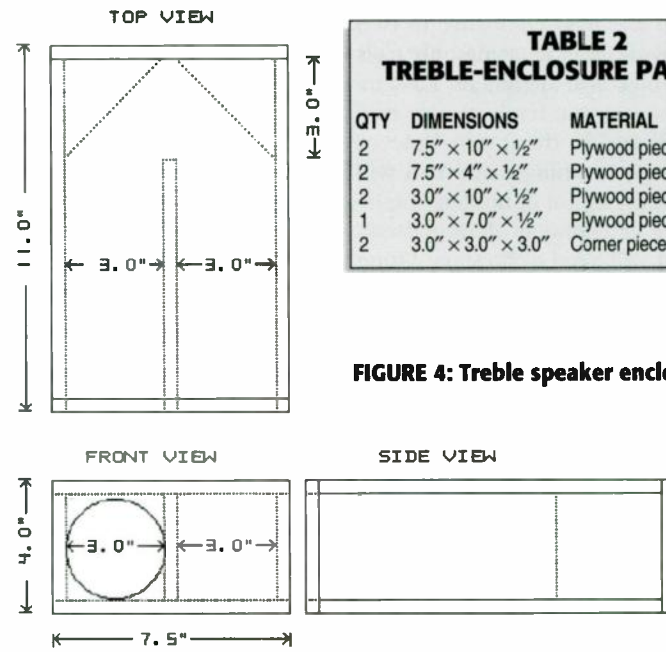

TABLE 2
TREBLE-ENCLOSURE PARTS LIST

QTY	DIMENSIONS	MATERIAL
2	7.5″ × 10″ × ½″	Plywood piece (top & bottom)
2	7.5″ × 4″ × ½″	Plywood piece (front & back)
2	3.0″ × 10″ × ½″	Plywood piece (sides)
1	3.0″ × 7.0″ × ½″	Plywood piece (center partition)
2	3.0″ × 3.0″ × 3.0″	Corner piece (45° hypotenuse)

FIGURE 4: Treble speaker enclosure.

cardboard according to the stub template of *Fig. 3*. Using this, I cut two pieces of absorber, but with the long leg longer than needed. In the case of the stub, you insert the absorber from the top. I threaded a wire through the stub passage and hooked it into the extra fiberglass so I could pull it through the passage. When it was in position, I worked the piece to one sidewall of the duct using a thin stick. I repeated the process with the other absorber, pressing it against the other wall so as to leave a 4″ gap in the center. I then cut off the excess absorber sticking out of the enclosure bottom.

HORN ABSORBERS

With both horn segments, I filled 66% of the duct volume with the fiberglass. The down segment includes the 90° turn near the driver and the duct from the back of the cone to the midpoint of the 180° turn at the bottom of the enclosure.

Again, I made a full-size cardboard template according to the down-horn drawing of *Fig. 3*, and then a 2″-thick laminate from two sheets of fiberglass using spray adhesive to join them. Using the template, I cut from the laminated piece a pair of 2″-thick absorbers for the down-horn segment.

Since I was unable to insert the pieces either from the top or the bottom, I cut the absorber along the dotted line in *Fig. 3* and inserted the long leg from the bottom and the short leg from the top. I used two long thin sticks to help insert the fiberglass from the bottom. Then I used a single stick to move the fiberglass to the side. With both pieces in place, there was a 2″ gap in the center. I inserted the end pieces from the top using a light coat of spray adhesive to rejoin the cut ends.

For the up-horn segment—which includes half of the 180° turn at the bottom of the enclosure plus the remainder of the duct to the port on the front panel—I made a full-size cardboard template according to *Fig. 3*. I then repeated the laminating and cutting pro-

TABLE 3
ELECTRICAL PARTS LIST

PART	VALUE	SOURCE	DESCRIPTION
C1, C2	8.0µF	Parts Express	Metalized Mylar®
C3, C4	8.2µF	Parts Express	Metalized polypropylene
R1	8.0Ω	Radio Shack	20W noninductive
L1	1.0mH	Parts Express	0.21Ω air-core
LP	8.0Ω	Parts Express	8Ω LPAD
SPKR1	8.0Ω	Parts Express	Audax HM130CO or HM130XO, 5¼″ bas midrange
SPKR2	8.0Ω	Radio Shack	40-1325A, 8Ω auto, 3½″ dual cone
BS1, BS2		Radio Shack	Chassis mount feed-through barrier strips
TP1–4		Radio Shack	Five-lug tie points

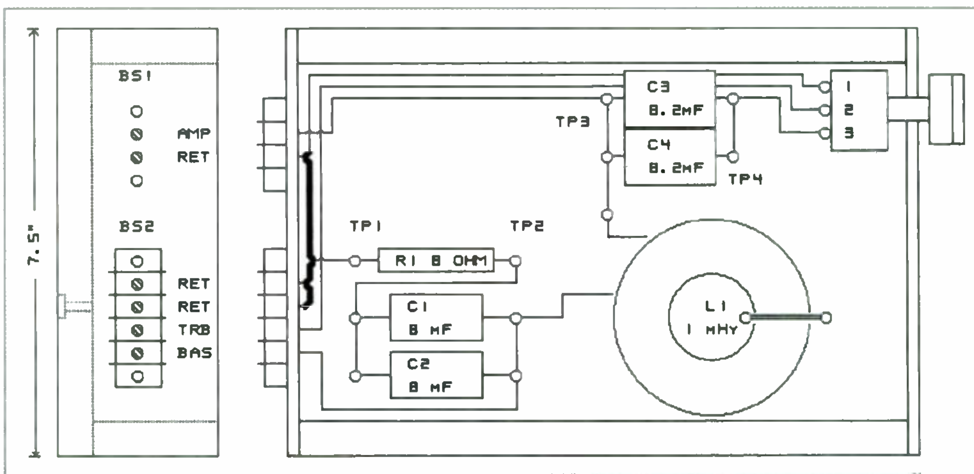

FIGURE 5: Speaker stand with crossover layout.

cedure I used with the down-horn absorbers, inserting the up-horn absorbers from the port, again using the dowels as an aid. Once in place, the pieces left a 2″ gap in the center.

At this point you can temporarily fasten the already fitted top and bottom. I used closed-cell weather-stripping foam for the seal, covering all edges with tape, and being especially careful to seal the internal path between the stub and the horn passage. After applying the weather stripping and making sure it would be compressed, I screwed the top and bottom in place.

FINISHING THE BASS ENCLOSURE

Since I am not a skilled wood worker, I could not use a natural finish on the cabinet because of blemishes on the first enclosure. Instead, I used an off-white latex paint that was quite satisfactory. Be careful to apply the paint uniformly around the speaker hole if you wish to mount the speaker without a gasket. I passed #18 speaker wire through the hole at the top, and soldered the wires to the speaker terminals, noting the polarity. I then fastened the driver with ¾″-long sheet-metal screws, with the terminals facing up. Allow enough wire to reach the network terminal board. You will make the connection to the terminal block later.

To support the grille cloth, I used a 7½″ × 12″ piece of 1/16″ hardboard with a cutout for the speaker frame and a 4″ × 6″ cutout for the port. I secured an oversized sheet of grille cloth to the hardboard with spray adhesive, and trimmed the edge of the cloth when the adhesive had hardened. You may wish to brush some black paint on the edge of the board to improve its appearance. I was careful in cutting the driver hole to ensure a snug fit so the assembly would need no hardware to hold it.

THE TREBLE ENCLOSURE

I built the treble box in the same manner as the bass enclosure, but with ½″ plywood. The details are shown in *Fig. 4*, and the parts list in *Table 2*. The materials list is given in *Fig. 6*. I completed the enclosure except for the back, which I installed after inserting the absorber. You should drill four screw holes for mounting the back at this time, making sure it will be a tight fit. The screws should bite into the two corner pieces.

I cut six 3″ × 11″ pieces of fiberglass and bonded them in two groups of three with a light coat of spray adhesive. Using the dowels, I stuffed the laminated pieces in from the back and trimmed the ends until they met when bent around the center partition. I drilled the hole for the speaker wire through the back and corner block close to the side wall. I did not bother to seal the back with weather stripping, but you may wish to do so. It should not affect the performance much either way.

I finished the treble box with latex paint to match the bass enclosure. The treble unit is not attached to the bass box, but only rests on it, which allows you to position the treble unit for the best time adjustment with the bass. The cable runs down the back to the network terminal block, and you can connect it when the third box is complete.

I used a 7½″ × 4″ piece of 1/16″ hardboard to support the grille cloth, and made a cutout for the speaker frame so the hardboard would lie flat against the front panel. I installed the grille cloth in the same way as with the bass enclosure. In the latest version, however, I replaced the grille cloth with black window screening.

PHOTO 3: Grille-cloth changes to the speakers, which have been raised slightly.

PHOTO 4: Absorbing columns included with system setup.

CROSSOVER

The stand and network layout are shown in *Fig. 5*, and the crossover schematic in *Fig. 6*. I used ¾″-thick scraps for these pieces, and made the ends of 1⁄16″ hardboard. The terminal blocks I used are made to solder to printed-circuit-board tracks. I made the three ground connections to a copper sheet, which I drilled to accept the return pins, and I made the other three connections by looping the wire before soldering it to the terminal-block pins.

I chose 1.25kHz as the crossover frequency, and included a Zobel to prevent the rise in speaker impedance from affecting the high-frequency rolloff. However, there also is a step in the response due to diffraction at a frequency that depends on the cabinet width. Diffraction data given in Olson's book[2] puts the half step of the first diffraction peak at 538Hz for a 7.5″ cube. For an isolated cubic speaker, this may be as high as 10dB.

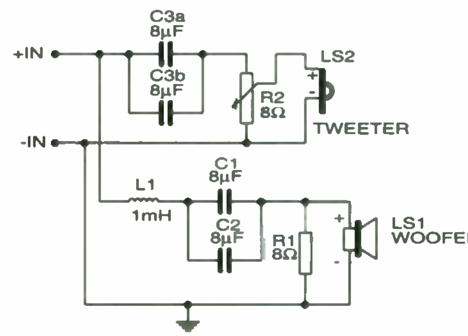

FIGURE 6: Crossover schematic.

After trying to incorporate a correction in the network, I finally decided to rely on the amplifier's tone control, since my receiver's owner's manual showed an inverse characteristic close to that required. Also, I was using absorbent screens behind the speakers to minimize the higher-order effects of diffraction and the effects of wall reflections (*Photo 1*). Data reported by Beranek[3] shows a 6dB diffraction step for a 2.5″ spherical microphone, which, when scaled in size, agrees roughly with the above.

The parts for the network are listed in *Table 3*, and the layout for the network is shown in the speaker-stand drawing (*Fig. 5*). The network that I built used high-quality components that I wired with #18 wire, using the specified tie-points or parts thereof. I spray-painted the outside of the stand semi-gloss black.

I used double-sided adhesive tape to fasten the bass enclosure to the stand. When this is done, you are ready to connect the speaker wires to the terminal block, and a heavier speaker wire from the amplifier outputs to the terminal block. Subsequent adjustments to my setup are shown in *Photos 2, 3,* and *4*.

RECEIVER PROBLEMS

My receivers had startling problems with the output circuit. Both of my AM/FM receivers had provisions for switching between two speakers, but the wiring from the motherboard to the A and B selector switches on the front of the receiver and from the switches to the connectors on the back were done with wire-wrap! Moreover, several of the wraps were loose! I soldered all the wraps and connected the speakers to both the A and B outputs, so when both the A and B buttons were pressed, the wire-wrap circuits were paralleled. Why connect the speakers with monster cable when the audio-output connections use wire-wrap?

OPERATION

Unfortunately, the sound you get depends very much on your listening room and speaker placement. These speakers should work well placed close to one wall. Using them in a 13′ × 26′ room, I have tried placing them against the long wall and also the short wall. The sound is better on the short wall. In both cases I avoided placing the speakers directly in the corners. The high-frequency coverage is far better this way, but there is more variation in sound level between a close-up position and one at the end of the room. The best listening position is close to the center.

FINAL ASSEMBLY

When you are finished with the listening tests and wish to button up the bass units, remove the drivers and the end pieces. Then move the fiberglass out of position so you can work a light coat of carpenter's glue against the sides. Then reposition the fiberglass, glue the top and bottom in place, and reattach the speaker after the glue has set. You may wish to insert fill over the screws and refinish the top.

REFERENCES

1. David B. Weems, *Great Sound Stereo Speaker Manual*, Tab Books, McGraw-Hill, 1990, p. 124.

2. Harry F. Olson, *Elements of Acoustical Engineering*, Second Edition, D. Van Nostrand Company, 1949, p. 20; available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-6371, FAX 603-924-9467, E-mail custserv@audioXpress.com.

3. Leo L. Beranek, *Acoustic Measurements*, First Edition, John Wiley & Sons, 1949, p. 182; available from Old Colony Sound Lab.

Kit Review

SWANS M1

By Dennis Colin

Swans M1 Kit. Available from Acoustic Technology International, 15 West Pearce St., Unit 2 & 3, Richmond Hill, Ontario, L4B 1H6 CANADA, (905) 889-7876, FAX (905) 889-3653, www.dulcet.com. 310H × 180W × 250D mm. 12 lbs without cabinets and 47 lbs with cabinets. Cost $410 (delivered).

The Swans M1 consists of a two-way bass-reflex minimonitor with 5″ paper/Kevlar® cone woofer and planar isodynamic (ribbon) tweeter. The crossover is second-order Linkwitz-Riley electrically, but the acoustic response seems to be fourth order. The cabinets are 12″ × 7″ × 10″ and the estimated internal volume is 7.3 ltr.

