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THE LOUDSPEAKER JOURNAL

The Mini Snail
BIG PERFORMANCE IN A TINY BOX
BILL FITZMAURICE

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B. LAMY

AN EASIER WAY TO DESIGN PASSIVE CROSSOVERS
G.R. KOONCE

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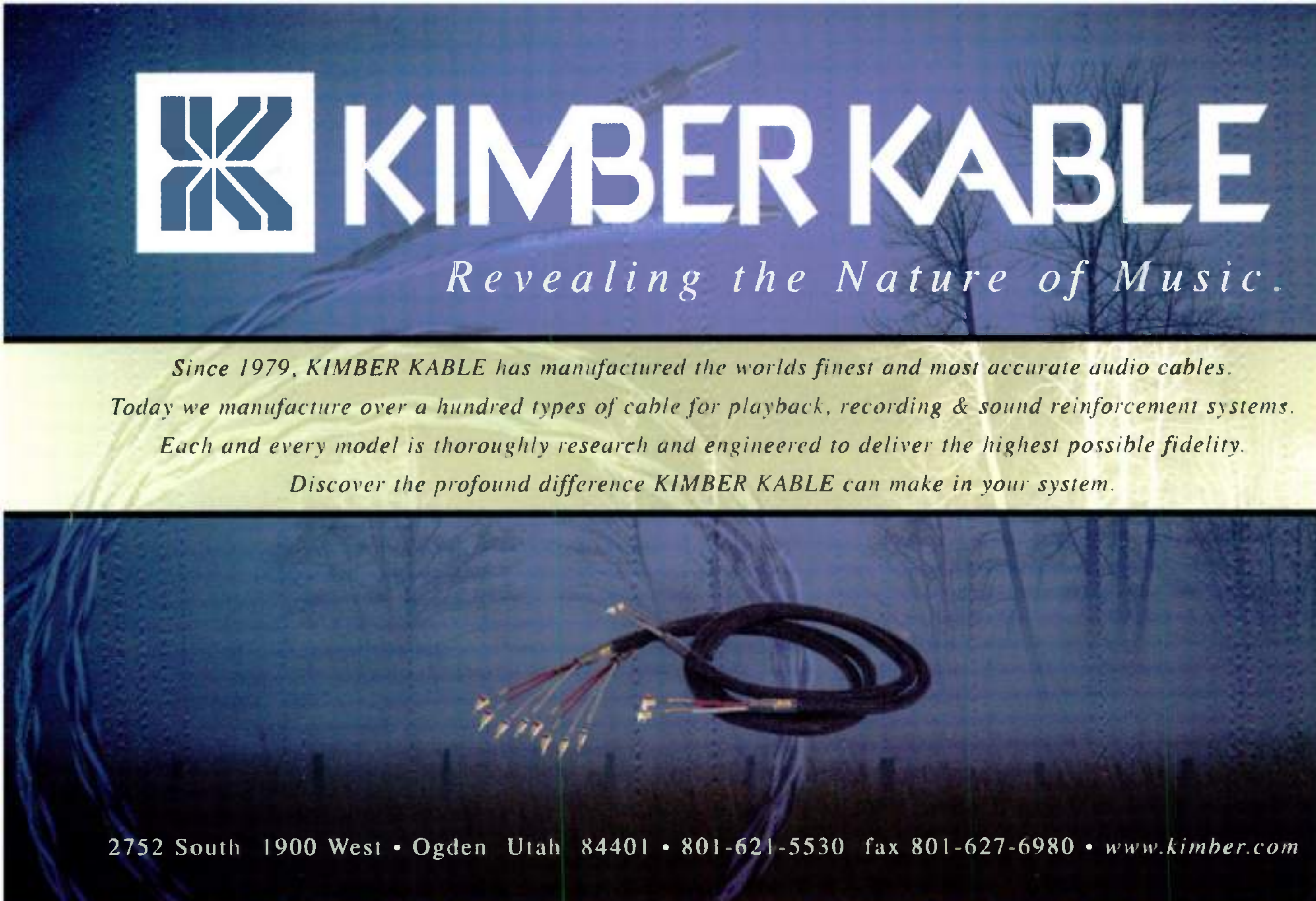
WE TEST: ACI'S POWERED SUB - TOM PERAZELLA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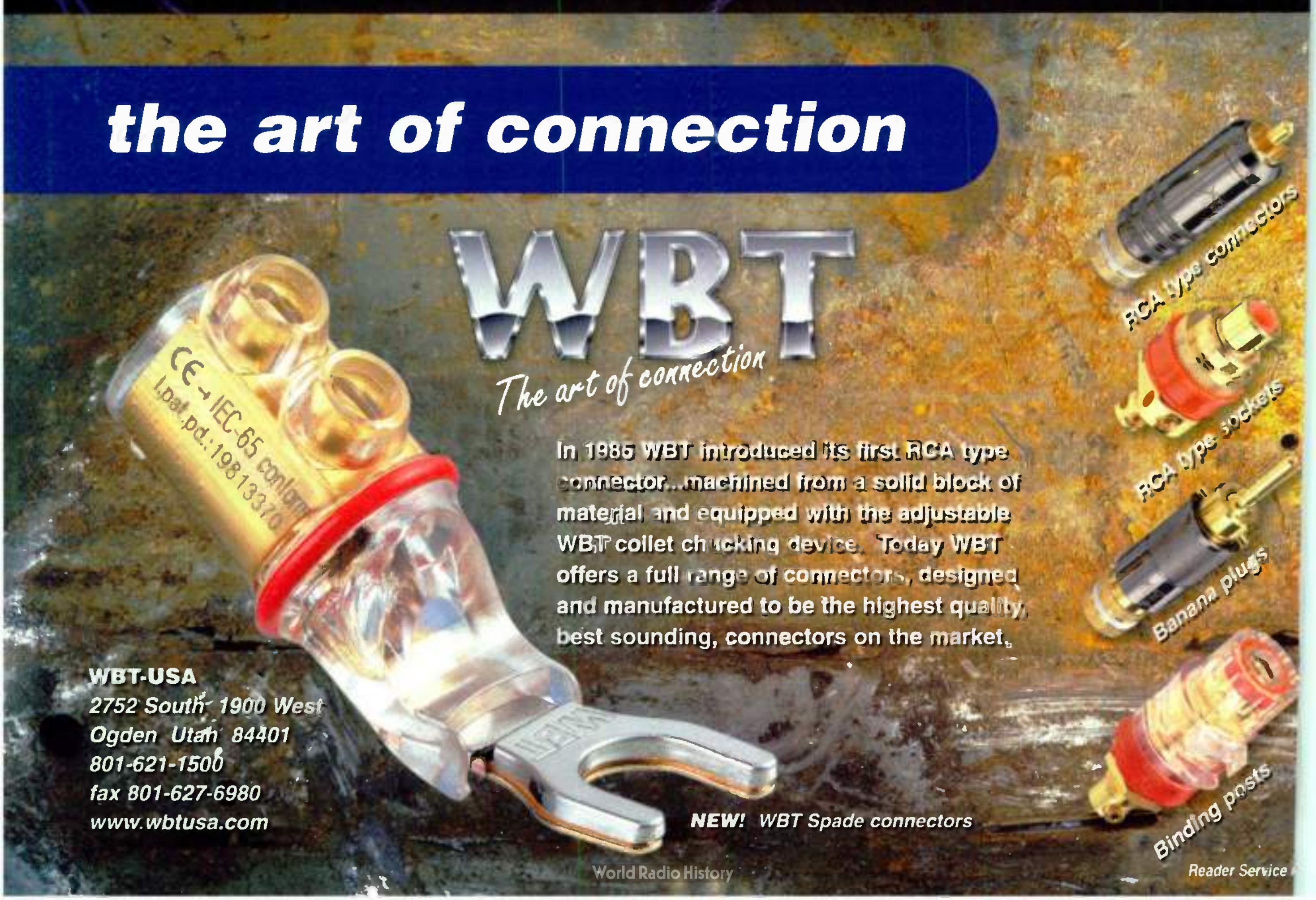
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Tang Band's NWF series of paper cones were developed for hi-fi speaker applications. Made from natural wooden fiber, they are intended to overcome the consistency problem of traditional paper cones and keep the paper sound quality. Advantages include light weight, high stiffness, and good damping properties. Tang Band Industries Co., Ltd., 4F, No. 8, Alley 23, Lane 91, Sec. 1, Nei Hu Rd., Taipei, Taiwan, R.O.C., (+886) 2-26570282, FAX (+886) 2-26580166, E-mail infotb@mail.tangband.com.tw, Website http://www.tangband.com.tw.

Reader Service #135

Good News

NEW BEAU
Thomas Transducers has adopted the name Beauhorn (rhymes with "blow-horn") for its Virtuoso horn speakers and its Obligato single-ended Class-A valve amplifier. You may recall that the company was forced to relinquish its established brand name Bravura last year to avoid a potentially costly dispute. Thomas Transducers, Songlines, 14 Acre Lane, Three Oaks, Hastings TN35 4NB, United Kingdom, (+44) 1424 813888, FAX (+44) 1424 812755.

CROSS-COUNTRY MOVE
True Audio has moved all its operations from Escondido, CA, to 387 Duncan Lane, Andersonville, TN 37705, Voice/FAX (423) 494-3388.

COAXIAL ENCLOSURE
The 115FM floor monitor from L-ACOUSTICS is an active two-way enclosure featuring advanced coaxial technology and high output. Designed for touring, it is a dual-position monitor constructed from Baltic birch plywood, offering near (35°) and longer (55°) throw capability, with 4-pin Neutrik speakon connections directly accessible in both positions. It was developed to be used with the L-ACOUSTIC 115FM, an analog controller that provides consistent and safe operation through the use of an active sense return from the amplifier outputs. Cox Audio Engineering, 10741 Sherman Way, #7, Sun Valley, CA 91352, (818) 503-1550, FAX (818) 503-1553, E-mail Coxaudio@aol.com, Website http://www.coxaudio.com.

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COLLECTION OF CATALOGS
McFeely's latest catalog includes an array of square-drive screws, drill bits, measuring and marking tools, cabinet hardware, sanding supplies, and much more. McFeely's Square Drive Screws, 1620 Wythe Rd., PO Box 11169, Lynchburg, VA 24506-1169, (800) 443-7937, FAX (800) 847-7136.

In Mouser's most recent catalog, you'll find wire and cable, resistors, capacitors, speakers, power supplies, cabinets, kits, and more. Mouser Electronics, (800) 346-6873, Website http://www.mouser.com.

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Reader Service #22 →

About This Issue

Musician and speaker designer **Bill Fitzmaurice** just keeps gliding along (albeit quite slowly) with his Snail series. His latest horn design, entitled MiniSnail, proves that you don't need to sacrifice good sound with a small design. This PA system not only is a crowd-pleaser, but is also easy to transport to your next live performance ("The MiniSnail," p. 8).

The ravages of time take a toll on everyone and everything, including speakers, which are susceptible to the deadly disease foam rot. **David Kelley** and **P.-T. Ho** offer a remedy for your ailing speaker ("Saving Aging Foam Cone Surrounds," p. 16).

Frequent contributor and noted speaker expert **G.R. Koonce** offers a different approach to passive crossover development. Crossover modeling takes the tedium and drudgery out of this time-intensive process ("Modeling for Designing Passive Crossovers," p. 20).

In his two-part series, French author **B. Lamy** picks up where he left off with the construction of the innards, designing and assembling the box, and testing this sleek-looking 4-way unit with a servo-controlled sub ("The Force," p. 32).

In this issue's "Kit Review" (p. 50), **Thomas Perazella** does a thorough job of reviewing ACI's Titan subwoofer. He assembles the kit, constructs an enclosure to house the unit, and provides acoustic measurements.

Happy New Year From The Staff Of Speaker Builder!

Speaker Builder®

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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Contact Advertising Department, Speaker Builder, PO Box 876, Peterborough, NH 03458, 603-924-7292, FAX 603-924-9467, E-mail advertising@audioXpress.com.

Speaker Builder (US USSN 0199-7920) is published every six weeks (eight times a year), at **$32** per year, **$58** for two years; Canada add **$8** per year; overseas rates **$50** one year, **$90** two years; by Audio Amateur Corporation, Edward T. Dell, Jr., President, at 305 Union Street, PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH and an additional mailing office.

POSTMASTER:
Send address change to:
Speaker Builder, 305 Union Street
Peterborough, NH 03458-0876

Speaker Builder
THE LOUDSPEAKER JOURNAL

VOLUME 19 NUMBER 8 JANUARY 1999

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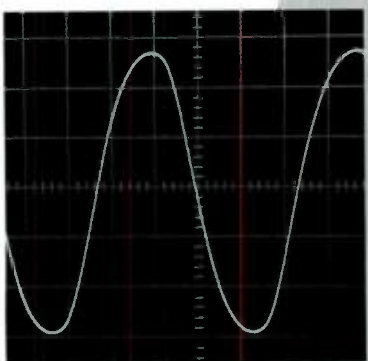

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THE MINISNAIL

By Bill Fitzmaurice

The Snail series of folded horns (*SB* 6/97, 3/98, and 4/98) has shown that you can obtain high efficiency and wide bandwidth from modestly sized cabinets. My latest project, the MiniSnail (*Photo 1*), is intended to demonstrate how small a folded-horn cabinet you can construct that will still outperform similarly sized "traditional" designs.

I intended to build a small PA system for small venues that do not require running the bass, keyboards, or kick drums through the PA; a cabinet attuned primarily to the vocal range, 100Hz–15kHz, would be sufficient. However, to overcome the high SPLs of the stage amps, it would require high sensitivity in excess of 100dB/W. At the same time, small size, light weight, and low cost would also be desirable. Could a very small Snail design do the job?

PHOTO 1: Finished MiniSnail.

BIG LITTLE SNAIL

In fact, the MiniSnail has powerhouse performance in a pint-sized package. Not much more than 2ft^3 in volume, and weighing only 28 lb, this diminutive blockbuster will easily handle small-club (100 seats or less) requirements.

This cabinet utilizes an eight-inch woofer from Carvin, model PS-8, which has a two-inch voice coil and is rated at 200W. (Carvin drivers appear to be OEM units sourced from Eminence, and have excellent price/performance ratios.) I used two "twin bullet" Motorola #KSN1177 piezo tweeters for the main high-frequency duties, and a Motorola #KSN1167A single "bullet" is mounted on one cabinet side to eliminate the usual high-frequency "dead spot" directly in front of the band. I mounted the twin-bullet drivers vertically, since testing showed that their horizontal dispersion is wider when they are so aligned.

The PS-8's Thiele/Small (T/S) specs are an f_S of 80Hz, a Q_{TS} of 0.37, a V_{AS} of 0.43ft^3, and an SPL of 94dB/W. T/S theory predicts a maximum flat response from a box of 0.37ft^3, tuned to 82Hz, with a resulting f_3 of 84Hz. With a sensitivity of 94dB, it has an average output capacity, with 100W input, of about 114dB at 1m—plenty for a living room, but about 6dB shy of that which a live performance requires.

On the other hand, once it's mounted inside the MiniSnail folded horn, the average sensitivity jumps to over 102dB (*Fig. 1*), which allows it to reach the required output levels. (I took these measurements with the cabinet on a stand 7′ high, where it normally would be placed for proper dispersion. When it's on either the floor or a platform, such as a stage, the response below 150Hz is on average 4dB higher.)