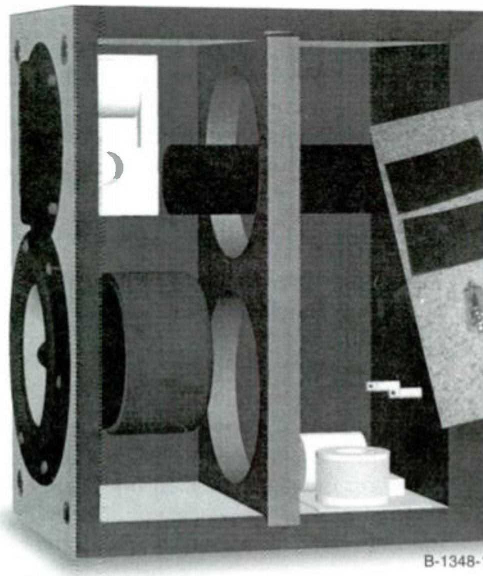

FIGURE 1: Pictorial supplied with the instructions.

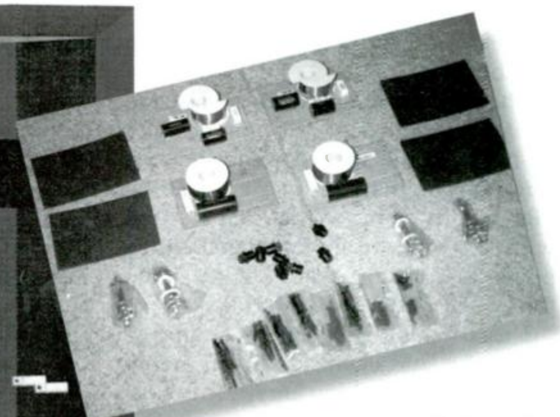

PHOTO 1: Crossovers, terminals, and hardware.

KIT DESCRIPTION

This is a beautiful kit. The cabinets, with impeccable cherrywood rounded sides are as much a work of architecture (*Fig. 1*) as they are a delight to behold. And with ¾″ MDF plus bracing in an enclosure of only 12″ × 7″ × 10″, I think they'll vibrate about as much as a tree trunk!

Continuing with the parts, the crossovers (*Photo 1*) are classic quality—heavy-gauge air-core inductors, tweeter film caps and Bennic woofer cap, and sand-cast wirewound resistors. The manufacturer recently upgraded the woofer crossover (CO) supplied with this kit and provided it free to customers with the older CO. How's that for service?

One caution: this evaluation kit came with the tweeter output polarity marked inverted on the CO board, but the schematic shows the tweeter noninverted (negative ground), and the preinstalled CO (with this kit) has a red dot over the (–) output marking, with a red wire soldered thereto. Just to be sure, I will listen for phase cancellation with pink noise to avoid what Hilary Paprocki (Audax A652 Review, *SB* 4/98) went through because of a manufacturer's negligence. I won't measure the system response to verify this (it could prejudice my review); more on this later.

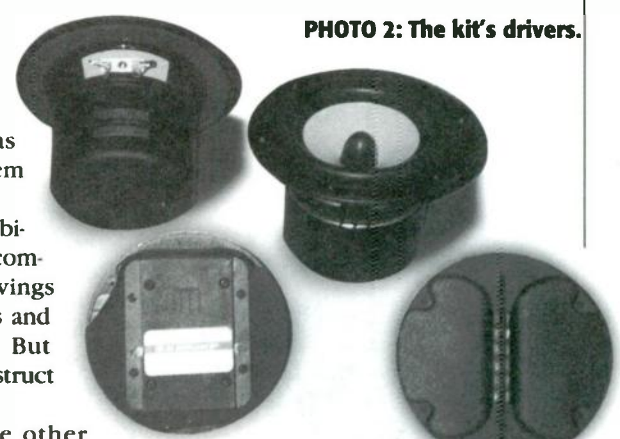

PHOTO 2: The kit's drivers.

I'm saving the drivers (*Photo 2*) for dessert—let's just say that if they sound as good as they look, this system will be something else plus!

Since I didn't build these cabinets, I didn't need the very complete three-dimensional drawings supplied with the instructions and that nice pictorial (*Fig. 1*). But they're all you'll need to construct them.

Photos 3 and *4* show the other parts. Worth special mention are the terminal posts (*Photo 4*): I've seen smaller car-battery terminals! The posts are 5⁄16″ in diameter, accepting 3⁄16″ wire, and the knobs are ¾″ solid metal. You could easily jump-start a large car engine through these connectors!

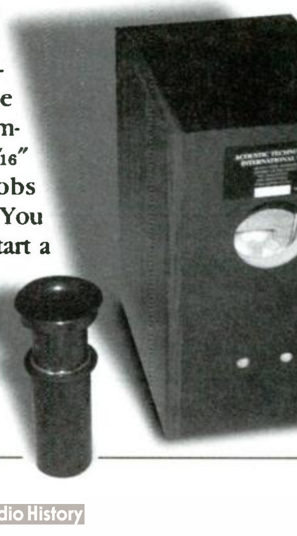

PHOTO 3: Cabinet rear and port tube.

PHOTO 4: Heavy-duty terminals and cabinet pictorial.

ASSEMBLY

While the tweeter COs were already installed in these cabinets (with screws and a damping pad) (*Photo 5*), I siliconed the woofer CO in place (*Photo 6*). I soldered wires to the COs before installing, having learned this the hard way with my Audax A651 review (*SB* 6/97).

Next, I installed the terminals. It was necessary to make a small notch on the sides of the cabinet holes for a "keyway" to hold the outer terminal washers' alignment tabs. Since I had no suitable tool to tighten the nuts inside the tight cabinet space, I dripped some nuclear-grade glue ("Zap-A-Gap CA+" by Pacer) on and around the aforesaid nuts. Great stuff—just don't get any on your fingers and touch anything you don't want a long-term relationship with!

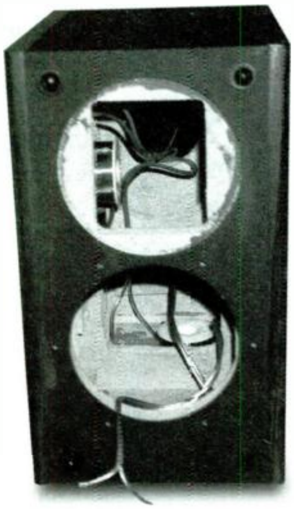

PHOTO 5: Crossovers installed.

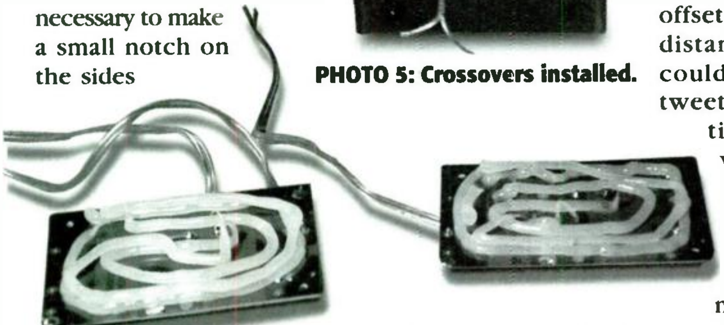

PHOTO 6: Woofer crossover with silicone.

On one cabinet (S/N 0025125) the tweeter faceplate cutout was slightly too small; not having carpenter's tools, I used a portable drill with a grinding wheel until the tweeter fit. *Photo 7* shows damping (Acousta-Stuff, I think) in the bottom of the cabinet (front and rear of bracing wall). I used rubber cement to keep it from creeping near the port opening. *Photo 8* shows completed loudspeakers.

PHOTO 7: Cabinets ready with one damping.

PHOTO 8: Completed speakers.

POLARITY NUISANCE

Taking a lesson from Mr. Paprocki, I decided to be certain that the drivers' polarity was correct. First I clip-leaded the drivers to the CO outputs while driving the input with pink noise (*Photo 9*). By holding the woofer and tweeter facing me and offsetting their relative distances from me, I could hear, with one tweeter polarity, a distinct cancellation with aligned driver distances; with opposite relative polarity, I heard a nice, solid pink noise sound.

The strange thing was that one tweeter (S/N 00131) sounded correct following the polarity markings, while the other (S/N 00133) sounded equally correct with reversed-from-markings polarity! The same thing was true with both crossover pairs. I used the red dot on the tweeter CO as positive.

Also, I checked the woofers' polarity with a 1.5V battery; both units were correct. Then I drove both tweeters in parallel directly from pink noise, following the polarity markings. Sound? Out of phase!

Since you cannot see the deflection of a ribbon tweeter with DC, and I don't have access to a laser interferometer, I used an acoustic pulse test (*Photos 10*, *11*, and *Fig. 2*). The narrow (20µs) pulse fed to the tweeters is short enough, and the repetition interval (500µs) long enough, to establish absolute acoustic polarity. *Figure 3* shows three things: first, the two tweeters are of opposite polarity, second, S/N 00131 is the correctly marked one (the first half-cycle of response is positive polarity, same as electrical input), and third, this is the tweeter that sounds correct when following the polarity markings. Very satisfying—I wasn't hearing things. Case closed!

PHOTO 9: Tweeter polarity test.

BREAK-IN

The manufacturer recommends a 72–hour break-in. Not wanting to wait this long, nor convinced of the need, I used pink noise for about two hours to give the M1s some exercise before I listened to music. I fed pink noise through an old Pioneer receiver amp, with slightly boosted bass and reduced treble, so as to exercise the M1 woofer without endangering the tweeter.

Figure 4 shows the spectrum, while *Fig. 5* shows a 1/10-second noise waveform sample at the speaker terminals. Note that the power level is only an average of 1W (re 8Ω). But being noise, the voltage reaches at least a 10V peak (12.5W re 8Ω). Because of the low-frequency spectral emphasis, this was enough to cause a woofer excursion of about ¼″ pk-pk, which should be enough exercise.

LIVE VERSUS REPRODUCED TESTS

Many reviewers say things like "The flügelhorn in *Danse du Fromage* sounded excellent!" The problem is that the recording engineers probably added some fromage (cheese) with their equalizers (makes them feel like part of the artistry!). Now, if your speaker has an equal but opposite anomaly in its response, the flügelhorn will sound natural, but on better speakers it will sound like, well, cheese!

So it's necessary to use a wide variety of recordings that you know sound good on many (good) speakers. But I think a simple live test like a miked voice is needed to completely rule out colorations in recordings at which you were not present.

My wife and I listened to one another's voices, in each case alternating be-

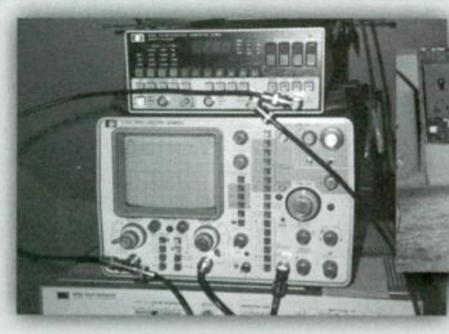

PHOTO 10: Pulse generator used in polarity test.

PHOTO 11: Pink-noise phasing test.

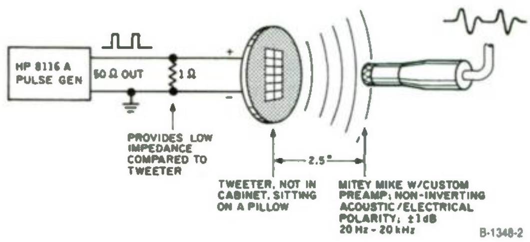

FIGURE 2: Acoustic pulse test.

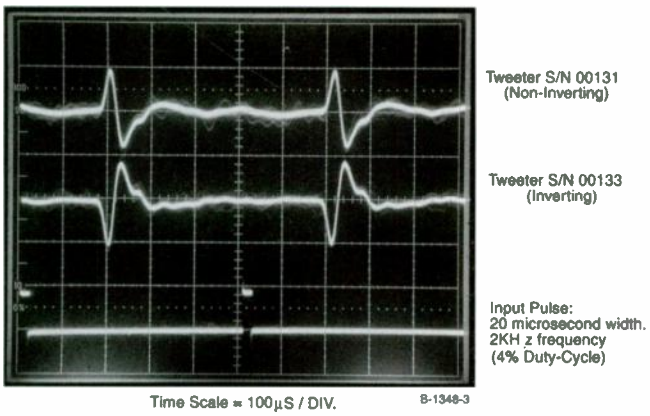

FIGURE 3: Tweeter polarity results.

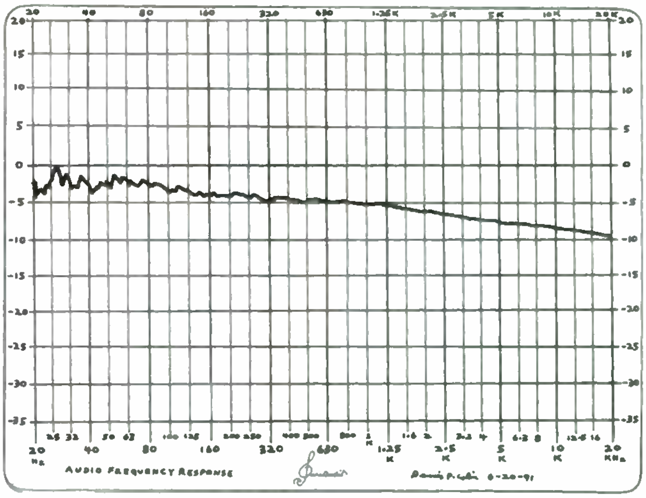

FIGURE 4: Spectrum of shaped pink noise used for breaking in.

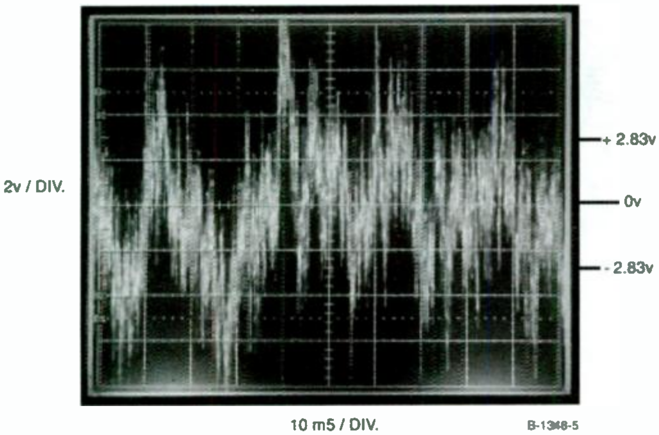

FIGURE 5: A 1/10 second noise waveform sample at the speaker terminals.

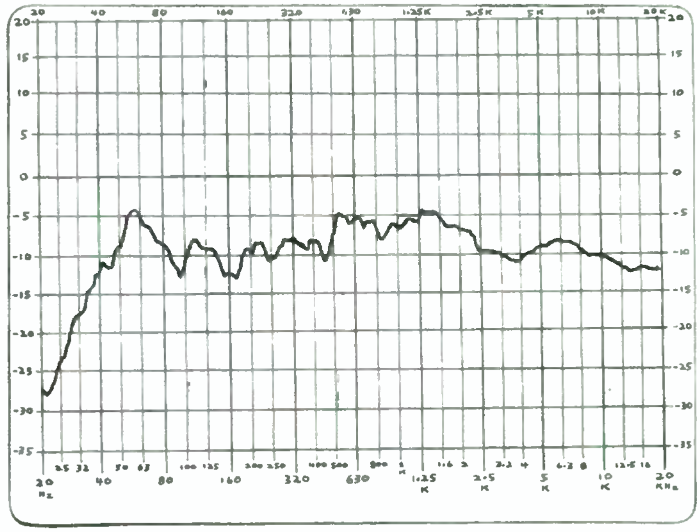

FIGURE 6: Frequency response in 3000ft³ listening room.

tween speaking directly from a position next to the M1, and then into a Mitey Mike capsule driving the M1.

The mike was far enough from the loudspeaker so that acoustic gain was well below feedback, and the listener heard much more reproduced voice than direct. My wife simply described my reproduced voice as "natural." That was good enough for me; she is an experienced music-lover, and at least as auditorially perceptive as I am.

When I listened to her voice alternating between direct and reproduced, I thought the reproduction was so natural that I couldn't be sure what the difference was, if any. Very good, indeed!

One more live source I tried was two spoons clinking together. I've found very few loudspeakers that could reproduce this sharp, distinctive, and familiar sound accurately. But this M1 amazed me. I heard *no* transient blurring or coloration, even though I expected to hear the delay dispersion of the (estimated fourth-order acoustic) CO. With all other second- or higher-order CO systems I've tried this on, including the Audax A651 and the A/D/S 300C, I did hear a slight transient smearing. But on the M1, it just sounded as though there were clinking spoons coming from the speaker, with pinpoint solidness of image, tonality, and transient reproduction.

Finally, I listened to pink noise, and was surprised at the solid coherence, even up close. Even well off vertical axis, there was no crossover "phasiness"; apparently the tweeter's controlled vertical dispersion smoothly rolled off the highs before they could cause delay interference with the woofer. Also, the pink noise sounded very smooth, with no noticeable coloration. "Hey!" I hear you saying, "I don't care about PA system use, spoons, or noise--how are these things with music?"

LISTENING TESTS

While they were still connected to the old receiver used for noise break-in, I placed the two speakers on top of it, about 3″ apart (*Photo 12*), sat right in front of them the way minimonitors are sometimes used in studios, and switched the receiver to FM.

Listening to an FM station playing a Mozart opera, I heard the full range of strings sounding so real, so well differentiated, and with such natural spatial depth, that it was easy to imagine being eight years old again and hearing my father playing the violin at the concerts I attended then!

One other thing: even from 1′, I could not hear any separateness of woofer/tweeter radiation. Within about ±15° (vertical) and ±45° (horizontal) of axis, the sound was coherent, focused, and so nice!

My equipment includes a Nakamichi AV-1 receiver (100W/ch), Yamaha CDC 755 CD changer, Yamaha KX-W592 cassette player, and Miracord turntable with Shure R700E cartridge. For the live-voice test, I used the Mitey Mike II capsule and the Panasonic WM-60AY.

My listening room is approximately 20′ × 18′ × 8½′ (about 3000ft³), moderately damped with stuffed chairs, a couch, 10′ front-wall drapes, and full carpet. The rear and side walls are reflective, but dispersed by a stairwell and numerous doorways. The wood floor is very rigid. The acoustics are very good, and the sound is satisfying with all good speakers I've tried. The M1s, placed on 25″ stands, were about 3′ from the front wall and 5′ from the side walls, and the speakers were about 10′ apart .

PHOTO 14: Close-up monitoring.

SOURCE MATERIALS

LPs

"The Blue Danube," Ormandy, Philadelphia Symphony Orchestra, Columbia MS6217

Mozart Symphony #36, Istvan Kertesz, Vienna Philharmonic Orchestra, Super Analogue Disc, KIJC 9128

Blues, Ballads, and Jumpin' Jazz, Lonnie Johnson and Elmer Snowden, Analogue Product APR 3001

CDs

Ocean Front Property, George Strait, MCA MCAD-5913

Natural High, The Commodores, Motown MCD 08014 MD

Rhapsody in Blue, George Gershwin, Mercury 434341-2

Tango, Julio Iglesias, Columbia/Sony CK67899

Marches in Hi-Fi, Arthur Fiedler, Boston Pops, RCA 09026-6149-2

Two Fisted Mama, Katie Webster, Alligator ALCD47777

Dedicated to the One I Love, Linda Ronstadt, Elektra 61916-2

Concerto Under the Stars, Madacy CT-2-2979

Cassettes

Blue, LeAnn Rimes, Curb D4-77821

Do You Know What It Means to Miss New Orleans, recorded from DSS satellite

ABOUT THAT MISMARKED TWEETER

My wife, whose hearing probably extends beyond my 15kHz limit, noticed a slight roughness from one tweeter. Not knowing which was which, she identified the wrong-polarity one as the source. (Note: having identified the reversed polarity, I corrected for it in the wiring.)

I listened closely and did hear some difference from the other unit. Possibly slight distortion, I thought. However, when listening in stereo, even up close and personal, whatever I had heard quickly disappeared. It was such a small, not frequency-colored effect that I didn't notice it affecting the overall reproduction of music.

Igor Levitsky, the designer, was very apologetic about this tweeter that got by QC, and is sending a replacement. However, since I wasn't aware of the difficult-to-notice sonic effect until my wife pointed it out, and even then didn't notice it affecting the music in normal listening, I went ahead with the review. And even my wife (who should be the reviewer) said "These are very good speakers," and simply enjoyed the music with me.

My impressions are as follows:

- Blue (LeAnn Rimes)—so much feeling that people first thought it was Patsy Cline (my wife and I did).

Recording quality? I have this on a lowly cassette, yet it sounded better and more real than most of my CDs. This shows you what the recording companies can do when they care.

This recording sounds good even on a telephone, but is so hauntingly captivating on good speakers that you start thinking Patsy is reincarnated! The M1s reproduced this song with the most natural sound I've heard. LeAnn's voice was as good as there, and so were the guitars, drums, and other rhythm instruments. The string bass was so natural I wasn't even aware of the 55Hz rolloff unless I listened for it.

I would say the fidelity was at least equal to the A/D/S 300C minimonitors in my car, but with one advantage: in a room, the A/D/S has a crossover-region deficiency and "phasiness" off the vertical axis. This also causes slight image smearing even on axis. But the M1s didn't do this. The image stayed focused and the sound was natural everywhere in the room. When exceeding about 60° horizontal or 15° vertical, I noticed only a smooth HF rolloff, similar to natural acoustic filtering.

Most amazing is that I could hear *no* coloration; it sounded as though any frequency-response anomalies were as small as the variations caused by moving about in normal situations. There was just no perception of tonal emphasis, deficiency, transient delay smearing, or image blurring.

- Next, I played the *Blue Danube* (must be something about that color!) from a vinyl LP recorded circa 1960 BC (before CDs). The pops, scratches, and even the non-$2000 cartridge didn't prevent the strings, horns, and everything else, from sounding so good I wanted to throw out all my CDs. The M1s were simply not in the way.

I don't know about the overall frequency balance, but the musical sensation was such that I just got up and pretended I knew how to waltz with my wife.

- "Sunny Side of the Street" (*Blues, Ballads, and Jumpin' Jazz*) is on an Analogue Product vinyl platter, that serves up the most natural, funky, snappy guitar-pickin' sound I've heard since CDs came along to "improve" things. Listen to guitar picking on a CD, then vinyl, then live—anyway, the M1s reproduced all the snap, rattle, and twang of those strings with invisible speed (no audible delay smearing) and natural tonality.

- "Can't Buy Your Way Out of the Blues" (*Ocean Front Property*). George Strait is as well-recorded as he is good. The voice, guitars, fiddle, piano, and drums sounded as real as you can expect from two-channel stereo. But the bass guitar really surprised me with its naturalness and depth, in spite of the 55Hz rolloff. There was no bass or other coloration of note.

- *Tango* by Julio Iglesias—a masterpiece. Julio wrote, "I was probably born because of the Tango!" And this dance's legendary passion is recorded par excellence—dramatic ocean of strings, guitars, drums, deep bass, and, of course, Julio's voice magically transmitting the soul of Argentina.

The M1s provided a most excellent reproduction of this recording. I only wished for the 10″ Focal I'm used to, with flat room response to 25Hz, and my home-theater surround.

- *Concerto Under the Stars* was excellent. The bells in "Fledermaus" were so solid in time and space that they were as clear as, well, you know! And "Clair de Lune," one of my all-time favorites, was simply ethereal. (Claude Debussy's piano, when reproduced uncolored, has a habit of sounding like that!)

- *Mozart Symphony #36*—if you highly value natural string tonality, you should highly like these speakers!

- "Do You Know What It Means to Miss New Orleans"—classic found on DSS satellite (if you can identify, please inform me). There's a very nice clarinet leading this piece. Now, my wife plays clarinet in this room, so let me say this about that recorded one on the M1s: There may be some difference from live, but when not thinking about it, I didn't notice any. Also, the acoustic bass, drums and cymbals didn't want for anything, except perhaps 1000-channel stereo.

THE ULTIMATE COMPARISON

Two months ago, I heard world-famous musicians in a string ensemble at the Lake Winnepesaukee Music Festival (Wolfeboro, NH). I sat about 20′ away. The sound was just glorious! I closed my eyes and tried to imagine this sound coming from loudspeakers; I couldn't!

But on hearing the M1s play similar music two months later, I thought their reproduced string tonality was as close to real as any speaker I've heard.

This is not an immediate live-vs-recorded test, depending on memory as it is, but I can still close my eyes and remember the live sound. As Ed Dell says, pleasure is a great stimulator of memory. And the only difference I can be sure of is the inability of two-channel stereo to re-create the entire 3-D sound field of a 100,000+ ft^3 hall—no fault of the speakers. But as for tonal realism, I think these M1s are as good as speakers get, regardless of price.

SUMMARY OBSERVATIONS

1. The tweeter sounds state-of-the-art to me, which is no surprise considering its well-designed ribbon construction. What most surprised me, though, is the top-to-bottom neutrality, literally free of audible coloration, even compared to a live voice! I guess that speaks well for tonal accuracy.

2. Regarding imaging, the M1s are uncannily solid—no crossover audibility, let alone smearing, even off vertical axis (just a very natural HF rolloff).

3. Horizontal dispersion is very wide, maintaining tonal neutrality and imaging. This, I believe, is responsible for the unusually wide, deep, and open spatial reproduction.

4. The transient response sounds free of any percussive delay-smearing effect.

5. Transparency and detail resolution sound extremely good, possibly as good as an electrostatic.

6. The bass quality is so natural and detailed that I hardly missed the below-55Hz stuff. (But since I love deep bass, I would add a good subwoofer.) With the M1s, I heard the things a live acoustic string bass produces: both the extended series of harmonics and the vibrating string "pulsiness"; the latter requires excellent phase coherence as well as a smooth frequency response.

7. With 86dB (1W/1m) sensitivity and 50W handling, they won't safely sustain above 103dB SPL. But to find criticism of the M1s' musical realism, you'll need one of those "high end" magazine reviewers who can "hear" the benefits of plutonium-enriched superconducting binding posts (those on the M1 are close).

COMMENTS ON MEASUREMENTS

Now for a study of the correlation between sound (as I hear it) and sonic parameters (as Joe D'Appolito reveals with inarguable accuracy). The following figure numbers are Joe's, except as noted.

Figure 4—Full-range response: Since I didn't notice the gently sloping response below 1.4kHz, and Joe referred to room dependence, I measured the M1 in my listening room with the mike on the couch where I sat (my *Fig. 6*). While there is an overall falling slope of 5dB from 1.4kHz down to 85Hz or so, room gain picks up the bass where Joe's *Fig. 4* starts to roll off, around 60Hz. You can easily tune out the gradual slopes in Joe's *Fig. 4*.