While the box frequency as built would be about 80Hz, the impedance graph (*Fig. 1*) shows that the horn-loading of the box-vent output lowers the box frequency to about 60Hz, which extends the driver's f_3 down to 60Hz as well. The f_h (a term I coined to denote the corner frequency of the horn) is noted by the impedance dip at 130Hz, above which the cabinet sensitivity rises dramatically.

SMALLER IS BETTER

Why use an 8″ woofer instead of a 10″ or 12″? Because the increases in sensitivity that horn loading provides make a larger woofer unnecessary by allowing better midrange response while keeping cabinet size, weight, and cost down. An 8″ woofer also permits using a small cabinet that still has three "bends" of the horn, maximizing the horn's length, and thus its sensitivity.

However, more bends in the horn also cause high-frequency attenuation. To counter the attenuation effect of three horn bends, the throat has both an oval-shaped hole in the baffle, and a curved throat reflector. The result is effective loading of the driver all the way up to 3kHz. (I consider that the effective loading of the driver at high frequencies ceases when the cabinet no longer has a sensitivity greater than that of the driver mounted on a flat baffle.)

The listed piezo tweeters have a nominal low-end cutoff of 3.5kHz, which results in a response dip in the range from 1.5–3kHz. This dip is intentional, since my experience has shown that I must always roll off my equalization (EQ) no less than 6dB at 2.5kHz to control feedback. By having a

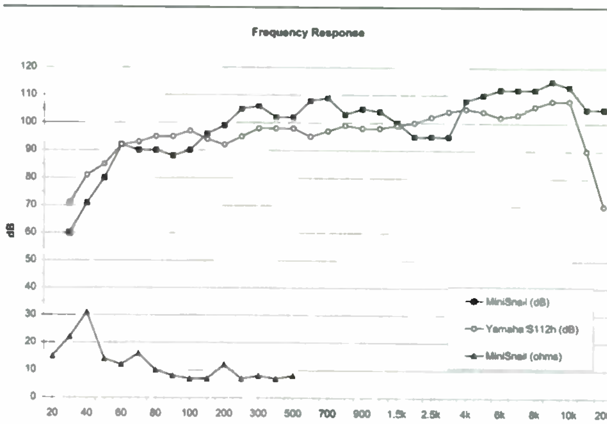


FIGURE 1: Impedance graph.

natural response dip at this point, I need less of an EQ reduction. This would be most beneficial to those who do not have either a graphic EQ or a sweepable mid EQ, which are often not available on inexpensive PA consoles. (If you desire a flat response curve, there are Motorola units available that work down to 1.8kHz.)

For comparison's sake, I tested the response of a typical commercial small PA unit, the Yamaha S112H. With a 12″ woofer and dynamic horn tweeter, this cabinet performs well—better than the MiniSnail below 130Hz. However, that is a moot point for vocals, which don't extend much below 120Hz. From 150Hz–1kHz, the MiniSnail is much more powerful than the Yamaha. At the high end, the dynamic tweeter of the Yamaha dies abruptly above 10kHz, while the MiniSnail's piezos cruise happily along to well above 20kHz.

In side-by-side testing in live performance, the MiniSnail simply sounds better and is definitely louder. While the Yamaha is about the same size as the MiniSnail, its larger woofer and particleboard construction bring its weight to 50 lb—nearly twice as much as the MiniSnail. The average cost of a speaker similar to the S112H ranges from $150–$300, while you can build the MiniSnail for as little as $100 per unit, depending on the materials and finish used.

I built my MiniSnails using ½″ plywood for all parts except the horn panels, which are ⅜″ plywood. You could use cheaper waferboard instead of the ½″ plywood, though it is harder to work with. Also possible are ¾″ plywood, particleboard, or MDF, but the self-bracing Snail design does not require heavy panels to eliminate vibration; all you'd gain would be more weight and additional cost. You fasten all joints with construction adhesive and drywall screws piloted and deeply countersunk to allow chamfering of the finished cabinet edges. Use either 1″ or 1¼″ screws, the shorter size where necessary to prevent penetrating exterior walls.

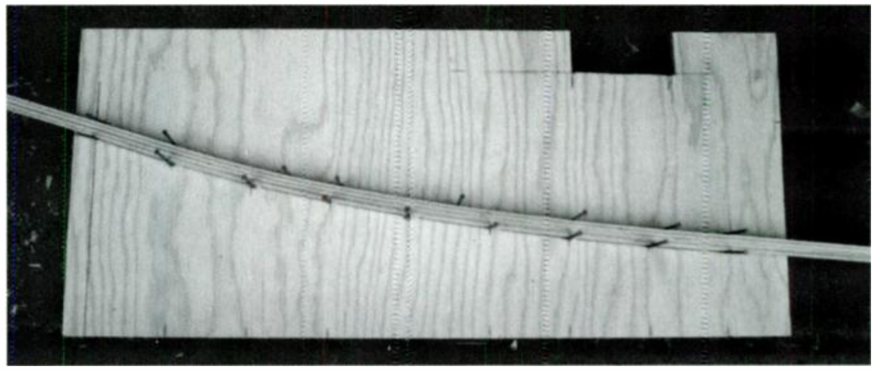


PHOTO 2: Creating the brace patterns.

TABLE 1

PARTS LIST

Note that all part sizes are nominal, depending on the actual material used for construction, and relate to the materials suggested in the text for each part. The part numbers (1–16) refer to the circled identifying labels on *Figs. 2* and *3*.

1. Top	12″ × 13½″
2. Bottom	12″ × 13½″
3. Baffle	11 × 10⅝
4. Rear horn brace	8″ × 17″ (see text)
5. Front horn brace	10″ × 9″ (see text)
6. Rear horn plate	11″ × 15¾″
7. Front horn plate	11″ × 11¼″
8. Front panel	11″ × 11¼″
9. Sides	23½″ × 13½″
10. Upper reflector	10 ¹⁵⁄₁₆″, 4″ PVC pipe (quartered lengthwise)
11. Lower reflector	10 ¹⁵⁄₁₆″ × 8¼″ (4″ or 6″ plastic drainage pipe)
12. Lower reflector braces	5″ × 5″
13. Duct	3″ PVC pipe, 4″ long
14. Back	12″ × 24½″
15. Throat wings	6″ × 3¼″
16. Reflector cap	6″ × 3½″

PS-8 driver available from Carvin, (800) 854-2235, at approximately $50. Motorola drivers available from various sources, including Radio Shack.

FIRST STEPS

As in all the Snails, the first step is to make the patterns for the horn braces. Once you have cut the original pattern from plywood, you can trace and cut copies from ¾″ plywood or ¾″ or 5⁄4″ stock. If you are making two or more cabinets, remember to trace and cut enough braces before you discard the pattern.

The patterns for the rear braces and back brace are cut from a piece of plywood measuring at least 8″ × 17″. Parallel to that piece's longer edge, draw a line ½″ from the right edge, and on it mark a point 2½″ from the end, followed by points every 2″ down the line, for a total of nine (*Fig. 2*). Through all nine points, draw lines perpendicular to the edge. On these lines, mark off the following distances to the left of the original nine points, starting with the first point you marked: 2⅛″, 2 5⁄16″, 2½″, 2¾″, 3 1⁄16″, 3⅜″, 3¾″, 4 5⁄16″ and 5″.

Into each of those points drive a 4d or 6d nail. Rip a sliver of ⅜″ plywood about ½″ wide and at least 18″ long, and pull that sliver tight along the inside (the left side) of the curve formed by the nails, securing it in place with additional nails (*Photo 2*). Now trace lines along both sides of the sliver, and remove the sliver and the nails. With either a bandsaw or jigsaw, cut the pattern along both lines. The part containing the original measuring points is the back brace. Cut away the selvage from this above and below the first and last lines

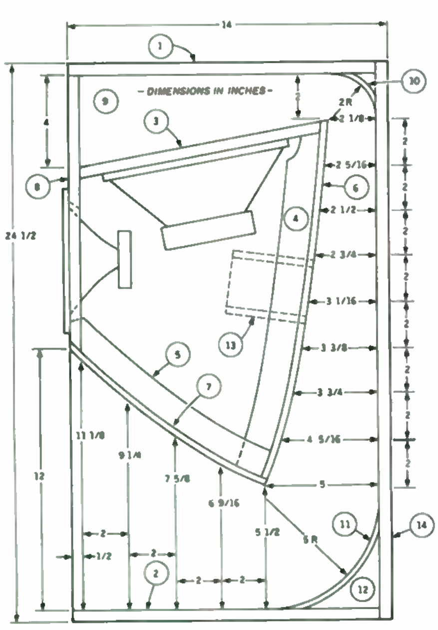


FIGURE 2: Side view of MiniSnail.

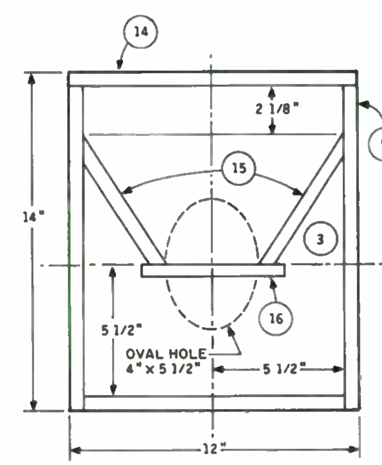


FIGURE 3: Top view.

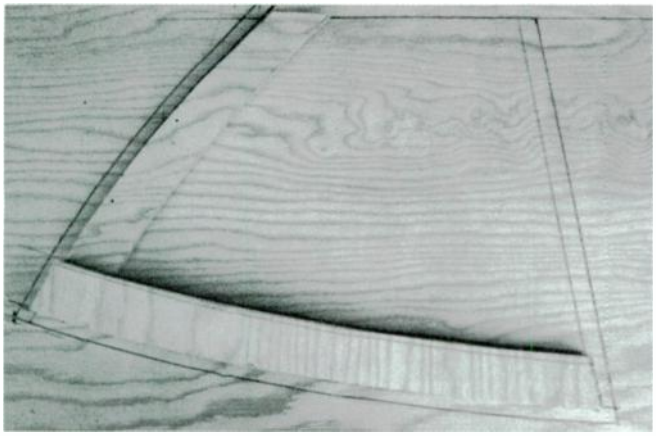


PHOTO 3: Tracing the brace locations.

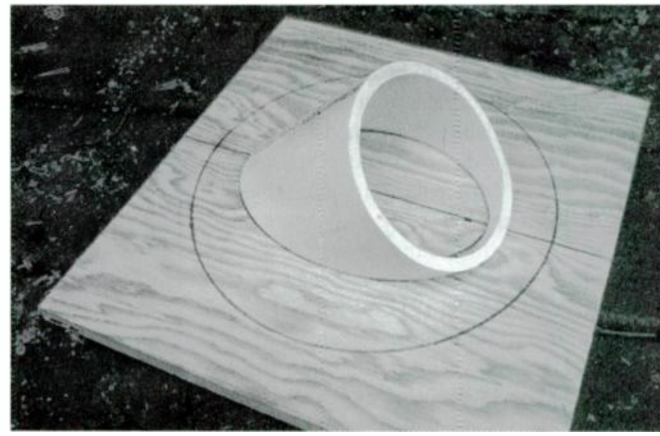


PHOTO 5: Tracing the baffle hole.

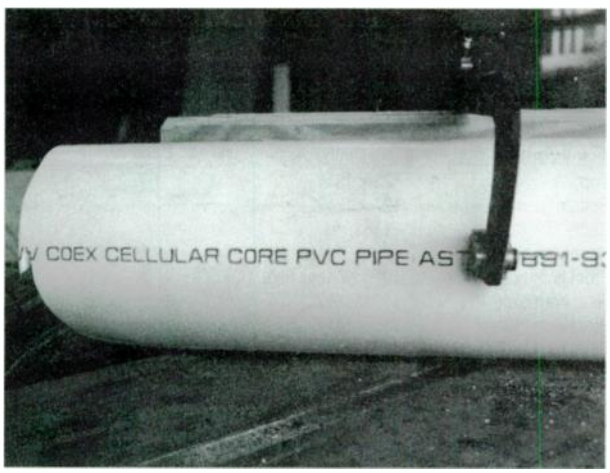


PHOTO 4: Cutting the PVC pipe.

marking the tracing points, and make a duplicate of this pattern, one for each cabinet you intend to build.

Label the remaining portion of this pattern piece "rear horn brace," and put it aside. The pattern for "front horn brace" is cut from a piece of ½″ plywood measuring 10″ × 10″. As before, starting ½″ from the right edge, mark a point, followed by other points every 2″, for a total of five, with one additional point ½″ from the fifth mark. Again, through each point draw lines at right angles to the edge. Parallel to the board edge and ½″ in, draw another line. From that line, sequentially from right to left, mark points on the perpendicular lines at 0″, 1¹⁄₁₆″, 2 ⅛″, 3¾″, 5⅝″, and 6½″.

Drive nails into the points, again securing the tracing sliver against the inside of the curve of nails, tracing the sliver's curve, and cutting the two pieces thus formed. The "inner piece" is the front horn brace, the outer a cast-off you can use to trace parts locations on the cabinet sides. On the front horn-brace, draw a line 1½″ inside the first, marking the actual brace pattern, which you now trim to width. Trim the rear horn brace in similar fashion.

POSITIONING THE PARTS

The parts list (*Table 1*) identifies by number (1–16) the parts discussed below and shown on *Figs. 2* and *3*.

Mark one cabinet side with the positions of all parts. From the top rear of the panel, mark a point 2″ down and 2⅛″ in from the back, and mark another point 5½″ up from the bottom and 5″ in from the back. Mark a further point on the front edge of the panel, 11″ down from the top. These points delineate the locations of the rear and front horn plates, and you use them to place the pattern pieces on the panel and trace their positions.

There will be a ⅜″ gap between the lines marked by the back brace and rear horn brace; likewise between the lines traced from the front horn brace and the cast-off piece. From the horn plate to the cabinet's top edge, draw a vertical line ½″ in from the front edge. This denotes the position of the cabinet front. Mark a point on that line 4″ down from the top of the panel, and draw a line from that point to the point at the topmost end of horn plate 1. This denotes the top edge of the baffle; a line parallel to it and ½″ down marks the baffle's lower edge. Now mark the second side of the cabinet in mirror image to the first.

Place the horn braces on the side piece, marking their intersections with the baffle, the front piece, and each other, and trimming off the excess (*Photo 3*). Each cabinet will have three sets of braces, one set having an overlapping joint where the braces meet, and the other two having butt joints. Keeping this in mind, make from the patterns as many braces as you will need (if you anticipate building more cabinets in the future, as I do, save the original braces).

Cut a section of 4″ PVC pipe (the heavy, ¼″-wall type) about 6″ long, with one end cut off at a 45° angle. You can do this best on a table saw, clamping the pipe to a miter gauge with an extended plywood fence (*Photo 4*). Cut the baffle to size, tracing the driver location on it. Using the cut section of PVC, trace an ellipse, centered on the driver location, on the inside of the PVC (*Photo 5*). Cut out this oval, and chamfer the edges of the hole to a ¼″ radius using a router with a ¼″ round bit.

ACCESS-PANEL PLACEMENT

Choose one of the cabinet's sides as the site of the "side door" access panel, whose locaton is bordered by the inner edges of the baffle and the front, and by a line drawn midway between the inner and outer edges of the baffle braces. Cut out this panel with a jig-saw, and to each other attach

PHOTO 6: Beginning construction.