Figure 5—Cumulative spectral decay: Joe said, "Decay performance is quite good." My ears agree; I heard resonance-free sound.

Figure 6—Step response: This takes only about 0.2ms (2.7" of sound propagation) to acquire a near-ideal natural exponential decay—good enough to explain the flawless transient response I heard.

Figure 9—Horizontal polar response: In Joe's words, "This performance is excellent and far superior to the typical 25mm dome tweeter." It sure sounds so, with outstanding neutrality and image depth anywhere in the room.

Figure 10—Horizontal average: I support Joe's word, "excellent."

Figures 11, 12—Vertical polar: The ≥10° CO dip shows up only as a local 2dB dip in my room response.

Figures 13-18—Distortion: I didn't hear enough to notice at the 90dB and lower SPL I listened to.

Figure 19—Grilles: I didn't use them.

Figure 20—Pair matching: I agree with Joe: good enough not to notice.

Overall, I think Joe's measurements confirm what I heard—outstandingly natural tonality, imaging, and transient response. And I heard all this whether 1′ or 20′ away, on or off axis.—**D. C.**

TESTING THE SWANS M1

By Joe D'Appolito

I received a pair of the Swans M1 two-way small monitor loudspeakers for testing, most of which I conducted on the first unit (#0020018), along with some comparative tests on the second (#0025215).

Figure 1 is a plot of system impedance magnitude and phase. The plot displays the double peaked curve of a vented system. The impedance magnitude curve is relatively smooth and free of small glitches that would indicate cabinet panel resonances. The vented-box resonant frequency is at 54Hz, where a local impedance minimum of 9.39Ω occurs. Although impedance drops to 6.82Ω at 285Hz, the M1 easily qualifies as an 8Ω system. Impedance phase lies between +36° and −45° over the full audio range. This should be a relatively easy load for most amplifiers.

Normally an impedance peak occurs at or near the crossover frequency, where the woofer and tweeter crossover networks interact to form a parallel resonance. With the M1, however, a dip in the 2-3kHz region is followed by a slight rise. Above 4kHz, impedance falls steadily. This behavior is somewhat unusual. Perhaps some form of impedance compensation is used here.

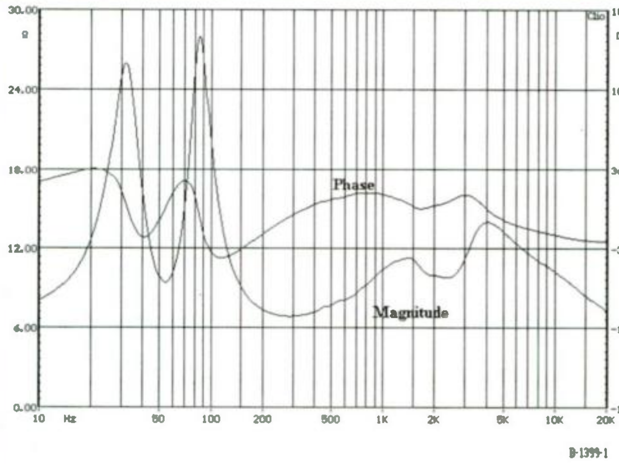

FIGURE 1: M1 impedance magnitude and phase.

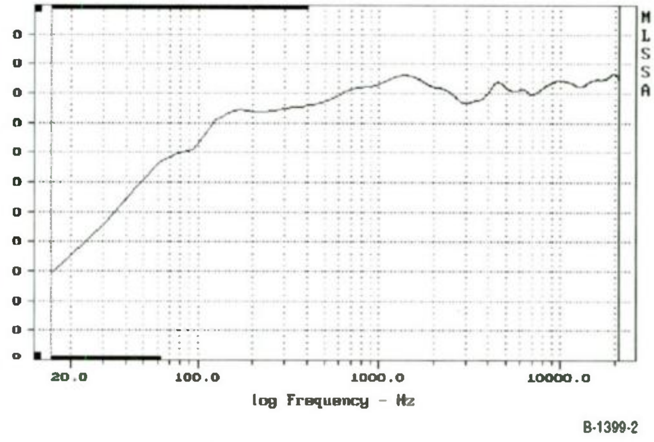

FIGURE 2: 1m quasi-anechoic frequency response.

FREQUENCY RESPONSE

Figure 2 shows the M1's on-axis quasi-anechoic frequency response at 1m. This response, representing the first 5ms of the measured impulse response, is valid down

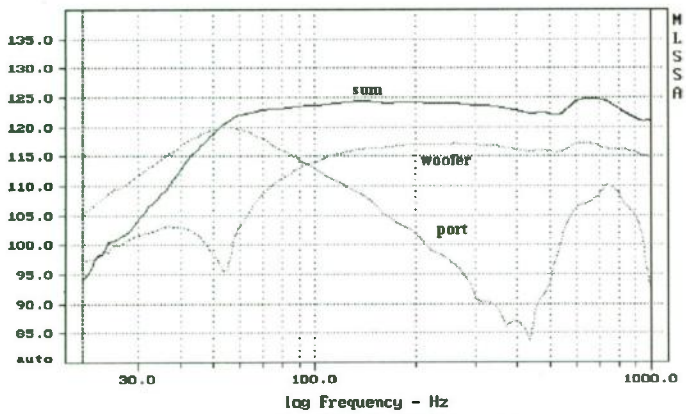

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FIGURE 3: Near-field woofer and port responses and their sum.

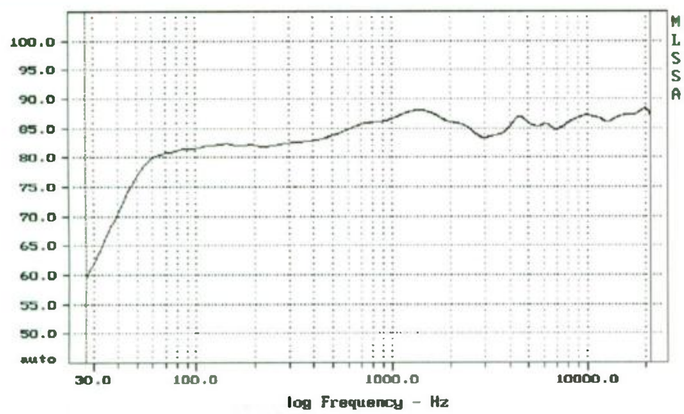

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FIGURE 4: Full-range response normalized to 1m.

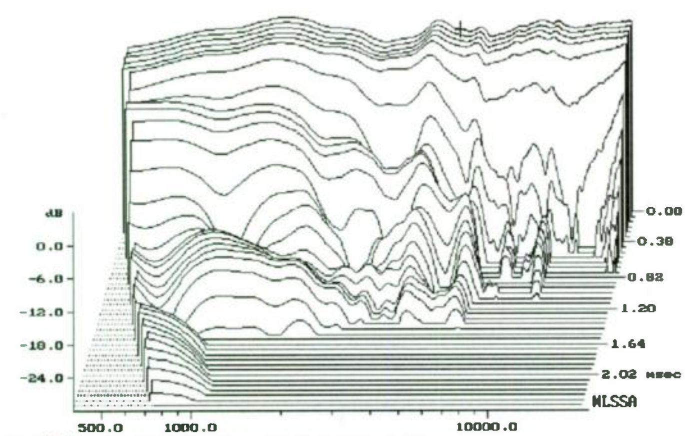

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FIGURE 5: System cumulative spectral decay.

to 200Hz. The response peaks 88dB at 1.4kHz. Between 1.4k and 300Hz response falls by 6dB, before leveling off in the 200–300Hz region. This is typical of uncompensated diffraction or spreading loss common in small systems.[1] Between 1 and 20kHz, system sensitivity averages 86.5dB. Variation about the average is +2, –3.5dB.

Near-field woofer and port responses are shown in *Fig. 3*. The dip in woofer response at 54Hz is another indication of the vented-box tuning frequency. Port acoustic output is also at maximum just above this frequency. There is a peak in the port near-field response at 750Hz due either to high-frequency leakage from the backside of the woofer or an "organ pipe"

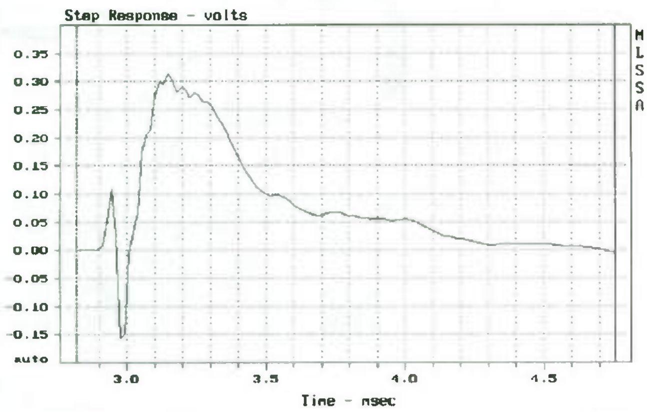

FIGURE 6: System step response.

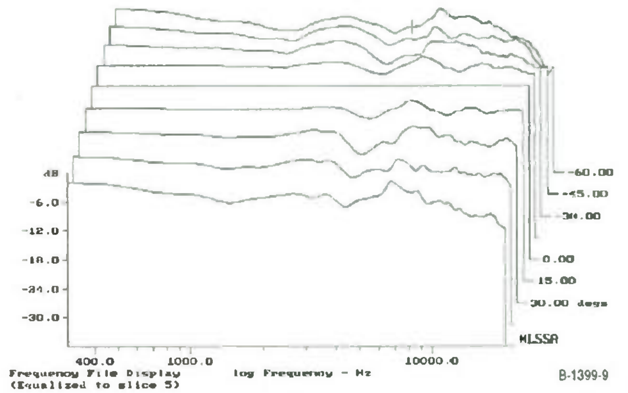

FIGURE 9: Horizontal polar response.

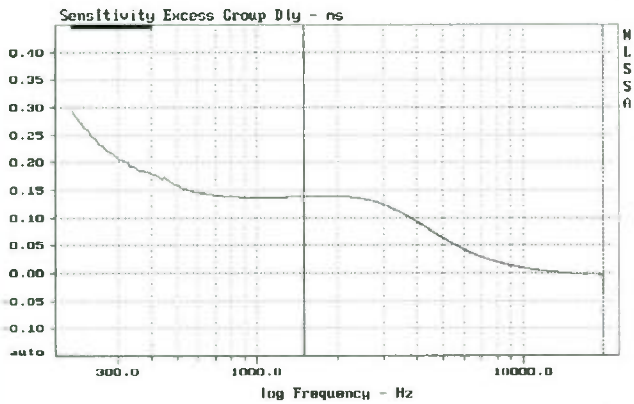

FIGURE 7: Excess group delay.

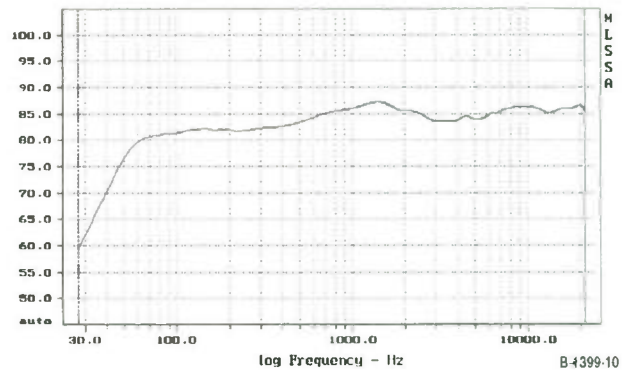

FIGURE 10: Horizontal response averaged over ±30°.

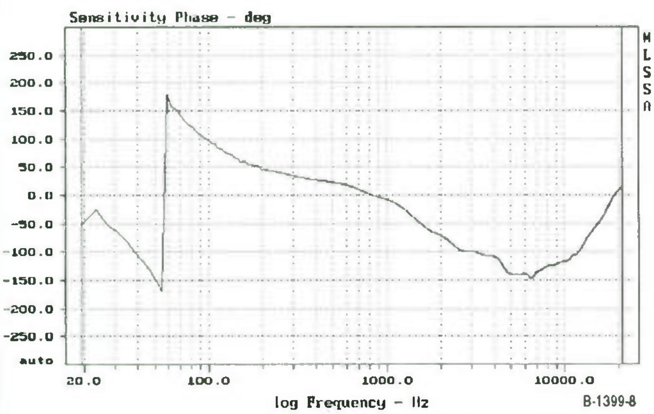

FIGURE 8: Phase response referenced to the tweeter.

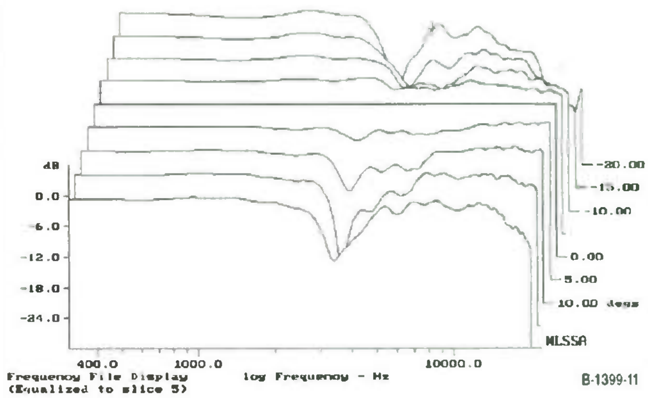

FIGURE 11: Vertical polar response (–20° to +20°).

resonance of the port tube.

The port and woofer near-field responses are summed by the MLSSA system, giving proper weighting to phase and the difference in areas of the woofer and the port, to obtain the complete near-field system response. This is also plotted in *Fig. 3*. The peak in port output at 750Hz causes a bump in the summed near-field responses. Because the port exits on the rear of the enclosure, however, this bump is not seen in the on-axis response of *Fig. 2*.

The system near-field response (*Fig. 3*) is spliced to the quasi-anechoic response (*Fig. 2*) at 203Hz to get the full-range on-axis anechoic response without the use of an anechoic chamber. This result is shown in *Fig. 4*. Overall system response is flat within ±2.7dB from 50Hz to 20kHz. The low-frequency –3dB point relative to 300Hz is at 45Hz. The falling response below 1.4kHz might give less weight to bass response and male voices. The extent to which the system sounds bass-shy will depend on the room and room placement.

The cumulative spectral decay (CSD) response for the Swans M1 is presented in *Fig. 5*. This waterfall plot shows the frequency content of the system response following a sharp impulsive input at time zero. On the CSD plot, frequency increases from left to right and time moves forward from the rear. Each slice represents a 0.11ms increment of time. The total vertical scale covers a dynamic range of 33dB. Ideally the response should decay to zero instantaneously. However, real loudspeakers have stored energy that takes a finite amount of time to die away. Prominent ridges parallel to the time axis indicate the presence of strong system resonances.

The first time slice in *Fig. 5* (0.00ms) represents the system frequency response. Tweeter decay is very rapid, falling 30dB in under 1ms. Also, there are no strong ridges in the frequency region covered by the woofer. All in all, the decay performance is quite good.

Figure 6 is a plot of system step response. It is obtained by a numerical integration of the system impulse response. The ideal step response should be a single rapid rise, followed by a smooth decay through the 0.00 level. *Figure 6* shows two separate arrivals of acoustic energy. The initial sharper positive spike is the tweeter arrival. It is followed by the woofer arrival, peaking roughly 0.2ms later. The drivers are both connected with positive polarity, but the system is not time-coherent.

A better view of this behavior is seen in *Fig. 7*, which is a plot of excess group delay versus frequency referenced to the tweeter's acoustic-phase center. This is a plot of delay in milliseconds versus frequency (a detailed description of excess group delay appears in *Testing Loudspeakers*[2]). In a time-coherent system, this plot would be a flat line.

Above 10kHz, excess group delay is essentially zero, as it should be since it is referenced to the tweeter in this frequency range. The curve rises smoothly below 10kHz to a plateau of 0.14ms (140μs) between 600Hz and 2kHz. This plot shows that over its operating frequency range, the woofer is 140μs behind the tweeter.

The smooth rise between 10kHz and 2kHz is indicative of a low-order, low-Q crossover network. According to the manufacturer's data, the crossover is second-order Linkwitz-Riley. Below 500Hz, excess group delay begins to rise again. The 24dB/octave low-frequency rolloff of the vented system causes this rise in delay. The phase error due to the sharp low-frequency rollup extends out well beyond F_3. Excess group delay is a very accurate indicator of driver-time offset.

Figure 8 shows the system phase response referenced to the tweeter's acoustic phase center. At low frequencies, phase response is near +360° (0° on the plot), which is appropriate for a vented system. Phase response falls smoothly through 0° at 800Hz, reaching a maximum negative value of 150° before rising again to essentially 0° at 20kHz, as it

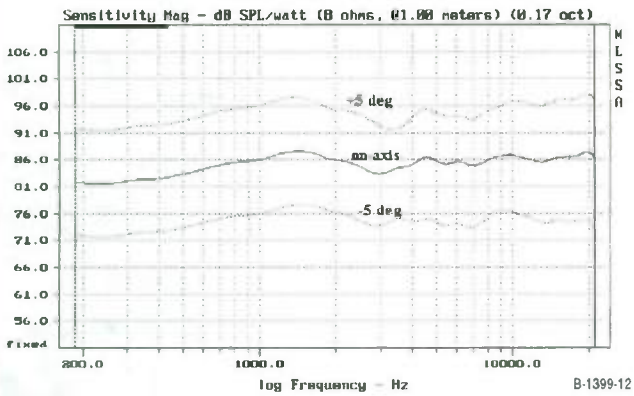

FIGURE 12: Vertical response at +5°, 0°, and −5° (offset 10dB).

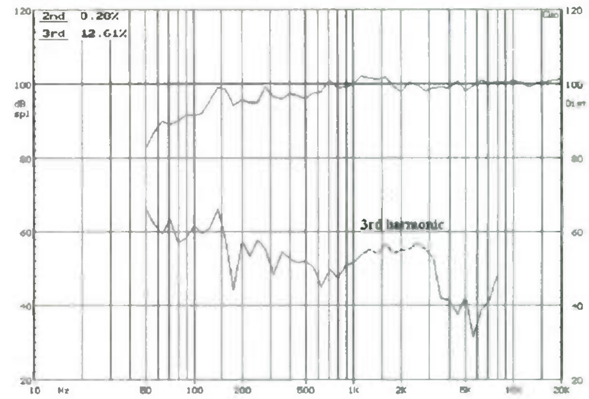

FIGURE 14: Third-harmonic distortion.

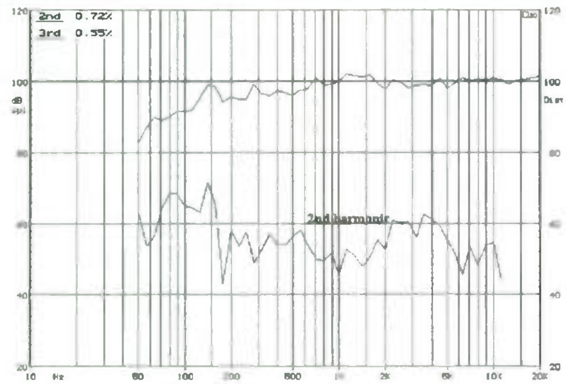

FIGURE 13: Second-harmonic distortion.

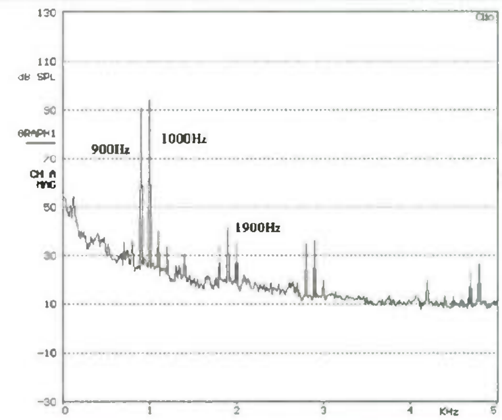

FIGURE 15: Woofer IM distortion.

should, because the phase response is referenced to the tweeter acoustic phase center.

POLAR RESPONSE

Polar response is examined in *Figs. 9, 10,* and *11. Figure 9* is a waterfall plot of horizontal polar response in 15° increments from 60° left to 60° right when facing the speaker. The microphone is placed at tweeter height for this series of curves. All off-axis plots are referenced to the on-axis response that appears as a straight line at 0.00°. For good stereo imaging, the off-axis curves should be smooth replicas of the on-axis response, with the allowable exception of the tweeter rolloff at higher frequencies and larger off-axis angles.

The expected rolloff of tweeter response is evident at higher frequencies and larger off-axis angles, but on the whole, the horizontal polar response is excellent. Tweeter response is down 4.8dB at 15kHz and ±45°. The corresponding figures at ±30° and ±15° are 3.5 and 1.1dB, respectively. This performance is excellent—far superior to that of the typical 25mm dome tweeter.

The average response over a 60° horizontal angle (±30°) in the forward direction is shown in *Fig. 10*. It is somewhat smoother than the on-axis response, because edge-diffraction effects decrease as you move off-axis horizontally. This is excellent horizontal power response, suggesting good direct-field coverage in the primary listening area with little if any change in timbre with position. Image stability should be very good.

Figure 11 is the waterfall plot of vertical polar response. Responses are shown in 5° increments from 20° below (–20°) the tweeter axis to 20° above it. Response at –5° is within 0.5dB of the on-axis response over the entire frequency range. As *Fig. 12* shows, response at –5° is actually smoother than the on-axis response. The worst-case dip at +5° is –2.7dB. At ±10° and beyond, there are deep response dips at the crossover frequency, ranging from –8 to –12dB. The manufacturer claims good vertical coverage within a 15° arc. I did not test at ±7.5°, but the data shown would suggest good coverage within at least a 10° arc.

HARMONIC DISTORTION

I ran harmonic-distortion tests at an average SPL of 90dB at 1m. This level may be somewhat high for a small monitor speaker like the M1, but it is consistent with all previous tests in this series and makes comparisons with early results easier. Ideally, harmonic-distortion tests should be run in an anechoic environment. In practice, it is important to minimize reflections at the microphone during these tests. Out-of-phase reflections can pro-

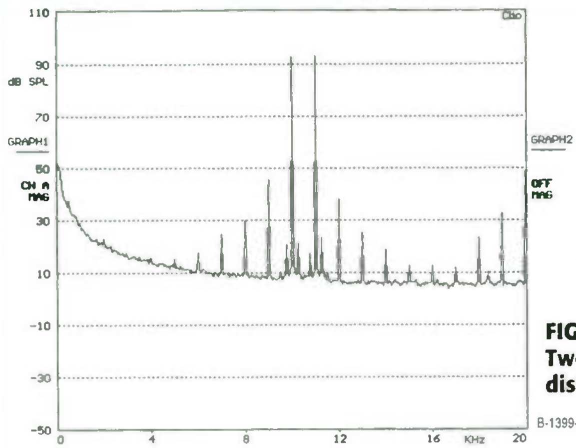

FIGURE 16: Tweeter IM distortion.

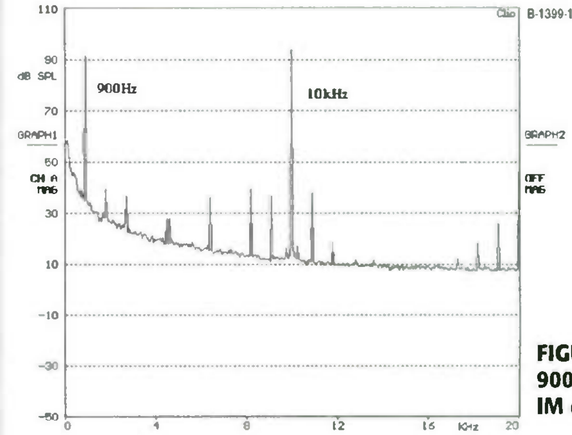

FIGURE 18: 900Hz/10kHz IM distortion.

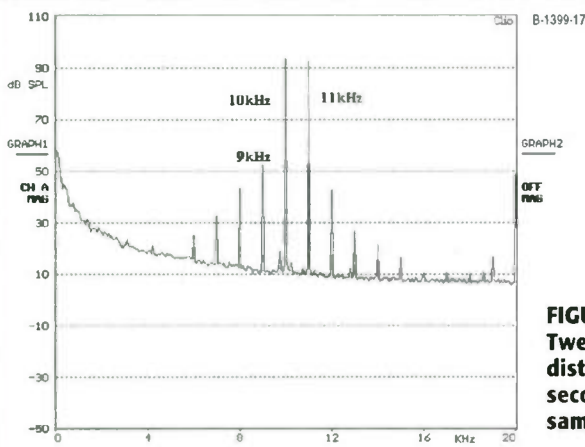

FIGURE 17: Tweeter IM distortion: second sample.

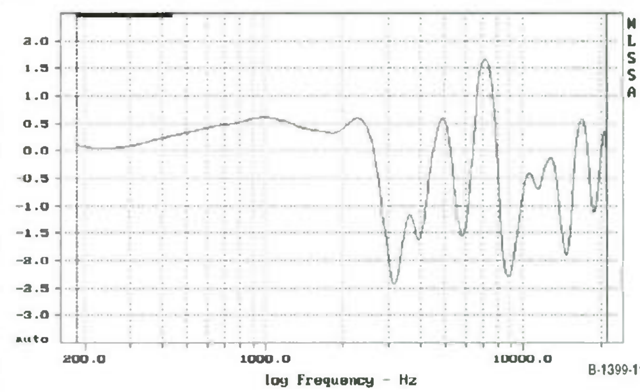

FIGURE 19: Effect of grille on system response.

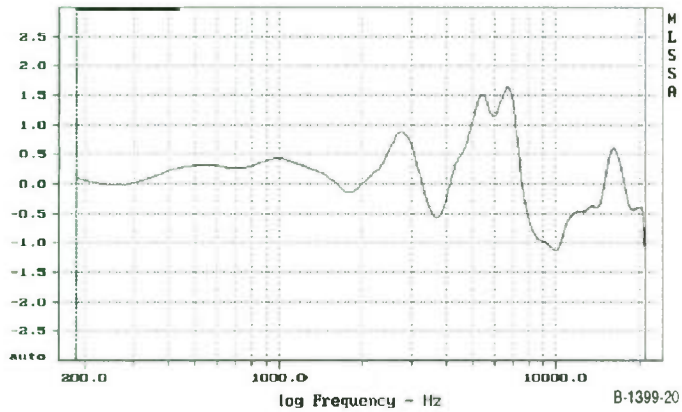

FIGURE 20: Pair matching.

duce false readings by reducing the level of the fundamental while boosting the amplitude of the harmonic. In order to reduce the impact of reflections, I placed the microphone at 0.5m from the loudspeaker.