Swans M2 kit

High sonic resolution
Loudspeaker
with room friendly performance

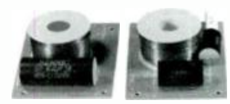
The Swans M2 is a floorstanding model that features several technological achievements and sound quality distinctions.
The speaker system is a two-way bass-reflex design with MTM driver configuration. The front baffle is very narrow with rounded edges to reduce cabinet diffraction for better clarity and imaging. The internal panels and corner reinforcement bars substantially suppress unwanted cabinet vibrations. The bottom part of the cabinet is sealed and can be filled with sand or lead shot for better stability and further performance improvement. A port is mounted on the rear panel.
The drivers used in the Swans M2 represent a new high performance design from Hi-Vi Research. The 5-inch paper/Kevlar cone bass-midrange has a rubber surround, cast aluminum frame and a magnetically shielded motor system. This driver utilizes a central phase plug to avoid air compression, improving frequency response and dispersion. The extremely rigid cone is hand coated with a special dampening compound to further maximize its performance. The cone is coupled to a selected grade rubber surround, this provides break-up free operation and very low distortion even at high power levels. These key features greatly contribute to the Swans M2's clear transparent sound and effortless dynamic performance. Swans M2 delivers amazing bass without runing in "doubling" or Doppler distortion problems.
The tweeter is a high-tech planar isodynamic design that employs Neodymium magnets and extremely light Kapton® film, with flat aluminum conductors.
The vibrating element of the tweeter is almost weightless in comparison to a conventional dome driver. This unit provides an immediate and precise response to any transients in original signal, and gives the Swans M2 an exceptional ability to reveal the true dynamics of instruments with a complex high frequency spectrum.
The crossover is a second order Linkwitz-Riley type resulting in an in-phase connection of the drive units. The crossover frequency between the two drivers is 3.3 kHz and only high quality polypropylene capacitors are used. Each filter has it's own dedicated board mounted on a special rubber interface to reduce vibrations and microphonic phenomenon. The filter boards are spaced inside the loudspeaker with the inductors positioned at right angles to minimize the interaction.
Swans M2 provide very even acoustic power dispersion. The important horizontal early reflections that create spatial impression and add to the overall presentation have the same even spectral balance as the direct sound, these are crucial features of a good loudspeaker.
On the contrary, the vertical dispersion is well controlled in the midrange and high frequency domain in a 15° arc symmetrically to the reference axis. While 15° create adequate room for adjusting a listening position, the floor and ceiling reflections are well down in amplitude. This feature greatly contributes to the clarity of sound and imaging of the system.
Swans M2 kit includes:
- 4x F5 paper/Kevlar bass-midrange drivers,
- 2x RT1C isodynamic tweeters with sealing gaskets,
- 2x dedicated tweeter crossovers,
- 2x dedicated bass-midrange crossovers,
- two ports and two Swans logos,
- two pairs of heavy-duty gold plated terminals.

Cabinets are not included.
For those who are interested in a home theater set up, the instructions and parts for correspondent central channel speaker are available.
The drawings of the cabinet shown here represent general dimensions required for optimum bass performance. Rounded corners are advisable as they improve imaging and clarity. Actual finish and appearance is a matter of personal taste. The system should be installed on adjustable spikes and slightly tilted back to aim tweeter axis at listening position.
Retail price: US$ 530.00 (delivered)
Delivery in US within 4-6 days.
Warranty 3 years, 30 days money back guarantee.

SWANS
SPEAKER SYSTEMS
The step beyond the limits

RT1C Tweeter

F5 Bass-midrange



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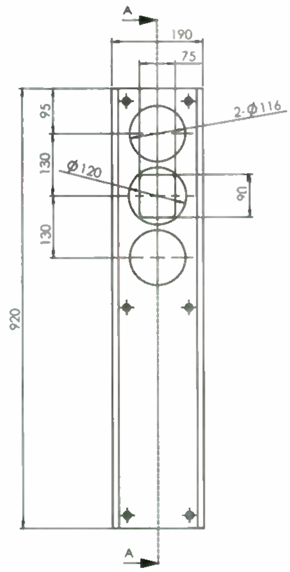

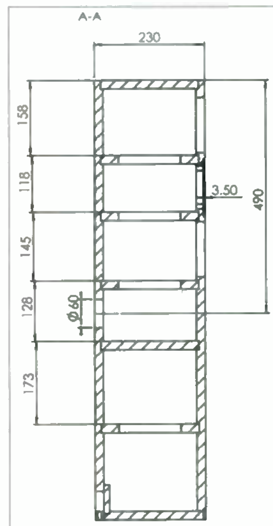

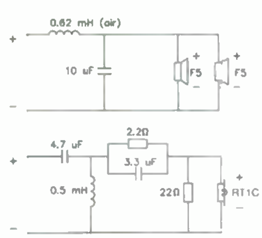

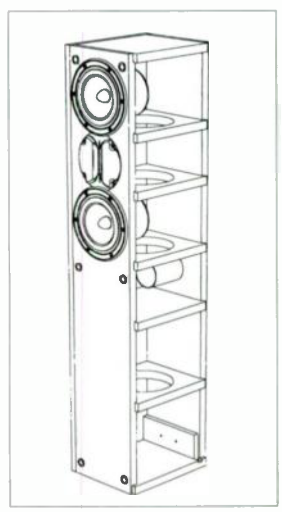

SPECIFICATIONS

Frequency response (1m,half space)	53Hz-20kHz ±2.5dB
Sensitivity,1W/1m (100Hz-8kHz averaged)	87.5 dB
Nominal impedance	4 ohms
Harmonic distortion At 90dB SPL, 100Hz-10kHz, 1m	THD less than 1%
Power handling	80W nominal, 150W music
Dimensions,HxWxD (without spikes)	920x190x230 mm 36 1/4x7 1/2x9 inches

Amplifier requirements:
30W recommended minimum.

ACOUSTIC TECHNOLOGY INTERNATIONAL INC.
15 WEST PEARCE STREET UNIT 2&3, RICHMOND HILL
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the baffle, the front, and one set of horn braces (*Photo 6*).

Attach the second side to this assembly. Cut the two throat wings somewhat higher than necessary (you'll trim them later). One end of each of these requires a 65° angle, which is difficult to cut, since table saws adjust only up to 45°; so cut at 45° and belt-sand some material off to achieve a reasonable fit. Install them on the baffle, using liberal amounts of adhesive to fill the joints between them and the cabinet sides (*Fig. 3*).

You can cut the 4" PVC throat reflector most easily by screwing a piece of PVC whose end you have trimmed to 45° directly to a panel-cutting jig on a table saw (*Photo 7*). When cut, use it to trace the oval cut on the reflector cap. You cannot screw the reflector in place, so use a hot-melt glue gun to attach it to the baffle/wing assembly, and the reflector cap to it, using the hot-melt as caulking to fill all gaps.

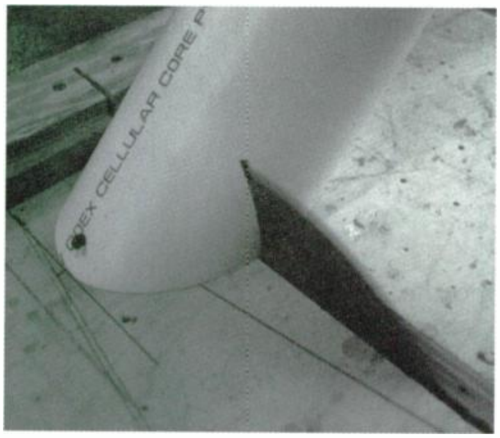


PHOTO 7: Cutting the throat reflector.

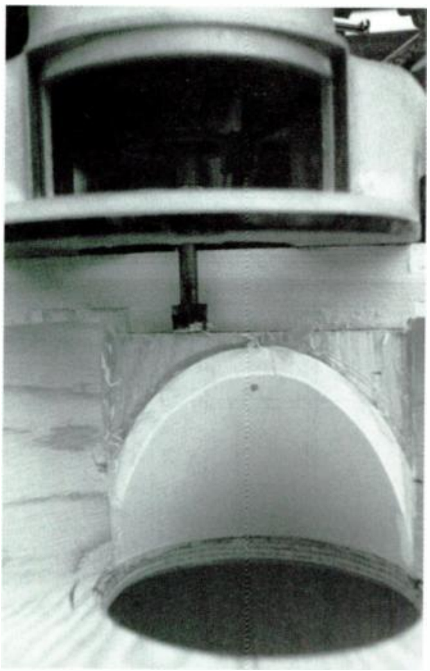


PHOTO 8: Trimming the reflector-cap height.

Place a straight piece of ¾" plywood or 5/4" stock across the cabinet top, and use a router and mortising bit to level the tops of the throat wings and reflector cap (*Photo 8*). Attach a set of horn braces to the second side, and temporarily set the driver in place. Mark the remaining horn brace 1 to trim off enough material so it will clear the driver frame, and then attach the brace set to the baffle, to the front, and to each other with an overlapping joint (*Photo 9*).

Now screw and glue the rear horn plate to the assembly, using long clamps to pull the pieces into place. To give the front horn plate something substantial to attach to at either end, fasten scraps of plywood or stock, about 1½" wide, to both the lower flange of the rear horn plate and the front, sanded to fit the contour of the horn braces (*Photo 10*). Attach the front horn plate to the assembly. After the glue has set, use a belt or disc sander to true and chamfer all the joints between the horn plates themselves, and between them and the front and the baffle.

INSTALLING THE DRIVER

Cut strips of 1¼"-wide plywood or stock to size to complete the flange on the access door, already partially formed by the horn braces. Attach these strips to the front and the baffle, with the joints between all the flange pieces caulked with glue for a tight seal. Drill a ¼" hole through the rear horn plate and run the speaker wire through it, caulking it with hot-melt glue.

Place the access door in position, drilling pilots for screws every four inches or so. Install the driver, wire it, rim the access flange with weather stripping, and screw the access panel in place. Now feed the cabinet with a fairly strong low-frequency signal, listening and feeling for leaks in the cabinet, and sealing them with hot-melt glue. When you've verified that the cabinet is airtight, remove the access door and driver.

Now attach the top and bottom pieces, followed by the upper and lower horn reflectors. The upper reflector is cut from 4" PVC, which you can easily quarter on a table saw by temporarily screwing the pipe to a piece of plywood, using the rip fence on the saw and eliminating the natural tendency of the PVC to roll around while being cut. Be aware that the PVC has a tendency to close on the saw blade with the initial cut (*Photo 11*).

After you cut it to size, glue the upper reflector to the top, using hot-melt glue. Cut the lower reflector from a piece of ⅛"-wall soft plastic drainage pipe, either 4" or 6" diameter, 11" long by 8¼" measured around the pipe. You can easily soften the plastic to accommodate the required 5" radius by placing it in a large pot of boiling water for a few minutes. Cut the lower reflector braces from either plywood or stock, and glue and screw them to the cabinet bottom.

PHOTO 9: Ready to attach the horn plates.

PHOTO 10: Rear horn plate installed.

Reader Service #26 →

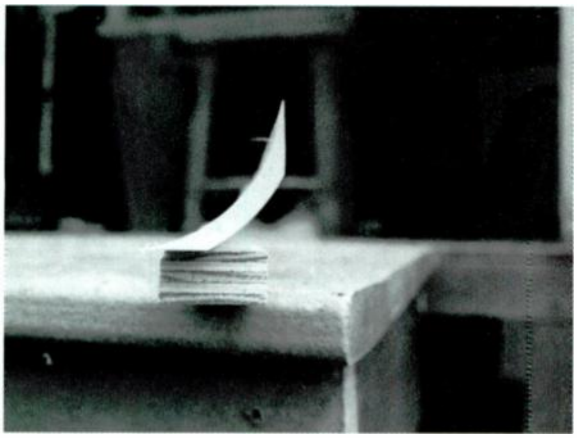


PHOTO 11: 4″ PVC quartered lengthwise.

PHOTO 12: Ready to attach the back.

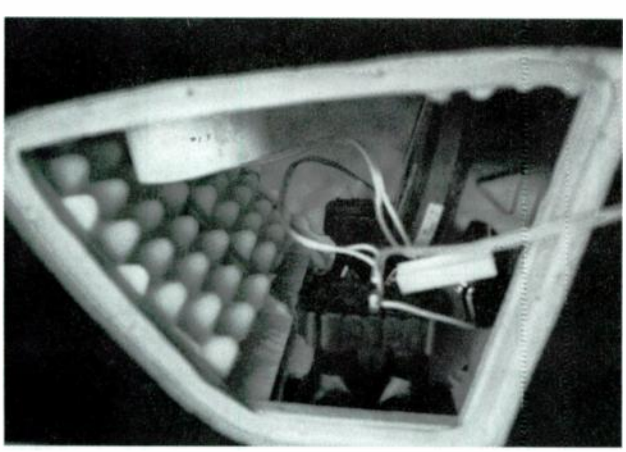


PHOTO 13: A tight squeeze.

Glue the reflector to the cabinet bottom with hot-melt glue, augmented with screws every two inches. (I've found that 7⁄16″ steel framing screws, available from most lumberyards, are ideal for this application.) Likewise, glue and screw the reflector to the braces. To do this, place a piece of scrap plywood across the cabinet back, and use clamps to pull the plastic in place against it, holding it securely for about half an hour until the glue sets. (This is longer than usual, since the heated plastic will keep the glue from setting at a normal rate.)

BRACING THE BACK

Next, attach the back brace, first cutting away any areas that might hit either the top or bottom reflectors. Glue it in place with hot-melt, using a board across the back to ensure proper alignment. At one side of the back brace, cut into the horn plate a 3½″ hole for the duct, which is 3″ PVC pipe, cut 4″ long and secured in place with hot-melt glue (*Photo 12*). Fasten the back in place, using plenty of adhesive to fill all possible gaps, as well as 7⁄16″ screws to pull the lower reflector tightly against the back from the inside.

Cut holes into the front panel for the tweeters (*Fig. 4*), and a hole on the side (or through the access door) for the side-firing tweeter. Assuming you are making two cabinets, one for each side of the stage, "mirror-image" the side-firing tweeters to cover the area directly in front of the stage not covered by the main tweeters. Also decide on a location for your jack of choice and drill or cut the appropriately sized hole for it.

As previously noted, the tweeters have a low-end cutoff at 3.5kHz. You may wish to use a different model, Motorola's KSN1165A, which works down to 1.8kHz. If

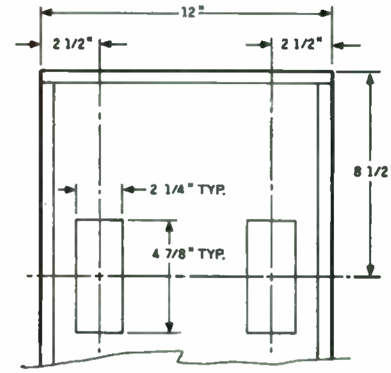


FIGURE 4: Front view.

so, you will need five per cabinet. since the "twin bullets" used on my original are actually two drive units on a single frame; you will need four KSN1165As to replace two "twin bullets."

Apply the finish at this point, be it paint, Formica®, carpet, or whatever. Line all the walls of the driver chamber with acoustic foam, and install the drivers (because of its light weight, I used screws on the woofer rather than tee-nuts and bolts). Carpet-covering the cabinet will act as a gasket for the tweeters; otherwise, you must gasket them with weatherstripping for an airtight seal. Solder the speaker wire to the jack and to the woofer, and wire the tweeters together in daisy-chain fashion, making sure to maintain the polarities.