Figures 13 and *14* show second- and third-harmonic distortion levels in dB SPL versus frequency plotted in ⅙-octave steps. System frequency response is also plotted on these figures. The second- and third-harmonic distortion levels at 50Hz are 8.3% and 12.6%, respectively. The corresponding numbers at 100Hz are 4.7% and 3.2%. These figures are rather high compared to other systems reported in these pages, but 90dB at these low frequencies is a rather high SPL for a 5″ woofer. Above 200Hz, all distortion levels are below 1%. Above 3kHz, where the tweeter is operational, third-harmonic distortion averages 0.1%, which is outstanding.

INTERMODULATION DISTORTION

Next, I measured intermodulation distortion. In this test, two nearby frequencies are input to the speaker. Intermodulation distortion creates output frequencies that are not harmonically related to the input. These frequencies are much more audible and annoying than harmonic distortion. Let the symbols f_1 and f_2 represent the two frequencies used in the test. Then a second-order nonlinearity will produce intermods at frequencies of $f_1 \pm f_2$. A third-order nonlinearity generates intermods at $2f_1 \pm f_2$ and $f_1 \pm 2f_2$.

I examined woofer IM distortion first by inputting 900Hz and 1kHz signals at equal levels. These frequencies are far enough below crossover frequency that they should appear predominantly in the woofer output. I adjusted total SPL with the two signals to 90dB at 1m. The M1 output spectrum for this test is shown in *Fig. 15*. The two largest spectral lines represent the input signals. The primary intermodulation distortion products are as follows: 1900Hz = 900 + 1000; 1100Hz = 2 × 1000 − 900; 2800Hz = 2 × 900 + 1000; and 2900Hz = 2 × 1000 + 900.

Other lines on the plot are due to harmonic distortion already discussed. All significant IM products are second-order. The dominant IM product at 1.9kHz is down 54dB from the main output, which is equal to 0.2%. This is better than some solid-state amps and most tube amps!

I next measured tweeter intermods

to page 54→

SB *Mailbox*

GRILLE REMEDY

Lionel Marks requested help with deteriorated foam grilles on his Beveridge ELS in *SB* 1/99 ("*SB* Mailbox," p. 54). I replaced those on my Beveridge IIIs in 1993 using custom materials supplied by Custom Sound (PO Box 463026, Mount Clemens, MI 48046). The material was cut to my specification and successfully attached to the aluminum strips with quick-tacking craft glue. If Custom Sound is no longer providing this service, perhaps Speaker-Works (4732 South Mingo, Tulsa, OK 74146, 800-526-8879) can help.

Bob Moore
Salem, OH

VCR SETTING FOR 2000

Here is a tip that you'll need for next year. You won't be able to use your VCR's programmed recording feature. Do not throw away your unit, however. Instead, simply set it on 1972, because the days will be the same. Please pass this on because you know the manufacturer will not share this information. They will want you to buy a new one that is "Y2K compliant."

Charles Hansen
Tinton Falls, NJ

DAMPING THE DAMPING FACTOR

Mr. Pierce's *SB* 6/97 article ("Much Ado About Damping Factor," p. 52) pretty well demolished the myth that extremely high damping factors are necessary for good, controlled bass reproduction. At the same time, his study shows that very low damping factors can produce rather high response peaks. His study ignored the effects of cable and series inductance resistances; however, you can still use his Table 1 to evaluate the effects of such interceding resistances. Merely read the table from right to left, ending at R_S.

For example, if G_H is to be limited to .11dB, the table shows R_S to be .8Ω. Interpret this as the total of internal and external resistance. Assume that the external resistance is .5% of the nominal resistance, which would correspond to .4Ω. Since the total is .8Ω, the internal resistance must be limited to .4Ω, which corresponds to a damping factor (DF) of 20. (I believe that the concept of DF came from the theories of the stability of systems, which, though perfectly valid, are not necessary to the rational design of bass systems. We now have Thiele/Small parameters, which we can use to calculate amplitude response, phase of response, impedance, phase of impedance, delay, and transient response.)

Source resistance affects performance not only in the bass region, but also in the rest of the frequency range. In that range, the source resistance and system impedance act as a voltage divider on the internal voltage. Obviously, if system impedance is uniform, no response anomalies will occur. But most speaker systems' impedance is quite variable.

Stereophile magazine adopted a simulated loudspeaker load for testing amplifiers in about 1995. This load has a minimum impedance of 4Ω at about 3.1kHz, with a zero phase angle, and a maximum impedance of 19.2 at about 1.2kHz, again with a zero phase angle. Then, if the previous case (internal impedance of .4Ω) is used with the simulated load, the 19.2Ω case will produce a voltage at the speaker terminals that is down by .18dB from the internal voltage, and with the 4Ω load the voltage at the speaker terminals will be down by .83dB relative to the internal voltage, for a net difference of .65dB.

If the deviation in this region is to be limited to .11dB, then it can be shown that the internal impedance of the amplifier must be limited to about .065, which corresponds to a DF of about 123. From this analysis, it would appear that for the same level of response deviation, a higher DF is necessary to control response deviations in the upper frequency range than in the lower range.

As to what's available in amplifiers in terms of DF, of 24 units reviewed in one year's issue of *Stereophile*, six had DF between 130 and 400, six were between 50 and 100, five had DFs between 13 and 40, and seven were between 1 and 8.

D.J. Meraner
Scotia, NY

Dick Pierce responds:

I generally agree with Mr. Meraner's analysis. Let me restate, though, my position on the whole notion of the so-called damping factor in an attempt to clarify my objections to the concept. As a single figure-of-merit describing the amount of control the amplifier has over excessive motion of the speaker cone, damping factor in its current form is essentially useless.

To illustrate the point, a speaker with an undersized magnet—say, one with a system Q of 2.0—will simply not be controlled any better by an amplifier with a damping factor of 100,000 than a damping factor of 30. The speaker will be excessively boomy no matter what amplifier you connect it to. This points out the obvious flaw of the damping factor specification: you cannot predict system performance using an irrelevant specification of but one of its components. Mr. Meraner's comments merely enforce that point.

My beef with damping factor comes from the misappropriation of the term "damping." In the context of mechanical or electrical resonant systems, it has a very specific meaning, and is, in fact, proportional to the reciprocal of the Q. It is not, and has never been, a measure of but one small and insignificant part of the loss mechanisms of such systems, and it is the total losses in these systems that is responsible for damping. Damping factor, as misused and abused by manufacturers, reviewers, authors, and consumers alike, is an empty, useless, and wholly misleading specification.

In conjunction with my criticism of damping factor—often when it is used in the context of whether or not speaker cables can impart different sonic characteristics to a system—I have been very clear on what role damping factor plays: different cables may well make a difference in the sound of a system—that is not what I am disputing. Rather, to point to damping—specifically, the control of the motion of the cone at those frequencies where it is most needed (resonance)—as the source of the difference is either naive or dishonest. I would assign naivete to the vast majority of consumers and dishonesty to the marketing works at the manufacturers and magazines.

If we want to see the effect of non-ideal output impedances on the performance of the system, fine, and that could be very significant. To do so requires detailed knowledge of the properties of that non-ideal impedance, along with the characteristics of the load and other factors as well. How many manufacturers provide such information? How many reviews of amplifiers are done with realistic signals and loads? And how would we go about presenting the results of such reviews in a way that is meaningful and comprehensible to the users?

Regrettably, we have been trained–much through modern marketing practices–to expect that there are single-number figures-of-merit that will tell us all we need to know to make a rational purchasing decision. From soap that's 99 and 44/100% pure to vehicles with five second 0 to 60 MPH times or 50 MPG, we've grown used to being fed single-all-important numbers. It's assumed that our attention span will support nothing more. As a result, these numbers become less and less meaningful the more marketing credence the manufacturers try to cram into them. And damping factor, unfortunately, is one of those numbers that has huge marketing significance and little technical relevance.

One of the most unfortunate outcomes of this is that the subjective crowd, led by some of the more strident writers for the high-end magazines, has seized upon the technical irrelevance of numbers such as damping factor, THD, and signal-to-noise and built an entire army of strawman arguments, railing against the entire concept of measurement. They use advertising spec sheets as their examples of science, then argue against that science. They look at a number like THD or damping factor, which most competent practitioners know is meaningless, and then proceed to lambaste those competent practitioners for promulgating meaningless data.

It's not unlike the battle cry often heard against CD, where many subjectivists scoff at the old "perfect sound forever" line, using it as a club to beat the technological crowd. In fact, the "perfect sound forever" is a pure marketing line, and no engineer has ever subscribed to that silly thesis. No matter, it is seen as a rallying point for some in the subjective crowd. It is a classic strawman argument.

ABOARD THE B-LINE

I enjoyed your B-Line article (*SB* 1/98, p. 8). My first speaker was an AR (sorry, I sold it). Then I heard a transmission-line, and I was hooked! When you select a speaker for t-line use, what specs do you look for?

D.G. Hall
Gardena, CA

Regular Contributor John Cockroft responds:

As a general statement, I'd say that a rather wide range of speaker parameters will work successfully in my various hybrid transmission lines. In fact, I believe that if someone handed me a driver of totally unknown (to me) parameters, it would probably work in my lines.

Having said that, some parameters are to be favored. I prefer the tight natural bass offered by drivers with relatively low Q_{TS}. This can be said without qualification if F_{SA} is low enough, say in the mid 20Hz and below range. Here the higher rolloff of the bass frequencies will be occurring in a low frequency range and will offer the most natural bass to below the range where most music is realized. This kind of bass doesn't even sound like bass; you are merely aware that the individual musical instruments seem to go down to the lowest areas of their capability, and hall noises often sound as though something is going on outside somewhere.

For most people used to hearing reverberant bass, this lean bass takes a bit of getting used to, but once you are happy with it most other woofers sound artificial and unnatural. Q_{TS} values of from about .25 to about .3 and maybe a bit higher should give you this sort of sound. For people who haven't reached this state of happiness, speakers with higher Q_{TS} would certainly be in order.

Speaker design is a matter of achieving what you like to listen to. There is no magic number for anything in speaker design. I relate what will produce the type of sound I prefer, but that's just one sound in a continuum of sounds that can be produced.

Where you are dealing with higher F_{SA} frequencies, the choice is more limited. If you opt for lower Q_{TS}, you might find the frequency rolloff of the response curve begins at too high a frequency. In that case, you could obtain a more natural frequency balance by raising Q_{TS} so the underdamping effect of the higher Q_{TS} will tend to straighten out the response curve at the lower end. This will be at the expense of a slightly fuller, looser bass.

There will be a spot where these two factors will be optimized to your taste. Only you will know when you have reached it to your satisfaction. You could try a speaker with a Q_{TS} of .5 or .6 and compare it to one of about .3 or .35 Q_{TS} in a given line, and you'll probably know more than I do about the matter (your ears are probably younger).

After reading your article on the B-line (*SB* 1/98, p. 8), I just had to build one to see whether I could get that much bass out of a small transmission-line. Well, I've finished building, but I don't have the low-frequency extension I had hoped for. Perhaps you could shine some light over my shoulder.

First, when I called Parts Express, they said that the Dayton woofer you used (295-220) has been improved and replaced with #295-305. F_S of the improved version is listed at 33Hz (vs. 35Hz) and Q_{TS} is 0.33 (vs. 0.29).

Second, I used polyester and wool for stuffing. After not hearing the bass I desired, I removed all of it and weighed it on my kitchen scale—a little less than 1.5 lb. I used another 20 oz bag and then weighed out the remaining 9 oz from stock I had on hand. I stuffed more in each end and I left the area behind the woofer free, having lined it with felt to ⅛″ thickness.

I've positioned the B-line up against a wall, with the woofer on the side. I'm listening to *Stereophile*'s CD #3, track 17, the bass decade warble tones. I've adjusted the volume to a moderate level at 50Hz. At 40Hz, it drops quite a bit; at 31.5Hz, I can barely hear it, so 25Hz is out of the question. From what you write in your article, it seems that I should be able to hear the 31.5Hz tone quite loudly... and even the 25Hz tone. I pulled the woofer out and fed it a 30Hz sine wave of sufficient amplitude to break it in for several hours. After reinstalling it, I did notice more bass, but not much.

I'm curious as to whether my improved version no longer fits the design. I'm tempted to pull out my Mitey Mike and measure both woofer and port in the near field.

Are t-lines supposed to roll off above a certain frequency? This one plays all the way past 3kHz. I noticed that your design called for rolling off the Simplines on the low end to remove the bass.

From the graph of Bailey's article (Fig. 1), am I correct in assuming that the frequency response of the B-line is supposed to be relatively flat from 20Hz to beyond 100? Please explain.

I re-read your article on the Simplines and noticed you used lead shot to lower the resonant frequency of the small driver. Is that applicable in this case?

Mark Florian
Austin, TX

Regular Contributor John Cockroft responds:

If I follow your stuffing story correctly, you have about 33 oz of stuffing compared to the 28.5 oz in the prototype. Since you say you left the area behind the speaker free of stuffing, you have an even higher stuffing density than is first apparent.

You should fill up the whole line at this time, using all of the stuffing. Then remove some of the stuffing from behind the speaker and carefully save it until after you glue the final panel in place. This allows you to get back inside the area behind the speaker to add more glue or whatever else you want to do there. Then replace the stuffing before installing the woofer. You should also do the same at the other end, where there is no support for the stuffing until the last panel and the top plate are in place.

I later raised the stuffing density in my B-line to 1.125 lb/ft^3. This took a total of 30.8 oz. Simply pull out a couple of ounces first and then pull some of the stuffing down into the space behind the woofer to even things up a bit. I eventually cut the hole bigger and installed an 8″ MCM woven carbon fiber woofer, which I like. I also like the 6½″ version.