The drivers are close enough together inside the chamber so that you may solder the 4Ω resistor directly to the positive lead of the woofer on one end and to the positive lead of the tweeters on the other; connect the negative leads with a piece of wire (*Photo 13*).

Finally, attach handles and protective corners as desired, and a stand "tophat" bracket if you plan on stand-mounting your cabinets (*Photo 14*), and screw the access panel in place.

SPEAKER PRESENCE

When you set up your system, you'll find these speakers have much lower-midrange presence than traditional cabinets that strive for flat response. Cut EQ at 250–300Hz to eliminate "muddy" response, and boost at

PHOTO 14: Finished MiniSnail on tripod stand.

100Hz for a more "chesty" bass on male vocals. Below 80Hz, pull the sliders all the way down, and engage the high-pass filters on the mike channels (if your board has them).

Likewise, if your power amp has a high-pass filter, usually around 80–100Hz, engage it to avoid sending low-bass program to the speakers while conserving amp headroom. The cabinets' uppermidrange "dip" from 1.5–3kHz should help control feedback, while the very high sensitivity above 5kHz means you'll need far less treble boost to get a clean and "airy" high end.

Aim the "twin bullets" at the farthest end of the room, and the side-mounted tweeter will eliminate the usual high-frequency "dead spot" in front of the band. Finally, at the end of the night, enjoy the fact that you have only half the usual weight to deal with when lugging your cabs out to the truck.

SAVING AGING FOAM CONE SURROUNDS

By David L. Kelley and P.-T. Ho

Long service eventually leads to cone-surround deterioration, often referred to as foam rot. The material shows such changes as loss of its original color and surface sheen, and a weak papery feel to the touch when you apply light pressure. In its final stages of deterioration, small holes begin to appear, followed by tearing that commonly occurs at the cone and frame junctions.

Further use, particularly under large cone-excursion conditions, renders the driver useless, since the surround cannot perform any of its design functions. It is a painful experience to remove the speaker grille and discover your driver's surround is a mass of rents and fragments (*Photo 1*).

PHOTO 1: The original Dynaudio 21W54, a superb 8″ driver, rendered useless from foam rot. Its surround had completely disintegrated as a result of long use and exposure to high heat levels. Every piece of the surround had broken free at the cone and frame junctions and collected at the bottom of the grille frame.

FINDING REPLACEMENTS

For many drivers, you can purchase new surrounds to replace the originals, which works well if the replacement has the proper dimensions for the driver and its original surround, and if it is carefully installed. Advertisements for replacement sources appear in audio periodicals such as *Stereo Review* and *Audio*. For those interested, we recommend "How to Replace Speaker Surrounds," by Dennis Eichenberg (*Stereo Review*, June 1996, p. 69), which also identifies several sources for surround replacement kits.

There are, however, outstanding older drivers for which these sources do not appear to carry exact replacements. Moreover, some manufacturers will not supply them because of a policy not to maintain parts supplies after a driver has been replaced by a newer design, even if many of the original's dimensions have been retained.

For example, the original Dynaudio 30W54 is a doped-pulp cone woofer with a foam, positive half-roll surround attached to the front face of the cone rim. Dynaudio gives its effective cone (piston) area (S_d) as 400cm^2, placing it just a little less than halfway between typical 10″ and 12″ driver-cone areas. Attempting to make do with an inexact replacement surround can not only necessitate meticulous trimming to make it fit the driver frame, but also face you with the difficulty of dealing with an improper overlap of the cone rim.

As an alternative to direct replacement, we offer a simple preventative method that essentially saves the damaged surround for continued use. The procedure this article describes aims at completely coating the old surround so that its functions are transferred to the new flexible covering.

DRIVER AND MATERIALS

The driver we used in this investigation is a Dynaudio 30W54 in its PHA version, that is, a polymer/oxide formulation. It, like the original mentioned previously, has a foam, positive half-roll surround, but in this case attached to the rear face of the cone rim. Its frame appears to be exactly the same as that used in the original 30W54, each designed for front mounting. The S_d of 400cm^2 is also retained. It has had normal in-home use for well over ten years and is clearly in its dotage.

To superficial observation while still mounted on its enclosure, it did not appear to be torn, but upon removal, we found three tears at the cone-rim junction, the largest being about 1¼″ long, and hence requiring immediate treatment. This driver is still available, with a replacement cost in excess of $240.

A material suitable for surround coating should offer low mass, durable flexibility, adequate adhesion, minimal shrinkage, easy application, and extremely low cost. The material we chose is an acrylic latex caulk containing silicone, named ALEX PLUS and manufactured by DAP in Dayton, Ohio. It comes in a number of colors, including our choice that applies white and dries clear. Being white upon application is useful in achieving uniform coatings, and drying clear allows you to retain the approximate original color, while the surface takes on an attractive shine. It adheres well to the foam, but not to the plastic cone. The amount we used for a single application, front and rear, is very small. The manufacturer claims a durability of 35 years for its recommended uses.

APPLICATION INSTRUCTIONS

The procedures are as follows:

1. Remove the driver from its enclosure.
2. With a soft brush, dust off the surround's front and rear surfaces. If tears are evident, use great care!
3. Place the driver on its magnet assembly, cone facing upward.
4. If the surround is torn, very carefully reach a finger under the torn area and press

PHOTO 2: The materials used are few. The brushes are artists brushes by Duncan, but this quality is not required. The plastic plugs are inserted into the frame's mounting holes to elevate the frame and surround when the driver is facing down.

gently upward to inspect the tear's length and resistance to further tearing. Mark the tear's location on the driver's frame, front and rear.

5. Select a small brush with a long shaft and short, beveled, densely packed fibers. Coating the rear surface of the surround requires reaching into hard-to-reach spaces (*Photo* 2).

6. Eject a small amount of caulk into a small, clear container and add a small amount of water to improve the flow when applied to the surround. Make sure it is evenly mixed with the caulk.

7. Load the brush and gently apply the caulk along the tear, going beyond its ends. Apply upward pressure to the underside of the tear to open it, and brush the caulk into the opening. For a long tear, stroke your finger gently along it to move the opening along its length. You will feel the caulk on your finger as the material passes through the tear. Now, smooth the caulk over the entire length of the tear's upper surface and beyond. Turn the driver over, and apply the caulk over and beyond the tear from the rear. If more than one tear exists, repeat the process for each.

Replace the driver on its magnet and allow it to dry overnight. Note that the 30W54 has a cast magnesium frame with no front-attached mounting gasket. In such a case, you must elevate the frame to avoid the coated surface touching the table top. Plastic plug inserts *(Photo 2)*, which often come attached to drivers during shipment, can be used for this purpose. If these are not available, wooden lifts taped to the table top will do as well.

8. Examine the tears the next day, applying pressure over the treated areas, to see whether the tear is secured. If all is well, begin the first complete application to both surfaces of the surround. Start application in one direction, brushing for approximately two to three inches; then return to the starting point and repeat in the opposite direction. Since transition from white to clear in thin layers occurs rapidly, especially in hot, dry climates, the alternating applications avoid unwanted overlapping.

Continue in this manner until your alternate applications meet on the other side of the driver, diametrically opposite your starting point. Brush with strokes that provide a smooth surface. Now turn the driver over and apply a coat to the rear surface. The driver's basket ribs make this application more difficult, but be careful and make sure you cover the entire surround surface. If you should get some on the cone, carefully use a damp cloth to wipe it clean. Again, allow it to dry completely overnight.

PHOTO 3: The finished product. Note the lack of mounting gasket. Also see the sheen of the coated surround.

TABLE 1

MECHANICAL PARAMETERS

	BEFORE COATING	AFTER FIRST COATING	AFTER SECOND COATING
Mass (g)	51.1	50.8	51.3
C_m (mm/Newton)	1.56	1.73	1.75
Mechanical resistance (kg/s)	2.30	2.22	2.34

Next day, hold the driver up to a bright light source and look for the torn locations from the rear. Because the coatings are now clear, light will pass through the original tears, so you can easily see them. Repeat the coating process once again and allow for overnight drying. We have used as many as three coatings with satisfactory outcomes, but prefer just two. The choice is yours.

9. When the procedure is completed, place the driver on its magnet and push the cone downward around its dust cap if it has one (*Photo 3*). As the cone moves up and down, listen carefully for signs of coil rubbing. This is most unlikely, because you have neither touched the spider nor displaced the surround. If you have a signal generator, set its frequency to 15Hz and increase the gain to produce a near-maximum cone excursion. *Photo 4* shows our repaired driver during its 12-hour endurance test. All was well.

MEASUREMENT PROTOCOL AND FINDINGS

We were interested in how the coatings might change the driver's fundamental parameters. Those most likely to be affected are mechanical ones: moving mass, compliance, and mechanical resistance. We made standard measurements of the electrical impedance of the system using the delta-mass method.

We used LinearX's Loudspeaker Measurement System (LMS) and Loudspeaker Enclosure Analysis Program (LEAP). LMS provided a 1kHz–10Hz sweep to measure electrical impedance. We made three series of measurements, one before the first coating, another 24 hours after the first coating, and a final series 24 hours after the second coating.

Each series included the addition of 15g, 20g, and 35g of mass. We determined that each coating weighed approximately 3g, and the amount of added water was less than 1g.

We exported measured impedance data from LMS to LEAP, whose single-curve impedance-fitting program calculated the free-air resonance frequencies, mechanical Q (Q_{ms}), and electrical Q (Q_{es}). From the free-air resonance frequencies (f_s) corresponding to different amounts of added mass, we calculated the driver's moving mass (M) before coating, after the first coating, and after the second. We found that the values of the moving mass—calculated using the three different amounts of added mass—agree within 3% or less in all cases. We then calculated compliance C_m from $1/[M(2\pi f_s)^2]$. Finally, we calculated mechanical resistance from $2\pi f_s M/Q_{ms}$.

The values of these parameters are listed in *Table 1*. You can see that—within the accuracy of a few percent of the measurement and fitting methods—no change in the driver parameters took place between the two applications of caulk. A small but noticeable increase in the compliance, and a corresponding decrease in the free-air resonance occurred after the first application, which is very likely the result of additional, but undetected, tearing of the surround during the application process.

For the loudspeaker designer, it is not the mechanical parameters, but the related Thiele/Small parameters which are of interest. The latter are tabulated in *Table 2*. Again, no significant change occurs be-

TABLE 2

RELATED THIELE/SMALL PARAMETERS

	BEFORE COATING	AFTER FIRST COATING	AFTER SECOND COATING
f_s (Hz)	17.8	17.0	16.8
V_{as} (ltr)	383	424	429
Q_{ts}	0.315	0.315	0.311

PHOTO 4: It is important to test the surround's performance under moderate-term and large-excursion operation after the coats have dried completely. By attaching a small white dot to the surround at its cone junction, you can see the approximate pk-pk excursion as the dot moves up and down. The excursion was reduced for this photo to avoid image smearing.

tween the two coatings, and the most noticeable changes after the first coating were in V_{as} (equivalent to C_m) and f_s. Even the change in V_{as} in this case is only 10%, and the corresponding change in f_s is 5%, unlikely to call for a different box design because of the coating process.

At higher frequencies, cone breakup occurs. The surround serves as the mechanical load impedance for vibrations emanating from the cone center. Breakup resonances are also affected by the cone surround, usually showing up on the electrical impedance sweep as well. We could not study cone breakup in any quantitative way, and can only note that the electrical impedance above the bass resonance peak has not been changed by the coating process—again, within measurement error.

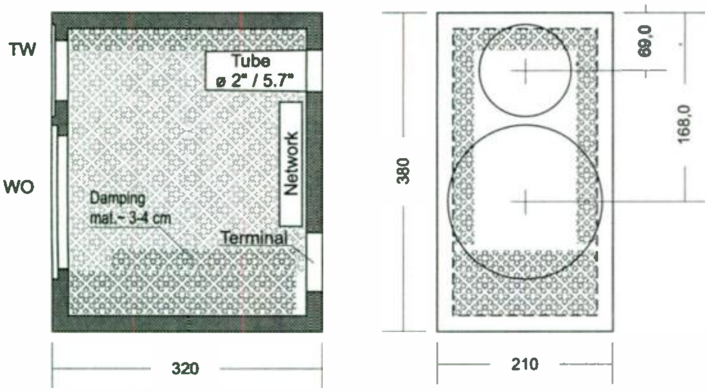
CONCLUSION

We suggest that you not apply coatings to drivers whose foam surrounds are performing properly and are not showing signs of deterioration due to age and use. It is always a sound idea to inspect your drivers regularly, particularly woofers, because of their power-handling requirements and large excursions. At present, there appears to be a trend for ever-increasing loudness levels. Hearing damage aside, older drivers can easily be overdriven in this loudness quest.

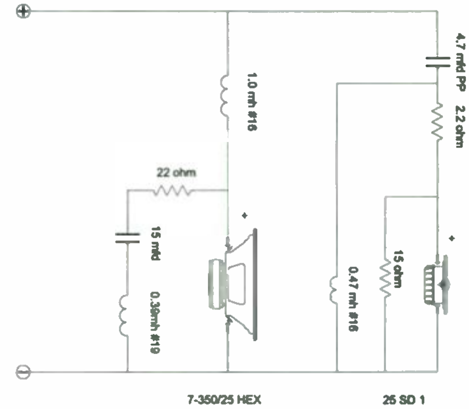
We must make it clear that we have not studied coated surrounds for years and years, so we cannot describe long-term outcomes. However, it is our belief that those of you who would like to save older drivers in their inevitable declining years really have little to lose by trying these procedures.

Caulk formulations are available in bewildering variety, each suited for specific tasks. We have used only one of these, and therefore cannot comment on any of the other types and brands. However, we doubt that any of these identify cone-surround coating as a "recommended" use.

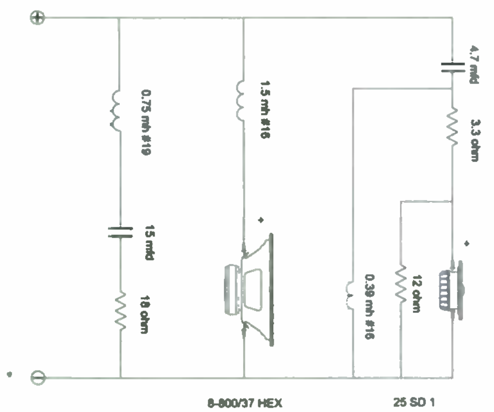
ETON Kits

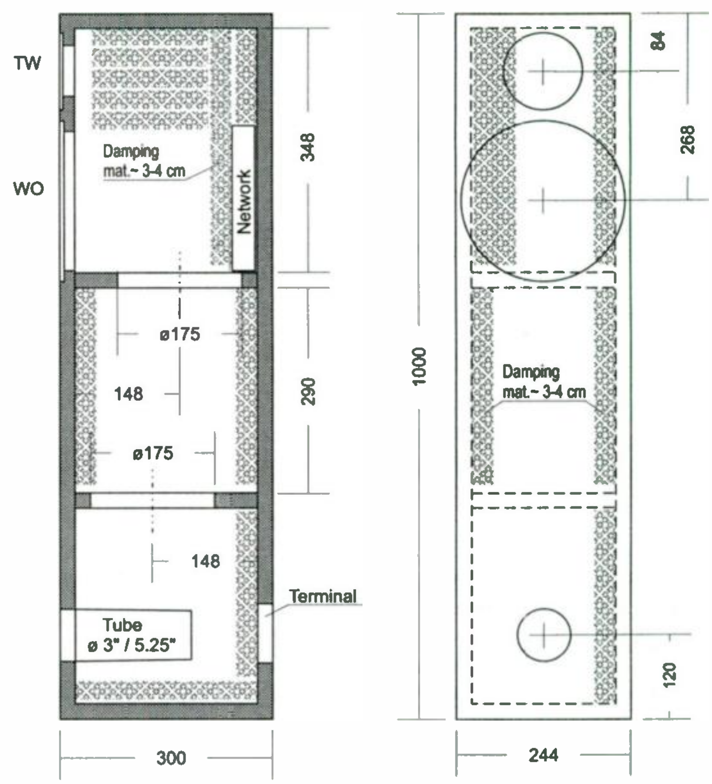


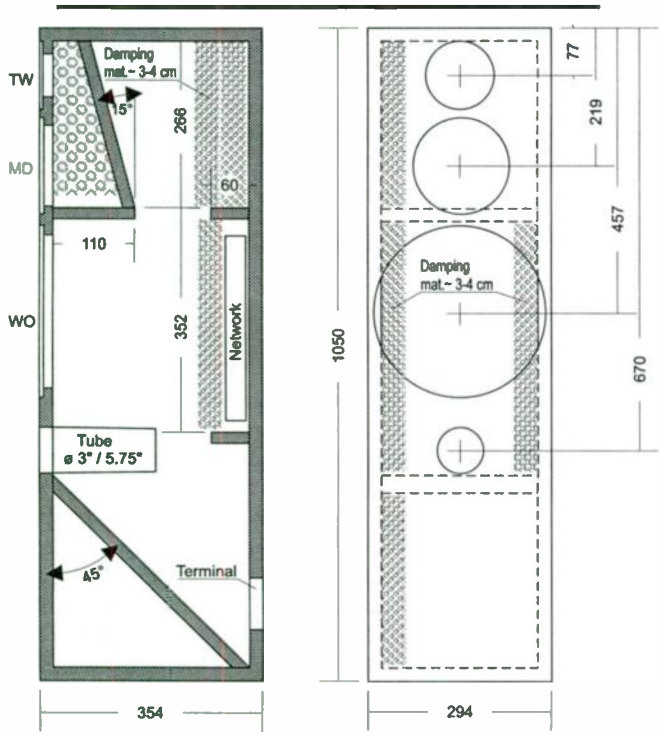
Eton 7.2 Kit (7-350, 25SD-1)



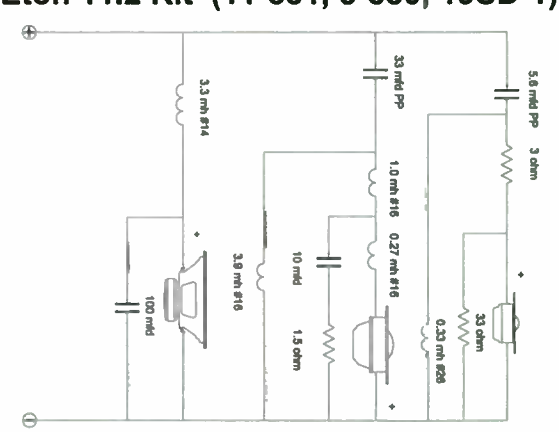
Eton 8.1 Kit (8-800, 25SD-1)







Eton 11.2 Kit (11-581, 5-880, 19SD-1)



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MODELING FOR DESIGNING PASSIVE CROSSOVERS

By G.R. Koonce

The classic approach to passive crossover (CO) development was to establish the acceptable frequency range for each driver by listening, testing, or guessing. You corrected the driver impedances to approximate fixed resistors by careful Zobel design. Then, based on these resistor values, you designed a CO of a certain order, a certain topology, and at selected frequencies based on your experience and prejudices.

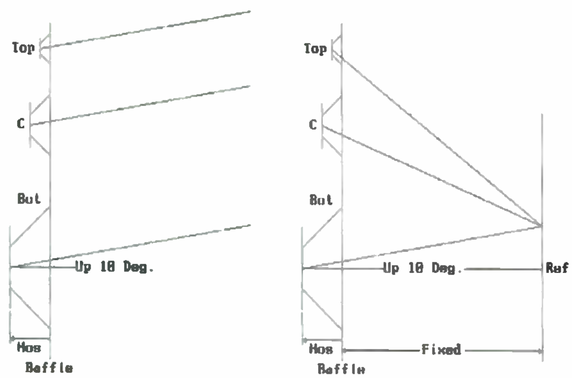

FIGURE 1: Summation at infinity (left) versus summation at a fixed distance (right).

While listening to the system with this CO, you then "fixed" the design by ear. This involved modifying the CO from the sound the system was producing to a sound you liked, and the process could consume a lot of time. Sometimes you would even give up, design a new CO, and start the fixing all over again.

WHICH ORDER?

You learned over the years what type of COs worked best. The first-order was great on paper, but required drivers with very wide, flat responses, and its performance could easily be destroyed by the drivers' physical spacing.

The second-order could prove to be very "dangerous." In theory, it had a null on-axis when used with drivers wired with the same polarity, but sometimes it sounded better with them wired that way. The second-order sometimes required lots of adjustment with techniques to fix the sound, such as moving the low-pass frequency up or down relative to the high-pass to produce underlap or overlap. Sometimes it seemed you just could not get a set of drivers to work together with a second-order CO.

I learned to prefer the third-order, since it appeared to work more in accord with what the books said, without a lot of fixing. Many designers selected the fourth-order network, but I always believed the parts count was just too high, and that a third-order electrical CO had a better chance of approaching a fourth-order acoustic CO response.

Then the computer-based quasianechoic test sets arrived, and with them anyone could try to test drivers. Once you got the test set, the only requirement was to learn to make good, repeatable tests. Without taking advantage of work already done by others before you, that learning exercise could take a couple of years.

Once you had learned to do good tests, a great truth of passive CO design became clear: no classic CO design used with real drivers, spaced as drivers normally are, ever tests as the book predicts! Testing revealed that second-order COs did not always show the null with all drivers wired with the same polarity; sometimes nulls appeared when drivers were inverted.

The books indicated that odd-order COs should have the main axis (axis of maximum response) in a certain direction, and that even-order COs should have it in a different direction, but when you tested the main axis, it was not where predicted. The only systems you found in testing that had the main axis consistently at zero degrees—i.e., perpendicular to the plane of the drivers—were those using a symmetrical driver layout. The classic CO designs just did not test as the books predicted when used with real drivers spaced over the front-panel area.

OTHER DESIGN APPROACHES

If the "classic" CO design approach is not producing what you had expected in testing, why not just develop the CO via testing? This is a valid approach, but it involves designing a bunch of COs, gathering up all the components to implement each design, and then testing the system with each CO design at a variety of angles. You learn that this approach can take weeks, and after the enthusiasm of developing your first system this way, you start to lose interest and skimp on the number of CO designs you try with each system.

Suppose you stick with it and finally develop a CO that tests in your system with ruler-flat response on the axis on which you intend to listen. Will the system sound good? The answer is maybe. First, you are probably testing it at about half the distance you will use when listening to it. Your room simply will not support quasianechoic testing at the distance you normally use in listening.

Will the system perform the same at your normal listening distance as it tests at the shorter distance? With your present limitations, testing can't answer this question. The second reason a good CO-on-listening-axis test does not ensure a good-sounding system is that the testing shows the quasianechoic response on one specific axis, not the total (power) response of the system in your listening area as perceived by your ears. Maybe the ruler-flat listening-axis response doesn't in all cases produce the best-sounding system.

CO MODELING

So if home testing can't ensure the best CO design, what then? There is CO modeling. In this approach, you feed a computer program the driver-input impedance and measured frequency response as data files, plus the physical location of the drivers, and the program models the quasianechoic acoustic response of the system at some point in space. You can enter the CO component values into the program, or an optimizer type of modeling program can compute them, and the program will plot the acoustic response of the system.

One of the main reasons such a modeling program is better than testing is that it is a lot faster. You can try in hours more CO ideas than you could evaluate by testing in weeks,

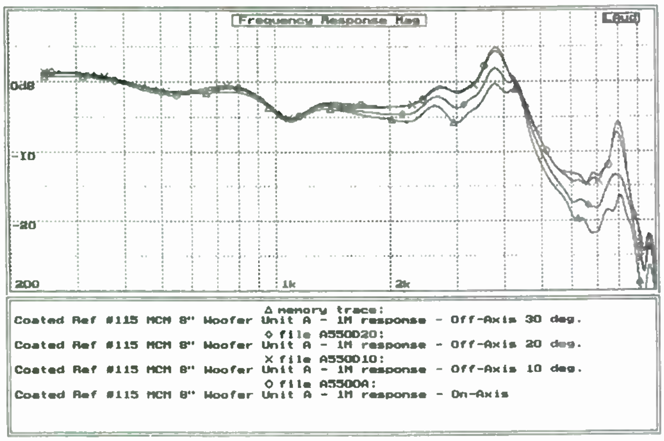

FIGURE 2: Measured response of an 8″ MCM woofer.

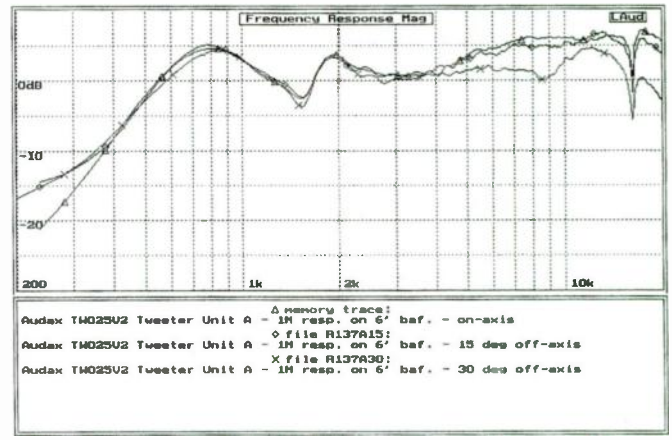

FIGURE 3: Measured response of the Audax 1″ dome tweeter.

and you don't need to purchase the components to implement each of the test COs.

A second important reason is that a good modeling program will display the acoustic response at any specified distance. You can thus model at the same distance at which you can test, verify by testing that the CO works as modeled, and then model at your listening distance to be sure the system performance is still acceptable. A modeling program can thus extend the distance over which you can predict the system's acoustic response beyond that available to you via testing.

Generally, your room also limits the range of angles at which you can test. Assume that you mount the drivers in a vertical test baffle. To test off-axis, you must swing the microphone off to one side, up or down relative to the driver axis. This moves the microphone closer to one of the room boundaries, thus reducing the echo-free time you have for quasianechoic measurement. If you wish to test to a given low-frequency limit, the angle range off-axis that you can obtain is generally limited. No such problem exists with a modeling program, which, if sufficiently accurate, can model over a considerable angle range.

MODELING PROGRAM REQUIREMENTS

There are two main requirements of a modeling program that would make it useful to me. First, it must be able to model the system in the same configurations in which I test and listen to it, and, second, it must produce results that match the test results for the same system. If the modeling can't do these two things, then it is of no more use to me than designing COs via the "classical" approach.

To meet these requirements, the modeling program must be capable of displaying the system acoustic response at the distance at which I test and over a reasonable range of off-axis angles. Programs that sum the acoustic response of the various drivers at infinity do not meet these requirements.

Figure 1 shows the difference between a program that sums the driver responses at infinity and one that sums them at a specified distance and angle referenced to a selected driver. Summing at infinity is much easier; it means all drivers show the same angle off their axis as the summation test angle.

When the program sums at a fixed distance and angle relative to a specified reference driver, the angle off-axis for each driver is different and can be fairly large. When dealing with off-axis angles of this magnitude, the assumption that each driver is a point source radiating equally in all directions is not acceptable.

Thus you must either feed the program response files for each driver at all the needed angles, which is unreasonable, or the program must model the change in each driver's response as you move off its axis. In the latter case, you need only provide the program with an on-axis response for each driver in the system—along with the physical dimensions for the driver—to enable the program to model the change in driver response with angles off its axis.

DATA FILES FOR MODELING PROGRAMS

For each driver, the modeling program requires a file representing the driver's input impedance, including both magnitude and phase angle, and a file for driver on-axis frequency response, again with magnitude and phase, covering the frequency range over which you plan to model the system.

It may be that a single file contains all the impedance and response information. Clearly, the data in these files must be in the format expected by the modeling program, so it is likely that certain programs will support only test-data files generated by certain computer test software.

The driver input impedance files are not a problem, and so need no discussion. The problem is how the frequency-response files are handled. Two important items relative to these files are how the dB response range is handled, and what phase-shift data is used.

As to the response range, the program might need the response files in true absolute dB or in relative dB. Absolute dB means the files report the actual sensitivity of the drivers, but the testing is more difficult, since you must be sure to test the drivers at the same distance and with the same voltage level. Setting this level can be very difficult if you are using an MLS or other noise-like signal, unless all the drivers are tested in the same session. To produce absolute response files you must also have a calibrated microphone for which you know the sensitivity.

For relative-response files, you simply set the basic response of each driver near 0dB. Such files are thus much easier to generate, but the program cannot set the padding the drivers require, since it does not know their relative sensitivities. The program might allow you to enter the factory-rated sensitivity, but I have found such data to be rather unreliable.

PHASE-SHIFT HANDLING

As for the file's phase-shift handling, when you measure the acoustic response of a driver far-field to ensure that you include all aspects of the driver's response, you are actually measuring two components of phase shift. The first is the shift from electrical input to driver acoustic output (which you wish); the second is the shift caused by the transit time from the driver to the microphone (which you don't wish).

The transit-time phase shift is a function of frequency and is unfortunately not easily corrected. When working with a single driver, this shift is not a problem, but when summing the response of two or more drivers, it is of major concern.

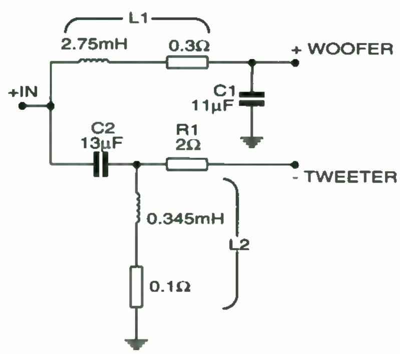

FIGURE 4: Crossover schematic.

For example, if you test at 1m, the transit-time phase shift is about 10,470° at 10kHz! In theory, if you could position the microphone with sufficient accuracy, these phase shifts would all subtract out, but this is not practical. An error of only 0.1″ represents a 27° phase error at 10kHz, and you probably do not know the driver's acoustic-center location that accurately.

To fully correct this phase shift, you would need to know very accurately both the velocity of sound and the distance from the driver's acoustic center to the microphone's acoustic center. The normal approach to get around this problem is to replace the measured phase shift with the Hilbert phase shift for the driver.

The Hilbert phase shift is that which a minimum-phase network would have if it had the same amplitude response as your driver. The minimum-phase network is a reasonable assumption for individual drivers (without a whizzer), but is clearly not valid for a full system with a CO network.

When you remove the transit-time phase shift by using the Hilbert shift, you unfortunately also lose information about the location of the driver's acoustic center. Thus, a modeling program using the Hilbert phase shift would require you to enter a horizontal offset (toward or away from the microphone) for each driver to correct for the acoustic-center (or zero-delay-plane) position relative to the driver mounting baffle. It is likely that some optimization/modeling programs are actually generating the Hilbert phase shift internally, and you must take this into account in your work.