You said you used a moderate level setting when you ran your tests. You may have experienced the Fletcher-Munson sound threshold effect. At lower frequencies sound must be played back at higher sound pressure levels for our ears to hear. That's why the "loudness" control was born a few decades ago.

Very low frequencies in the area of 30Hz and below present very little to none of what we generally perceive as "sound." Rather, what we perceive is a physical pressure which our senses interpret as an ambience. If you hear a "sound" in this area, you are hearing evidence of harmonic distortion (sometimes called "doubling," if it happens to be composed of second harmonics).

With little cone area and usually rather short strokes, 6½″ drivers are generally incapable of moving enough air to produce this sense of pressure, even though they may be moving to the signal. Further, low frequencies are subject to room size, room stiffness, room placement, and listener placement. I wasn't trying to present the little Dayton woofer as the optimum woofer to obtain low bass with this system. It was merely a driver I had on hand.

It was never my intention to re-create Dr. Bailey's system. I was merely trying to come up with a set of operative numbers to achieve a successful design for a given speaker. When used in high-quality systems, the woofers produced very natural bass that allowed instruments to sound like themselves in their lower registers, much as they sounded in the midrange (only lower). There was almost no sense of a woofer being involved, just the music. Since this was what I was trying to achieve I considered that my efforts were successful.

I recommend placing the B-line speaker face to the rear wall in a corner. In my room with a crossover of about 106Hz, the blend is seamless between the corner woofer and my Super Simpline Phoenix satellites. The crossover is a simple line-level passive second order one patterned after the one in my Simpline Sidewinder Woofer article (*SB* 4/95) (I substituted .147µF caps for the .1µF caps in the article).

There are reasons why the B-line's response curve might be different from the Bailey curve: The difference in stuffing densities offers different damping qualities as well as air speeds. I'm using Bradbury's equation (which he devised for wool) with polyester fiberfill. The line formats differ. Bailey's line allows the woofer to exhaust into a relatively large "chamber," from which it continues down a long tapered, folded line and exits through a port that is considerably smaller than the cross-section of the initial chamber.

Further, the system is a free-standing one that sits out in the room. The B-Line is a straight pipe of square cross-section that maintains its constant cross-section the entire length of the pipe. The exit port is the same cross-section (in some B-line designs I have reduced the exit port to about 85% of the line cross-section with no observable compromise). The B-line is situated up against the wall and makes use of the room boundary in the manner originally discovered by Roy Allison.

I'm not particularly concerned whether differences exist. What interests me is that successful hybrid transmission lines can be designed and constructed using the operational numbers I obtained from Bailey's response curve. I once read that Dr. Bailey's woofer was probably 25Hz (rather than the "about 30Hz" he states in his article). If true, this gives a more "elegant" situation, placing F_{SA} just about on F_3 of Bailey's curve. F_5 would be $.8F_{SA}$ and F_{10} would be $.66F_{SA}$ (you could say $.7F_{SA}$ for a set of figures easily kept in your head).

This results in a line length slightly shorter and a slightly higher F_{10}. For the Dayton driver in the article with $F_{SA} = 35Hz$, F_3 should = 35Hz, F_5 should = 28Hz, and F_{10} should = 23.1Hz. This is at least a step in the direction toward what you heard, so these numbers are perhaps the ones to be used.

If your main mission is to achieve more bass, you should consider using a speaker with a higher Q_{TS}. You could play around a bit with your Dayton woofer since it was rather cheap and it doesn't satisfy you in its present condition. A simple way is to add series resistance to the woofer: 2Ω would raise your current .33 Q_{TS} to .4125 and give you a sensitivity loss of about 1.94dB, 3Ω would yield a Q_{TS} of about .45 and give you a loss of about 2.77dB, and 4Ω = .495 Q_{TS} and produces a loss of about 3.5dB.

With those losses and the low power of the woofer, you'd be able to make qualitative tests, but these will be a bit anemic. If these seem like a step in the right direction, you could try adding mass to the cone (instead of the resistors).

The advantage of this is that you lower F_{SA} at the same time you raise Q_{TS}. Of course, you also lose the sensitivity. You are learning lesson number one in audio: Bass is a direct trade-off for efficiency!

By lowering F_{SA} you also put a greater strain on the cone suspension because of the greater excursions required by the lower frequencies involved. This has the effect of reducing maximum sound levels. You could use modeling clay, putty, or Mortite. Have some fun and then remove the mass.

My published article stated that F_{10} was .52 F_{SA}. It should be .55 F_{SA}, since .52 is about F 12.5, I think.

Afterburner For Aftershock

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transients, but testing the system on an old speaker set my mind at ease. If Afterburner is powered up when I power up the Pro Logic receiver, there is quite a thump. I must make sure Afterburner is turned on *after* the receiver. The slight thump that the receiver causes when I turn it off reminds me also to turn off Afterburner.

My final concern was putting my Sony 32" TV on top of Aftershock, not so much that it would vibrate off, but that the Aluma Pro Alusion 12" driver would affect the picture. It does not, even though the speaker is not shielded. *Photo 2* shows the TV, Aftershock, and Afterburner in their new home in Atlanta.

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with a 10 and 11kHz input pair, also adjusted to produce 90dB SPL at 1m. The results are shown in *Fig. 16*. A number of intermodulation distortion products appear at 1kHz intervals on the plot. The largest IM product at 9kHz is 50.4dB down from the main output at 0.3%. Total tweeter IM distortion is 0.35%.

Dennis Colin remarked in his review that one of the tweeters had a slight audible roughness. I tested the second tweeter for IM distortion, and the resulting spectrum is shown in *Fig. 17*. Total IM distortion for this tweeter is 0.8%. This higher distortion level may account for the perceived roughness.

The last IM test examines cross intermodulation between the woofers and tweeter, using frequencies of 900Hz and 10kHz. (A 1kHz signal would produce intermods that fall on harmonic-distortion lines, confusing the results.) This spectrum is shown in *Fig. 18*. The largest IM products fall at 6.4, 8.2, 9.1, 10.9, and 19.1kHz. Relative to higher-order crossover networks, the second-order crossover slopes allow more low-frequency energy into the tweeter and more high-frequency energy into the woofer, but the total distortion is still only 0.3%.

ADDITIONAL TESTS

I conducted all of the above tests with the grille off. *Figure 19* shows the response of the M1 with the grille on, but referenced to the response with the grille off. That is, it plots the difference in response under the two conditions. Below 2kHz, the grille causes the on-axis output level to rise about 0.5dB, but the response is still relatively smooth. Above 2kHz, however, the grille causes ragged response deviations of 5dB peak-to-peak. Listen to this system with the grille off.

Most of the tests described so far were conducted with M1 #0020018. One question of interest is how well the two samples match. Frequency-response matching with the grille off is shown in *Fig. 20*. This is a plot of the response difference between the 0020018 and 0025215 samples. Below 2kHz, the systems match with 0.5dB. Above 2kHz, you can see mismatches of +1.5 to −1dB. However, there is no clear trend in the mismatch that would pull the listener's attention to one speaker or the other.

A NOTE ON TESTING

The Swans M1 was tested in the laboratories of Audio and Acoustics, Ltd., using the MLSSA and CLIO PC-based acoustic data acquisition and analysis systems with an ACO 7012 ½″ laboratory-grade condenser microphone and a custom-designed wideband, low-noise preamplifier. Polar response tests were conducted with the aid of a computer-controlled OUTLINE turntable on loan from Old Colony Sound Lab of the Audio Amateur Corporation.

REFERENCES

1. J. D'Appolito, *Testing Loudspeakers*, Chapter 4, Audio Amateur Corporation, Peterborough, NH, 1998.

2. *Testing Loudspeakers*, Chapter 6.

Manufacturer's response:

First of all, I want to express our gratitude to Dennis Colin, Joe D'Appolito, and certainly *SB* for a great job that they have done. I should acknowledge that reviews with this kind of profound approach provide invaluable service to all of us readers, including manufacturers. While loudspeaker design still has a lot of white spots, sometimes there is too much unnecessary mysticism going on in the industry.

Providing a profound subjective evaluation, followed by what I would call an exemplary test and correlating the results gives us a clear picture about a product and brings us to a closer understanding of loudspeakers.

Regarding the wrong polarity mark on one of the ribbon tweeters, sometimes something you don't want to happen happens at a very critical moment. Obviously our tweeters undergo 100% polarity QC. The polarity mistake was a big surprise to me because I haven't yet heard from any of our customers about this problem. Anyway, I am thankful that Dennis Colin skillfully fixed this and went on with the review.

There is not much to comment on in such a positive review, except that I would like to respond to the slight gradual rolloff from the midrange to bass region measured by Joe D'Appolito. The Swans M1 were designed with a dual approach using thorough measurements and continuous listening. The initial samples of the speaker indeed had a woofer crossover that corrected that gradual rise due to diffraction loss at low frequencies. But eventually we found that overall presentation was just a bit laid off. The highly resolving character of the ribbon tweeter demanded the same clarity from the woofer. We modified the crossover and were very happy with the results.

Note that these measurements represent a quasi-anechoic condition and show only a direct sound coming from a speaker and do not account for reflections and reverberant energy. In a real room, especially in bass and lower midrange band, the spectrum of sound energy is quite different from the energy of direct sound. Complex room resonances and reverberation (its time gradually rises with decreasing frequency) create additional sound energy that we perceive in combination with direct sound. This dramatically affects the overall balance.

I measured Swans M1 pink-nose spectrum at a listening position in various locations. Sometimes it's shocking how room and positioning change what we think would be a smooth result in room response looking at an anechoic curve. In the case of M1, you can get quite balanced and resolving low-frequency sound by spending extra time carefully setting up and placing the speakers. In fact Dennis Colin's experience proves this point.

I would like to mention that now Acoustic Technology optionally offers a pair of gorgeous cabinets that are available in a piano black, solid walnut, or natural rosewood veneer finish. You just have to put Swans M1 together according to the user's manual and enjoy the music.

Igor Levitsky
Acoustic Technology

Ask SB

CROSSOVER DEBATE

I have been interested in building enclosures since age ten, when I acquired my first "orphan" speaker. Usually, out of frustration, someone gives me two speakers, in lieu of junking them, and then I make them work.

My questions pertain to "A Modest-Cost Three-Way Speaker System, Parts 1–3," by G.R. Koonce and R.O. Wright (*SB* 6–8/96). Why develop a crossover for a SEAS tweeter no one wants? Wouldn't an all 8Ω crossover avoid the "sonic degradation" mentioned?

Guy Bernardin
Havertown, PA

G.R. Koonce responds:

You first ask why we included the SEAS tweeter. Remember, we started out trying to build a two-way system, so tweeters purchased for testing were those with low resonance, including the SEAS 25TAC/G. When we changed to a three-way system we hoped for a first-order crossover, so staying with the low-resonance tweeters retained both a soft- and a hard-dome tweeter in the design. This would allow builders to compare the sound of these different types. Perhaps if manufacturers had supplied free samples of all available low-resonance tweeters, we would have selected different units; but when you are buying the drivers your sample quantity is limited!

Your first question indicates you do not like the SEAS tweeter, but I don't agree that no one wants them. In the three-way systems covered in the article, I admit I clearly prefer the soft-dome Audax TW025MO tweeter.

Last summer I helped a friend develop an 8″ two-way system for sale in modest quantity and it contained the SEAS tweeter. He and I both liked the sound of these systems very much and potential customers have commented on their sound. It seems not everyone likes the same tweeter sound, and different systems seem to sound best with a particular tweeter.

You next ask if an all 8Ω crossover design would have been better and avoided the sonic degradation discussed in the article. The sonic degradation had nothing to do with the design impedance of the crossover. Initially, we thought it resulted from the rush to package the crossovers for the first pair of boxes. As discussed in Part 3, the problem was an excess of energy around the upper crossover frequency and was cured on the second pair of boxes by modifying the high-pass section of the third-order crossover. Recent work has reinforced this result.

When changing from a piston driver nearing the top of its frequency range to a dome tweeter that will have a wide radiating pattern near the crossover frequency, you must avoid any peaking in the system response in this frequency range. The best sound may actually result with a slight response dip on and about the listening axis in this frequency range.

The three-way system did not use an all 8Ω crossover design, but as shown in Figs. 40 and 41 (*SB* 8/96, p. 34) once you were above the frequency where the impedance was controlled by the woofer motional impedance, the systems were at nearly constant 6Ω loads. In theory, there is some advantage to a design with all drivers at the same impedance, be it 6 or 8Ω, as even a crossover with the constant-resistance characteristic and terminated in fixed resistor loads starts to reflect a non-resistive (input impedance has a phase angle) load when it transitions between two different load values. (See a discussion of this effect in "Crossovers for the Novice," *SB* 5/90, p. 26.)

While a constant speaker input impedance does have merit for high output impedance amplifiers, and some designers include components in the crossover for this purpose, I have never been able to convince myself there is sonic merit to trying to maintain the speaker's input impedance constant or always resistive. As with many other areas in this hobby, the results you hear running this test may depend on the system, the location, the source material, and the listener!

Classifieds

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AD INDEX

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Tools, Tips, & Techniques

NARDEUX N-300

By John Badalamenti

PHOTO 1: Finished speaker.