ONE MODELING-PROGRAM APPROACH

The extreme amount of time needed to develop COs by testing convinced me a better way was needed. I had developed a program called IMPPcoEq.Exe (distributed on the Liberty Aids disk by Old Colony Sound Lab) that accepted input-impedance and frequency-response data generated by Liberty Instruments' IMP or Audiosuite tester, allowed you to enter a network ahead of the driver, and predicted the on-axis acoustic response of the driver with the network.

This program was basically developed for working with the Waveguide, but had proven to be useful in developing a CO for a single driver with a desired acoustic CO shape. Would it be possible to develop a program that took in the same information for up to three drivers, plus their physical positioning, and then plot the acoustic response of the system at a fixed distance and specified angle off-axis relative to one of the drivers? As developed above, such a program would need to model the directivity (response change with angle) for each driver.

I developed such a program, modeling the directivity as that of a round piston mounted in a baffle. Comparing the directivity modeling with the off-axis test results for cone drivers was very encouraging. All you had to do was play with the piston diameter you entered until the modeling results showed reasonable agreement with test results.

Dome drivers can show a response that changes shape, not merely rolls off, with angles off axis, so the modeling was not as effective with them. This is not considered a problem with tweeters, since you normally do not do a CO at the upper end of their response, but it might turn out to be a problem with dome midrange drivers.

VERTICAL MOUNTING

My program assumes you mount the drivers in a vertical line, entering both the center-to-center (CTC) spacing between them and the distances from the driver mounting flange to the source of acoustic output (horizontal offset). A major hurdle was identifying the proper horizontal offset for each driver, since the program used Hilbert phase data.

Again, I compared the modeling results for a variety of CO designs with test results and varied the horizontal offsets to optimize the agreement. Based on these offset values, I empirically developed equations for computing the offset based on the driver's physical measurement. The offset values that produced matching results were between where you would expect to find the driver's acoustic center and the zero delay plane.

This is clearly an area that needs further investigation, but this work showed that a modeling program can produce results that agree well with testing, which is something no previous CO design approach had accomplished for me.

My program is not an optimizer; you must enter the CO component values, since the program does not compute them. This means you see the change in system acoustic response each time you change a component value, which turns out to be very interesting

My program also uses relative-response files, so you cannot use it to set the padding between drivers. You must develop the correct padding value for the midrange and tweeter by testing or listening. While the program will work over the frequency range of 20Hz–20kHz, my room limits testing to about 300Hz at the low end, so I normally model over the 200Hz–20kHz range.

MODELING EXAMPLE

To date I have built four two-way systems

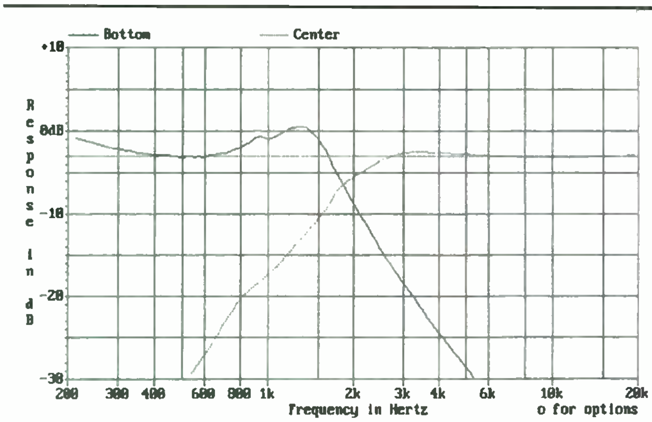

FIGURE 5: Modeled electrical outputs of crossover sections.

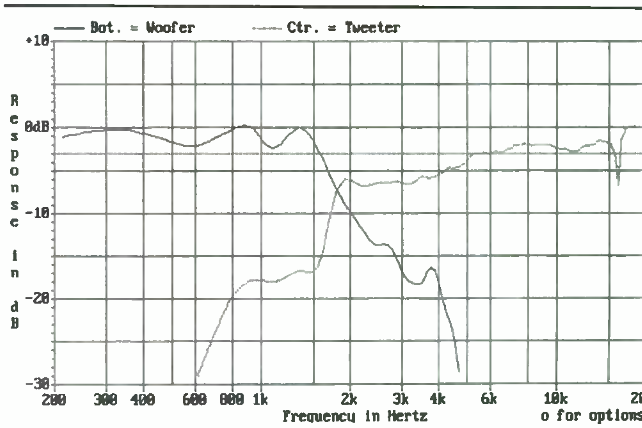

FIGURE 6: Modeled acoustic outputs of two drivers on-axis.

that had passive COs developed through modeling. All these systems worked out well when built, but I changed some CO component values for most of them after listening. The CO modeling work can get you close, but you still need to do final tuning by ear. The following is a description of one of these systems, showing how I handled the modeling procedure and how it agreed with testing.

To maintain the visibility of all the anomalies produced by the CO, I added no smoothing to any of the accompanying quasianechoic plots. Note, however, that the FFT period used in quasianechoic testing does apply some smoothing to the low end of the response curves. Since the distance used in testing multiple drivers must increase over that for an individual driver—generally reducing the FFT period you can use—I believe the modeling results are more accurate below about 1kHz than the multiple-driver testing results.

Figure 2 shows the measured response for an MCM model #55-1550 carbon-fiber-cone 8″ woofer (my driver reference #115). The woofer holds up well to about 3kHz even to 30° off-axis. Its main problem is that it has a "shelf" in its response of about 5dB in the 800Hz–1kHz range.

I thought I could correct this by building with as narrow a front panel as possible to offset the shelf with the theoretical 6dB of diffraction rise (the 2π to π transition for a box sitting on the floor). I have a program—based on the work of Ralph Gonzalez (see *SB* 3/87, p. 44)—that will process the driver frequency-response data file and modify the response amplitude for diffraction rise for a specified front-panel width.

Figure 3 shows the measured response of the Audax TW025V2 1″ dome tweeter. This is a low-resonance tweeter, but it has a rather nasty response dip in the 1.6kHz region. Using my modeling program, I developed a second-order CO *(Fig. 4)* for this driver pair by trial-and-error selection of the CO values. The 2Ω series padding resistor was a guess based on previous work with the tweeter, but it proved to be incorrect.

Figure 5 shows the modeled electrical output of each CO section driving its driver. The low-pass is clearly under-damped (the Q is higher than that for a Butterworth), while the high-pass is close to a Butterworth response. *Figure 6* shows the modeled acoustic CO on-axis response of each driver. The rather odd second-order networks combine with anomalies in the driver responses to produce a rather steep acoustic CO region.

Note that the diffraction-rise modeling for a front-panel width of 10.75″ is included in the woofer response used in this modeling work. *Figures 7* and *8* show modeling-summation plots at a 72″ distance for angles of 0° to up 40°—up meaning above the top-mounted tweeter, which is defined as the reference driver.

Notice that over this 40° vertical range, there are no major dips or peaks in the response due to the CO. Those who study published tests for production speakers will recognize that a 40° vertical range without major dip is exceptional. I obtained this result for both pairs of my drivers.

UNBELIEVABLE PERFORMANCE

To me, this performance with a simple five-component CO was unbelievable; I thought it could not be real. I found a test-baffle insert that would mount both drivers at nearly the design CTC distance, and gathered up components to implement the design CO. *Figure 9* shows modeling results at the test CTC distance, with the actual CO component values used, at the anticipated 60″ test distance, and without the diffraction-rise modeling on the woofer response.

Figure 10 shows the test results over the vertical range that my room will permit, and, to my surprise, the system behaved just as

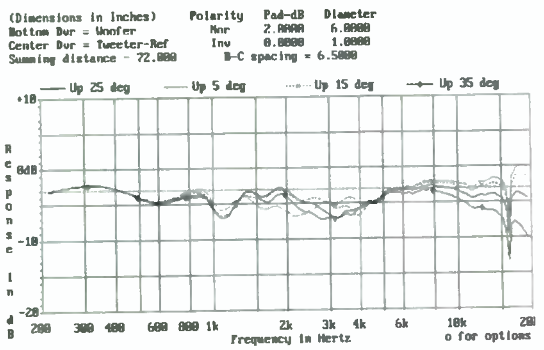

FIGURE 7: Modeled system acoustic output—up 5° to up 35°.

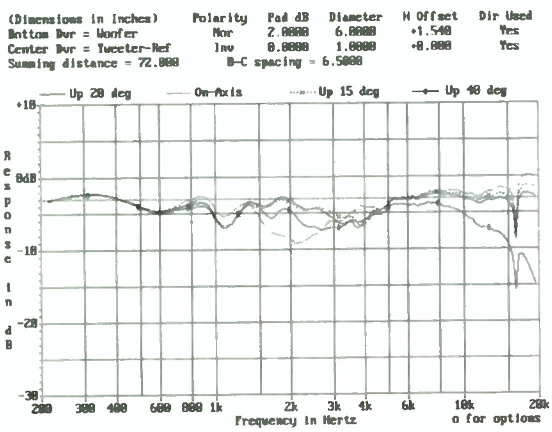

FIGURE 8: Modeled system acoustic output—on-axis to up 40°.

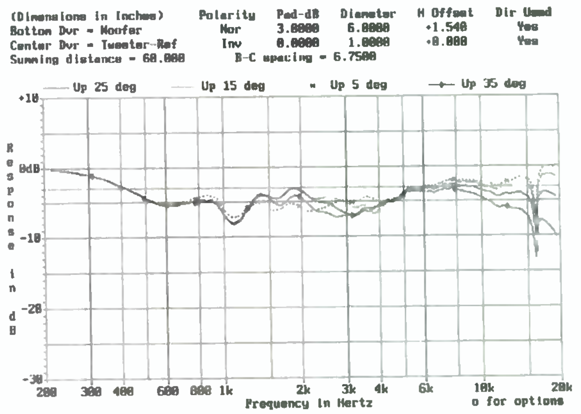

FIGURE 9: Modeled system acoustic output using test values.

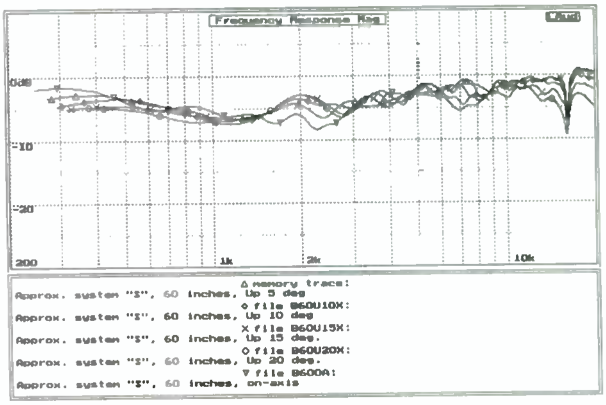

FIGURE 10: Measured system response at 60″.

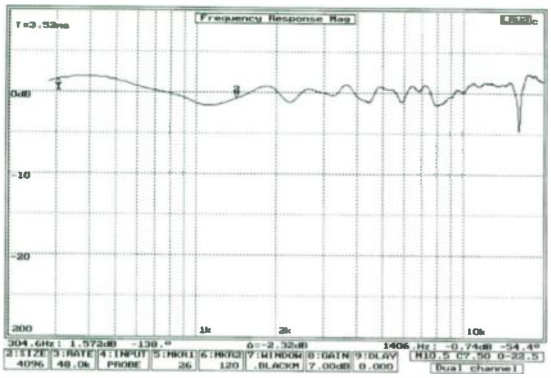

FIGURE 11: Measured response at up 15° and 60″ with 4Ω resistor.

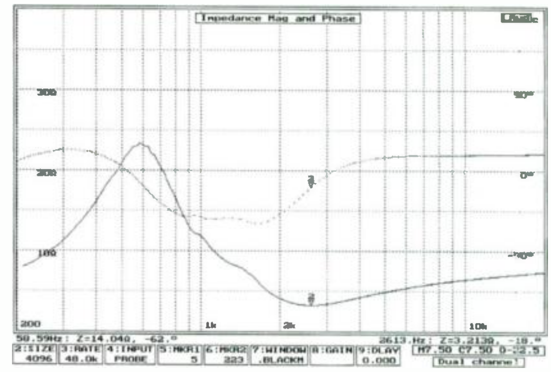

FIGURE 12: Measured input impedance to system.

modeled! The rising response at the upper end indicates my 2Ω series padding-resistor guess was wrong; testing showed the proper value was 4Ω. Again, remember that a modeling program using relative-response files is not valid for establishing the required driver-padding values.

Figure 11 shows the measured system response at the intended listening angle of up 15° with the 4Ω padding resistor, and it is on average a very flat response for a two-way system with five CO components. *Figure 12*, showing the measured system input impedance, clearly indicates that the price you pay for a simple CO network is a non-constant input impedance.

I constructed this system in a narrow box with a vertical front panel. The CO section worked just fine, but the system clearly sounded bass heavy. *Figure 13* shows the system response measured by ⅓-octave RTA and 1W into 8Ω of pink noise; the frequency values shown are the RTA band centers.

This is a good in-room response, but supports the bass-heavy nature of the sound. It would appear from *Fig. 13* that I was getting only a 3dB correction for diffraction rise, rather than the theoretical 6dB. I have not resolved this problem; theory clearly shows that the forward response should rise 6dB as you halve the radiating space. However, work by Roy Allison[1] shows plots with a 3dB rise, and the Waterhouse[2,3] equations indicate a 3dB rise for a single boundary.

Part of the confusion here may be the difference between total driver power output and output level on a specific axis, but I will leave this problem to the experts. Being a firm believer in empirical results, I have modified my diffraction-rise modeling program to permit selecting 3 or 6dB of rise. I will plan on correcting a 3dB driver shelf only in the future, thank you.

EVIDENCE OF PEAKING

Figure 13 also indicates peaking in the 100Hz region. I have noted that all systems using a passive CO will have some peaking down in this range as a result of the woofer-input impedance interacting with the low-pass inductor. *Figure 14* shows the results of near-field testing on one box without the CO, revealing a response that is flat within about 1dB to 200Hz, just as the box-design program had predicted.

Figure 15 shows the same results with the CO included, indicating added peaking and a rolloff at the upper end. Note that the relative level of the plots on *Figs. 14* and *15* is arbitrary; it is the shape you should compare. *Figure 16* shows measured gain of the low-pass network driving the woofer. It clearly shows that the network peaks (goes above 0dB), and then that it dips around 500Hz—just as the modeling had predicted *(Fig. 5).*

This low-end peaking brings up an area you should watch out for in CO development by modeling or even quasianechoic testing. Normally, your room limits the low end of your measurement capability so that you end up testing and modeling only the upper portion of the response.

The CO developed via modeling may have rather strange component values compared to those a classical design would offer. These values may interact with the woofer in the frequency range below your acoustic testing limit to produce unintended peaks or dips. Beware of this possibility.

Fortunately, I have a friend with a den having three all-glass walls that make all speakers sound too light at the bottom end. These bass-heavy boxes sound very natural in his den. I do not recommend that you try to copy this system unless you have the test capability to verify that the drivers you use have the same anomalies as those I used.

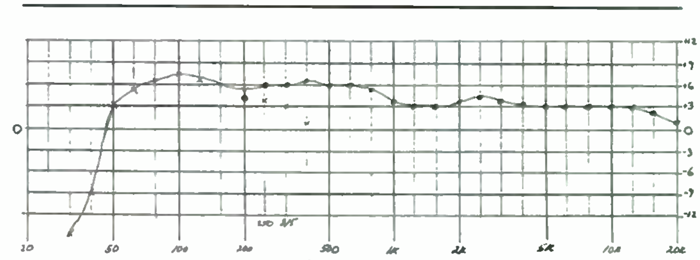

FIGURE 13: Response measured via pink noise and RTA.

WHAT CAN MODELING TELL YOU?

I have developed systems using CO modeling for one year now, and I'm convinced it is clearly the best way. It is the only approach that has produced designs that tested as predicted. The following are things I believe CO modeling can do for you.

1. It lets you see whether a group of drivers you plan to use in a system has a chance of working together as a system. Obtaining the data files before purchasing the drivers can save you some money. I don't think modeling is a replacement for breadboarding the system, which I still recommend, but if the files are available, you can do the modeling before buying the drivers.

2. It allows you to design COs with a predictable response on the listening axis and the ability to examine the response at other angles.

3. When you make changes in the CO based on listening tests, you can go back to the model to see just what areas of the anechoic system response these changes are affecting. You would think testing was a better way to do this, but the room requirements for quasianechoic testing of an enclosure greatly exceed those for driver testing. Also, the testing includes effects of the box, while the modeling does not. Comparison of the two could disclose a major problem in your

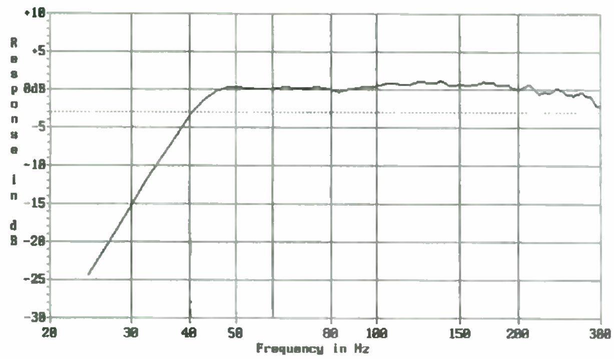

FIGURE 14: Near-field measured woofer response—no crossover.

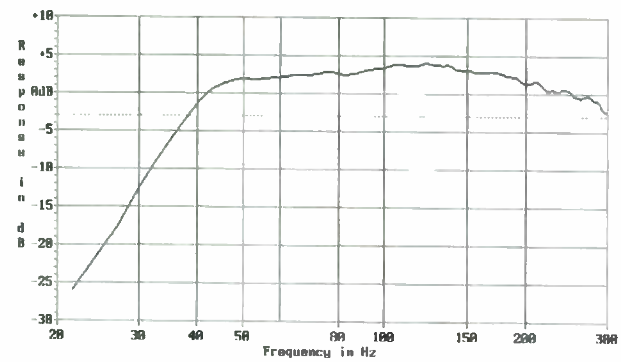

FIGURE 15: Near-field measured woofer response—with crossover.

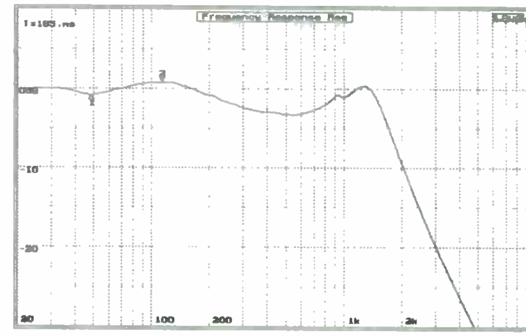

FIGURE 16: Measured gain of crossover low-pass network.

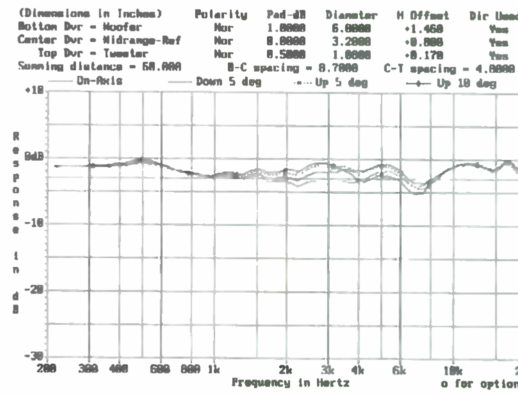

FIGURE 17: Modeled three-way system with in-phase acoustic responses.

front-panel treatment, box shaping, or grille structure.

4. It permits you to develop COs with many fewer components than the "classic" CO design approach. You can often omit the Zobels on the drivers and use a lower-order CO than you had originally intended.

5. It will bite you on occasion, as noted in the earlier example, because modeling can develop rather odd CO values that can have unanticipated effects in the frequency range below that where you can model.

6. It allows you to predict the quasianechoic response of the system at ranges that exceed those for which your room permits such testing.

7. For the system you are developing, it lets you select the optimum front-panel tip angle or stand height to place a seated listener on the proper axis. It also allows you to predict what degradation the response might have for a standing listener.

8. It gives new life to the lower-order COs. You can see just what a first-order CO will provide, and whether it will be acceptable. I find the second-order CO is now my favorite, since I can easily contour the response in the CO region by varying the Q of the networks without needing to add additional components. Note that raising the Q of a CO network causes the system input impedance to dip, so you must watch out for this.

Raising the Q of the high-pass section can also raise the power input to the tweeter at the low end of its frequency range, so beware of increased distortion or possible destruction. A first- or second-order low-pass section has a single inductor in series with the woofer, which reduces the requirement on coil resistance relative to the coils needed for a third- or fourth-order CO.

9. It allows you to establish the angle range over which your system produces a response free of serious CO-produced dips or peaks. I have found by modeling and testing that if you are using a vertical array of drivers and a tweeter with wide radiating angle, such as a dome, you should be able to get about 25° of acceptable vertical angle. This range has resulted during the development of both two- and three-way systems.

Working with a tweeter having a restricted vertical radiating angle, such as many planar tweeters, generally limits the usable vertical range to perhaps only 15° or so, making the front-panel tip angle or stand height rather critical. As seen in my example, if things work in your favor, you may be able to develop a system with a wider usable vertical-angle range.

If you wish to ensure such a wide range, I recommend that you consider the symmetrical D'Appolito W-T-W configuration. Two-way systems with small woofers have shown about 40° of usable vertical angle, but I currently have no experience with three-way or large-woofer two-way symmetrical systems.

10. Since the modeling can occur before you design the enclosure, you can establish the sizes of the major CO components and make a good estimate of the volume you should allow for an internally mounted CO.

11. If you used phony driver-impedance data files that represented fixed resistive loads, then you could use the modeling program to develop the electrical CO network shapes needed for active CO design.

12. A modeling program also allows you to impose special requirements on the acoustic CO responses if you so desire; however, this may require more time to accomplish. For example, I modeled a three-way system consisting of an 8″ woofer, a drum midrange, and a 1″ dome tweeter, with the requirement that the acoustic responses be in phase at CO.

The bandpass and high-pass are second-order, but the woofer did not go away gracefully, requiring a third-order low-pass. I used a Zobel only on the midrange, and I used series padding on both the midrange and tweeter, resulting in a total of 13 components in the CO. The modeled system response *(Fig. 17)* shows that the main response is near 0°, indicating the desired in-phase acoustic responses.

Of course, we know the real test of in-

phase acoustic response is that inverting the midrange driver should produce bad dips at the CO frequencies; *Fig. 18* clearly indicates this does happen. This example demonstrates that the "classical" rule of inverting the polarity of the midrange driver in a three-way system using second-order COs is not always valid.

ATTRIBUTES OF A GOOD MODELING PROGRAM

I consider the following capabilities to be those desirable in a modeling program. Some are necessary if you plan to be able to compare the results with testing, while others are simply desirable and useful features:

1. The program must use actual data for the drivers, not ideal response shapes. Be sure the program will work with the type of files your testing setup can export.

2. The program must allow you to position the drivers vertically (or horizontally) and insert forward/backward (f/b) offset. Perhaps someone has a way to use the actual measured phase shift, thus retaining the acoustic-center information, but if the program uses Hilbert phase shift, the results will not be accurate unless you enter the correct horizontal f/b offset values.

For predicting the final system response, these offset values must contain any other displacement effects, such as machining the drivers into the front panel, or intentionally offsetting the tweeter. If you plan to machine a tweeter flush into the front panel, then you should test it this way in the test baffle, since this may modify the tweeter's response curve.

3. The program must allow reasonable CO networks to be entered or developed by optimization. My program is limited to a six-branch ladder network for each driver, where each branch can be a series or parallel R, L, and C. It would be nice to have more branches, and if you wish to include time-delay or phase-slope correction, you may require a program that can handle lattice networks.

4. The program must allow you to specify the summation distance, so that it can produce results you can compare with test results. Summation at infinity is not sufficient.

5. The program must allow summation at a reasonable range of angles. Summation at just 0° or a small range in that area is not sufficient. You need to establish at what angles peaks and dips start to develop. My program will plot response at every 5° from up 40° to down 40°. This is marginally adequate; I would like a range of perhaps up and down 50°. As noted in my example, from 0° to up 40° was acceptable. It might actually have been acceptable at higher up angles, but my program could not show this, and my room certainly won't support testing at such high angles.

6. As explained earlier, if you sum at a fixed distance, the angle each driver makes with the summation point is different. Therefore, the program must allow you to specify which driver is considered as the reference for defining the off-axis angle.

7. The program must model the driver directivity to produce accurate results for summation at a fixed distance and to allow results for a reasonable range of angles off perpendicular to the baffle. It would be nice if

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1. Roy Allison, "The Influence of Room Boundaries on Loudspeaker Power Output," JAES, Vol. 22, No. 5, June 1974.

2. R.V. Waterhouse, "Interference Patterns in Reverberant Sound Fields," *J. Acoust. Soc. Am.*, Vol. 27, No. 2, March 1955.

3. R.V. Waterhouse, "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments," *J. Acoust. Soc. Am.*, Vol. 30, No. 1, Jan. 1958.

SOURCES

Old Colony Sound Lab
PO Box 876, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467

the program would allow you to turn this feature on or off individually for each driver.

8. The program should allow electrical inversion of any driver.

9. If the program uses relative dB files, it must provide a way to adjust the level of each driver on the summation plot. Even if absolute dB files are used, such capability is needed if you do a system with multiple woofers.

10. If the program is an optimizer, make sure it will allow you to change the CO values after the optimization. Watching how the system acoustic response changes as you change the CO component values is a great learning tool.

11. As noted earlier, COs developed by modeling can have somewhat unusual values, and thus exert a major impact on system-input impedance. It is important, therefore, that the modeling program be able to plot system Z_{in}.

12. If you use series COs, then you may prefer a program that supports that configuration. I have had no luck developing a series CO by trial and error, since changing any component value changes the response of all the drivers. I don't know whether an optimizer would perform better, or just result in loops that never converge. The point is that if you like the series CO, you may wish to have a modeling program that will support it.

13. I find many dome tweeters have a response shape that changes with angle off-axis. Many show a response with mild on-axis dips and peaks that tend to smooth out as you move off-axis. I find that sometimes modeling with the 10° or 15° off-axis response gives a better indication of the tweeter's average response. The modeling program should allow you to do this if desired.

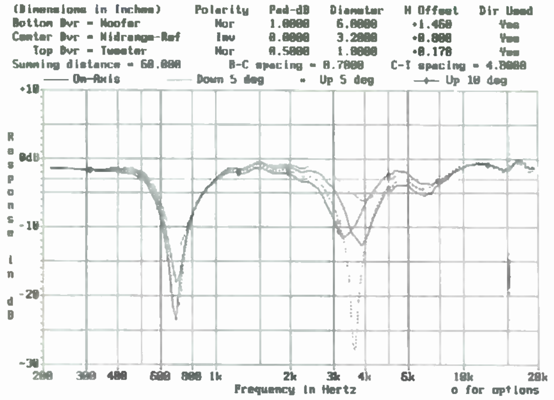

FIGURE 18: Modeled three-way system with midrange inverted.

THE FUTURE

I believe that CO modeling will eventually replace all other approaches to CO design used by the home builder, since it clearly has major advantages. Commercial modeling programs exist for those who do their own driver testing. Review the requirements given above to be sure the program will provide what you want before you purchase it.

What about those builders who do not have test capability? My hope is that they too, will be able to take advantage of CO modeling. It would be most helpful if manufacturers would supply with their drivers the files needed for modeling. It would be even better if you could obtain these files before purchasing a driver.

Perhaps someday a library of such files will be available, possibly on the WWW, so that you could use modeling as an aid in selecting drivers for your system. I believe this is something that we should all work toward.

THE FORCE, PART 2

By B. Lamy

Part 1 of this article (*SB* 7/98) dealt with the design philosophy, choice of drivers and their measurements, and filter design.

ENCLOSURE LAYOUT

Once the crossover points were known, I intended to perform some simulations to determine the overall layout of the enclosure. I had a collection of 12 drivers per side, and I didn't know exactly how to place them for best results, except that I was contemplating a line source. Primarily, I wished to put the driver enclosure on top of the sub enclosure (*Photo 1*) (better for the Allison effect, and more aesthetically pleasing, according to my spouse).

This dictated the width of the front plate. Since the midranges and tweeters don't require rear loading, and the P17REXs were to be fitted with their bowls, I sketched a very simple U-shaped box that I could attach to the top of the sub box.

It occurred to me that the absence of a back plate would allow me to put the power amplifiers inside the U, close to the drivers, partly solving a serious wiring problem. But how was I to place the 12 drivers on my front baffle? To solve this, I used CALSOD, with the following conditions:

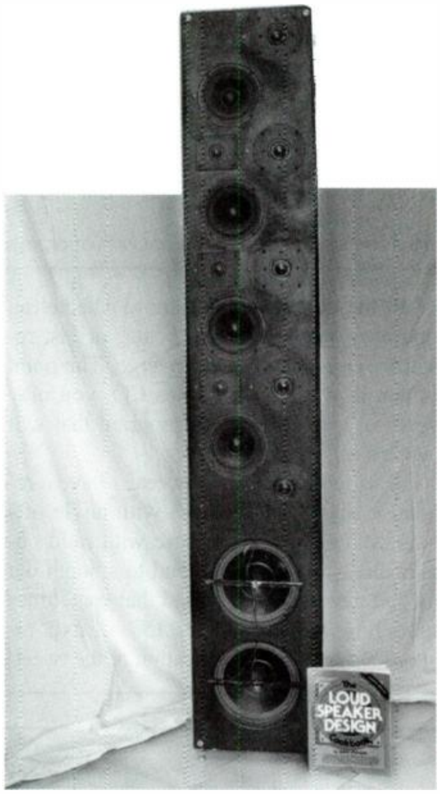

PHOTO 1: Completed prototype.

- Due to the almost perfect behavior of my corrected gyrator, I was able to abandon Liberty Audiosuite's measured frequency responses and to replace them with calculated filter responses (filter functions in the speaker-module definition, for those familiar with CALSOD).
- A more or less arbitrary constraint was to have all drivers of the same type on a vertical line (line-source principle).
- As with any multidriver concept, you can observe only a flat response on a certain axis, so my positioning criterion was that the responses at vertical angles of −10°, −5°, +5°, and +10° should sum as flat as possible compared to the power response (i.e., without phase consideration) of the system.
- The process of simulation was rather tedious, because CALSOD does not support more than seven drivers in a system. To overcome this limitation, I performed three steps. First, I chose a group composed of a P17REX, an LPG50, and a D260 and simulated them separately in various geometrical configurations (lateral and vertical spacing). Second, I exported each of these global-response files to create new virtual drivers D1, ..., Dn. Finally, I made simulations using three D260 drivers, the remaining two LPG50s, and one P17REX.