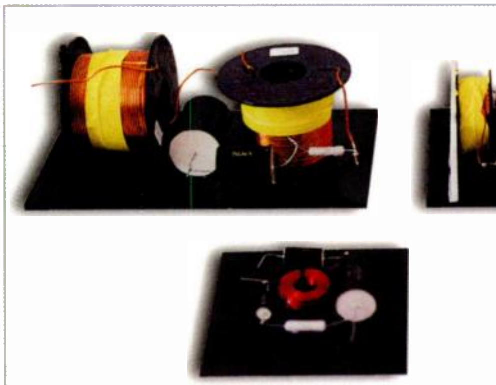


PHOTO 2: Crossover assemblies.

I decided to build my own loudspeakers because I could not find what I was looking for in terms of quality, price, size, power, and dual application for high-power, two-channel stereo and Dolby surround sound (*Photo 1*). I started with an ORCA design, and when I explained my desires, I was referred to Joe D'Appolito. I had of course seen his name in much of the literature I had been reading, and it was an honor to talk to him about my speaker project and to have him personally design my crossover network and cabinet schematics.

I told him I wished to use the best of the FOCAL line, a 15VX woofer, a 7K2 midrange, and a TC120tdx II tweeter. I desired a three-way system capable of delivering a clean, crisp sound with chest-slamming bass that would be full and rich, with solid depth throughout the audio-frequency range. I intended this system to provide professional sound for two-channel stereo and for playing

PHOTO 3: Crossover installation.

Dolby surround tracks to accompany video watching. Joe understood my goals and designed exactly what I wanted.

Once I had the schematics of the crossovers and diagrams for the cabinet assembly, I decided to go to Zalytron for all my crossover parts, loudspeakers, and cabinet manufacturing. I made friends with Eliot Zalys, a very nice guy who was most helpful in setting me up with all that I needed.

PHOTO 4: Unloaded cabinet.

Having purchased all the necessary parts, I assembled the networks for the drivers on individual boards (*Photos 2* and *3*). Then every week I went back to Zalytron and bought my Focal drivers one at a time until I had all of them. Eliot started to work on the cabinet assemblies and proved himself to be a true professional in this field. He also gave me some good hints about improving the design. The cabinets, solid 1.5″ MDF, are built like tanks, with a beautiful red-oak finish (*Photo 4*).

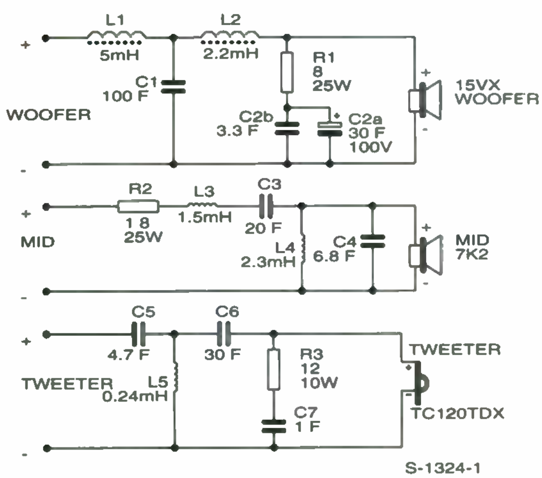


FIGURE 1: Crossover network.

The crossover network is shown in *Fig. 1*, with the values for the components in *Table 1*. Enclosure details are shown in *Figs. 2* and *3*. The front-panel

PHOTO 5: Cabinet loading.

layout (*Photo 5*) is identical to the original three-way system with the Audiom 7K and the T12OTi. Width and height are also the same. The overall depth has been reduced to 15.25". You need as much internal volume as possible with these overall outside dimensions. Therefore, I have spec'd out an interior tunnel enclosure for the new 7K2. Its cross-section is square, with internal dimensions of 7" × 7". It goes all the way from the front to back as shown in *Fig. 3*. The front wall is 1.5" thick. All other walls and top and bottom are 1" thick.

You have two options for the port. One of them,

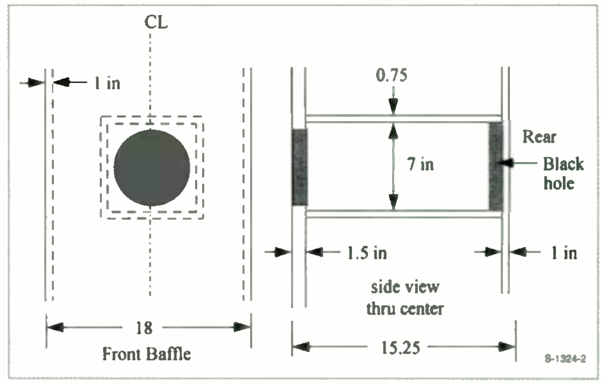


FIGURE 2: Detail showing tunnel for Audiom 7K2 interior dimensions 7" × 7" × 12.75". Line rear of tunnel with 2" Black Hole. Fill tunnel with high loft Dacron at ½ lb/ft³.

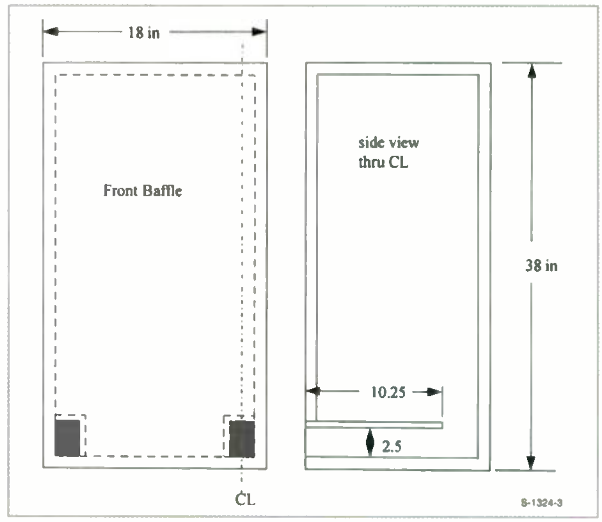


FIGURE 3: Detail side of vents. Use ¾ MDF. Vents are 2.5 in² and 10.25" long.

TABLE 1
PARTS LIST

L1*	5µH ferrite core
L2**	2.2µH ferrite core
L3	1.5µH air-core, #16AWG
L4	3.3µH air-core, #16AWG
L5	0.24µH air-core, #20AWG
C1	100µF, polypropylene (AXON)
C2	30µF, 100V electrolytic with 3.3µF poly in parallel
C3	20µF poly (AXON)
C4	6.8µF poly (AXON)
C5	4.7µF poly (AXON)
C6	30µF poly (AXON)
C7	1µF poly (AXON)
R1	8Ω/25W
R2	1.8Ω/25W
R3	12Ω/10W

* A #12 AWG air core can be used instead of the ferrite core, but the cost and size are prohibitive.
** A #14AWG air core can be used here with the same comment.

REFERENCES

Joe D'Appolito
Audio and Acoustics Ltd.

ORCA Design & Manufacturing Corp.
1531 Lookout Drive
Agoura, CA 91301
818-707-1629, FAX 818-991-3072

Zalytron Industries
469 Jericho Tpke., Mineola, NY 11501
(516) 747-3515

PHOTO 6: Speaker pair, one with grille and one without.

shown in *Fig. 3*, consists of two vents, one on either side of the front baffle at the bottom. The opening is 2.5 in^2. The other approach is to use a single 4″ ID PVC pipe 10.5″ long, positioned at any convenient point on the front baffle. I can't tell exactly where to put it without a detailed drawing of the 15VX, but it should be obvious once the box is together.

Figure 4 shows the computer-predicted response of the system and the individual drivers. *Figure 5* is a plot of system impedance, which is above 6 at all frequencies.

The money I spent on this project is considerable, but it is one of the best investments I've made. These speakers would cost at least two or three times as much in a high-end audio store. I have named them the Nardeux N-300 (dedicated to the stereo system from Loches, France, the Stereomatic 302S, which is the system used on the Himalaya amusement ride in Coney Island from 1965–74) for their 300W of music power. They sound fantastic, and are a beautiful addition to my living room (*Photo 6*).

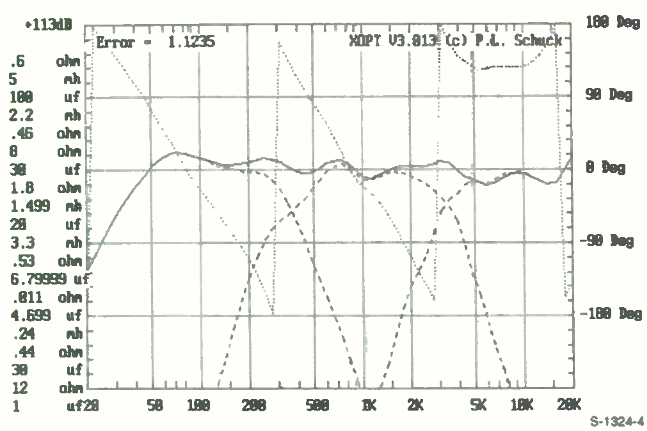


FIGURE 4: System response (solid line), driver responses (dashed), system phase (dotted) (phase with 2m delay removed).

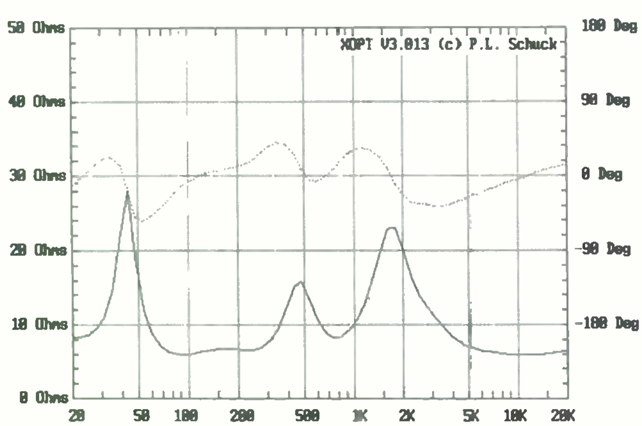
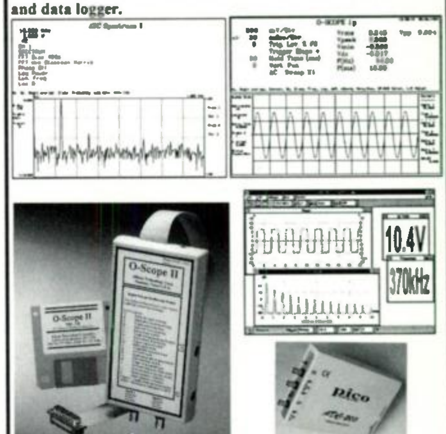


FIGURE 5: System input impedance magnitude (solid line), input impedance phase (dotted).



Showcase

A PANEL SPEAKER FOR TOMORROW

By Andrew Ware

I have recently completed my BA (Hons) Industrial Design Course at the University of Central England, Birmingham (UK). The opportunity to choose the subject for my final project allowed me to bring my interest in hi-fi into my work. I decided to design hi-fi speakers incorporating NXT technology. I did not give myself many restrictions when writing my design brief, so among these I decided not to work on a budget.

I visited the BBC Tomorrow's World Live exhibition at the National Exhibition Center in Birmingham last February, which featured innovative design concepts and new technology. This was where I first saw the NXT panel speakers demonstrated. Having decided consequently to explore the possibilities of this new concept as a degree project theme, I contacted NXT in the UK and received various literature, including a technical white paper, and I was therefore able to gain a greater understanding of the technology involved.

As you can see from the photos, the speaker panels are suspended on tensioned speaker cables, which carry the signal to them. This was possible only due to their having distributed mode operation. Having bipolar sound, their positioning enables the sound to be heard from both surfaces. The cable runs under the base and the terminals are located at the rear of the back foot. At the other end, the cables are attached to a plastic fixing, which in turn is attached to the barrel strainer via the tensioning cable. You can easily adjust this set-up. I covered the speaker panels in calico, but

you may choose other color options to suit various interiors.

I constructed my full-scale model (measuring 1415mm × 320mm × 430mm) in the university workshops with some assistance from the technicians. The model was entirely hand-finished, with machined aluminum, steel, and plastic fixings. The most time-consuming component was the plywood support beam, which I made by laminating 12 layers of three-ply finished with ash veneer, and then shaping it by hand.

My main objectives with this design were to break away from the loudspeaker boxes that are often seen, to produce an aesthetically pleasing form, to provide a focal point to a room, and to give the speaker panels an identity instead of disguising them. Having not given myself a budget to work to obviously made this easier for me.

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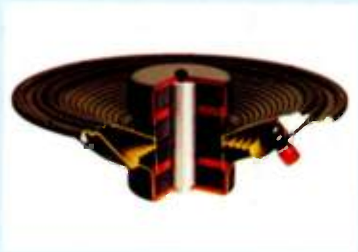
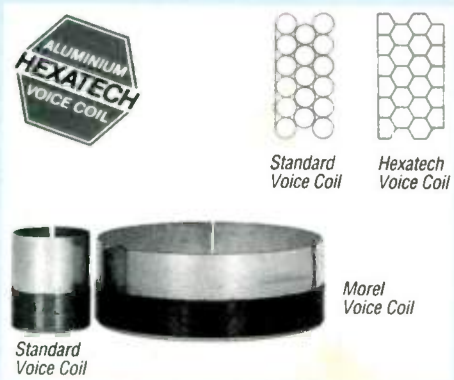
Integra



Push-Pull

Double Magnet

Morel Acoustics USA, Inc. has come a long way since 1978. Currently, the company has a diverse line of exciting products which includes over 40 models of tweeters, midranges, mid-basses, woofers and subwoofers. Being a leader in the field of speaker design, for our 20th year anniversary we are scheduled to launch several new products that are sure to attract attention.



Typical double magnet ducted woofer.

Double magnet tweeter

For further information please contact:

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