DELAYED RESULTS

Hours and hours later, I came to the results presented in *Tables 3* and *4* and *Figs. 16* and *17*, which correspond to the geometrical arrangement of *Fig. 18* and driver polarity. In *Figs. 16* and *17*, you can see that away from the optimal axis, there are very wide aberrations, at least to the extent of 10dB. This is unpleasant to see, but remember the following:

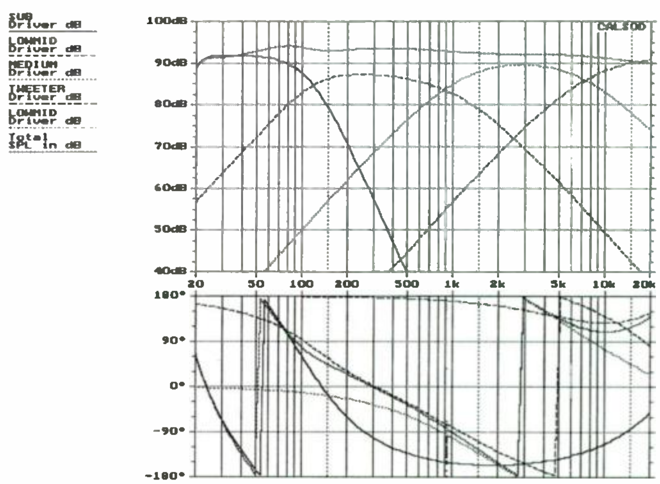

FIGURE 16: CALSOD responses for the single P17 + LPG50 + D260 group.

- In a real room, these responses tend to sum up, and I can tell from experiments that what you hear at a given location is somewhere in between the sum of these curves and the perfectly flat power response of the system. This is the price you pay for vertical beaming and extended dynamics.
- A 20dB-deep suck-out 50Hz wide causes less degradation than a 2dB one spread over three octaves (unless you listen only to sweep-tone test records).
- Don't trust too much a perfectly flat frequency response in commercial advertising before you have seen the phase response of the speaker.
- Line-source speakers demand that you sit in the sweet spot for optimal listening, and at a distance at least twice the height of the system (if you do not, the summing effect will be excessively diminished by the distance discrepancies between the drivers and your ears). My simulations were made at a 4m distance.

As you will see, the Force's electronics are quite important and rather complex. At the time I write this, the system is still in the prototype stage, and therefore I will not provide a detailed parts list or PCB layout.

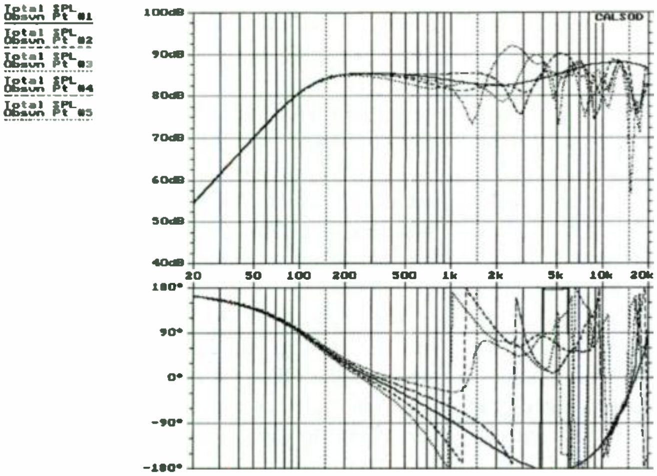

FIGURE 17: CALSOD responses for the whole array at various angles.

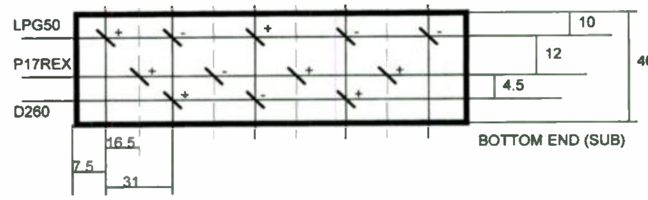

FIGURE 18: Front-baffle driver-hole locations; not to scale; dimensions in centimeters.

Nevertheless, the following holds true:

- I really designed, constructed, and debugged all the circuits I present here.
- I will describe any critical points (component selection, layout, and so on) so that you can make your own without nasty bugs.
- If you decide to realize the system and have problems, I will be glad to answer written questions.

FILTER

Figure 19 is the filter

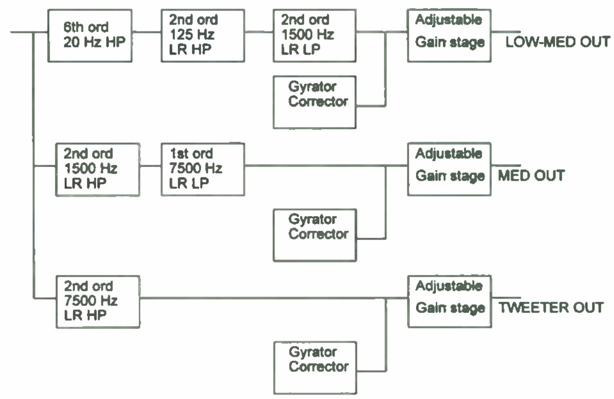

FIGURE 19: Filter block diagram.

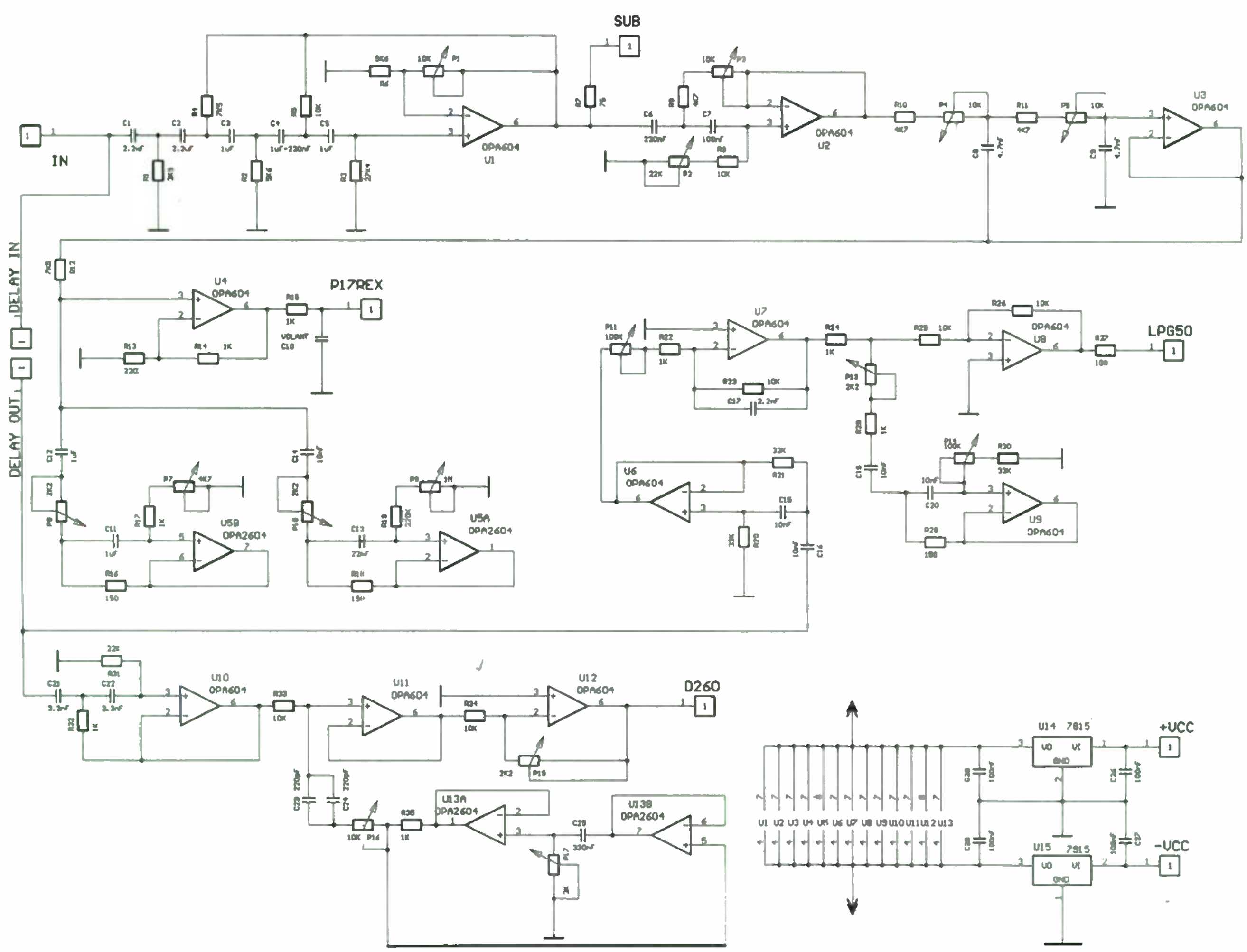

FIGURE 20: Four-way filter schematic.

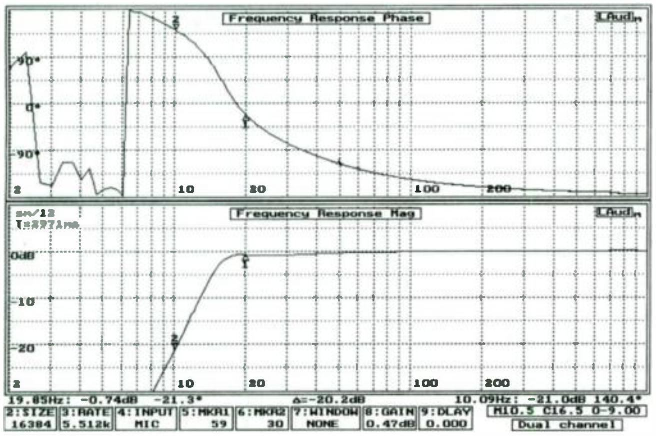

FIGURE 21: Response of the 20Hz high-pass filter for the sub.

block diagram. As you can see, it is quite in accord with what I described above, except that the 7.5kHz low-pass section of the medium filter is first order instead of second order. This is due to the natural rolloff of the LPG50FA, which required a first-order section to combine into the desired acoustical Linkwitz-Riley (L-R) shape. Each output is fitted with an adjustable gain stage that enables you to compensate for different drivers' sensitivities.

The schematic of the filter is shown in *Fig. 20.* U1 and the associated components form a fifth-order high-pass 20Hz filter, the purpose of which is to avoid excessive cone excursion of the subwoofers, especially with warped vinyl records. The response of this filter is shown in *Fig. 21.*

U2 and U3 are the 125Hz and 1.5kHz L-R filters for the P17REX, U4 is the adjustable gain stage, and R15 and C10 are dedicated to removing op-amp-generated out-of-band noise. U5A and U5B are the gyrators that compensate for the anomalies in the P17REX response.

The input of the filter is also connected to a delay cell (*Fig. 22*) that compensates for the acoustical-center misalignment between the P17 and the LPG50, which appeared to be 240µs. I copied it from a *Speaker Builder* article.[6] The phase response almost perfectly approximates a pure delay in the region of interest, as shown in *Fig. 23.* The output of this delay cell feeds the medium and tweeter sections.

U6 and U7 are used for the second-order high-pass and first-order low-pass at 1.5kHz and 7.5kHz. U7 is also used for variable-gain cell implementation. U8 inverts the phase for an overall in-phase response, and U9 is the gyrator-correction section.

U10 is the 7.5kHz high-pass for the tweeter. As you may see from the R32/R31 ratio, this is not at all an L-R shape, but is due to strong interaction between the gyrator correction centered on U13 and the U10 high-pass. This is caused by a particular response anomaly of my D260, and you will very likely need to change R32 if you use other tweeters. U12 is the variable gain cell, and U14 and U15 provide regulated ±15V supplies.

The filter sections are each followed by a buffer cell, which can, with very low distortion, feed long cable runs and the appropriate number of power-op-amp inputs (five for the midranges).

The schematic of the buffer cell is shown in *Fig. 24.* The value of ±VCC is ±20V, whereas the OPA604 is connected to the ±15V of the filter board. P1 is intended to accommodate various power-amplifier input sensitivities in case you don't use those described below.

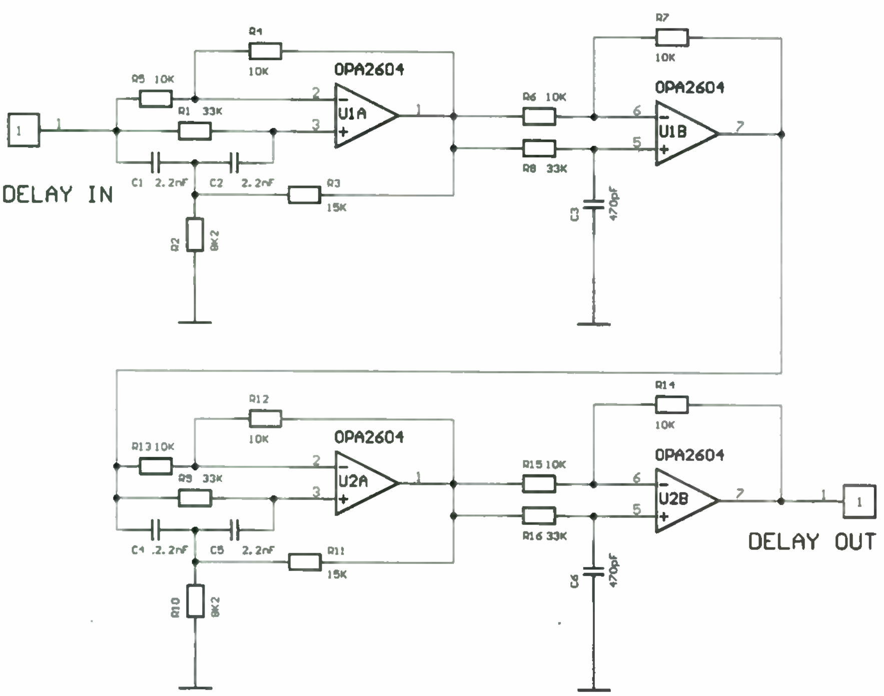

FIGURE 22: 240µs delay-cell schematic.

FILTER CONSTRUCTION CONSIDERATIONS

• Op-amp selection: Looking at the *Fig. 20* schematic would drive addicts of single-ended triode amplifiers crazy. A handful of op amps is a crime against music. Although I may point out that before the recorded signal

TABLE 3

CALSOD SIMULATION FOR P17, LPG50, AND D260

```
CIRCUIT
!
SPK 1 0 1 0 0.15 0 positive
SPK 2 0 1 0.12 0 0 negative
SPK 3 0 1 -0.045 0 0 positive
!
DRIVER 1
LOWMID
!
SOUND PRESSURE
!
SEN    92    DB
!
BUT 120 2 HIGHPASS
LKR 1100 2 LOWPASS
!
DRIVER 2
MEDIUM
!
SOUND PRESSURE
!
SEN    92    DB
!
LKR 1100 2 HIGHPASS
LKR 7500 2 LOWPASS
!
DRIVER 3
TWEETER
!
SOUND PRESSURE
!
SEN    92    DB
!
LKR 7500 2 HIGHPASS
!
TARGET SPL
SEN   92  DB
RAB 1 0 0 1 active
```

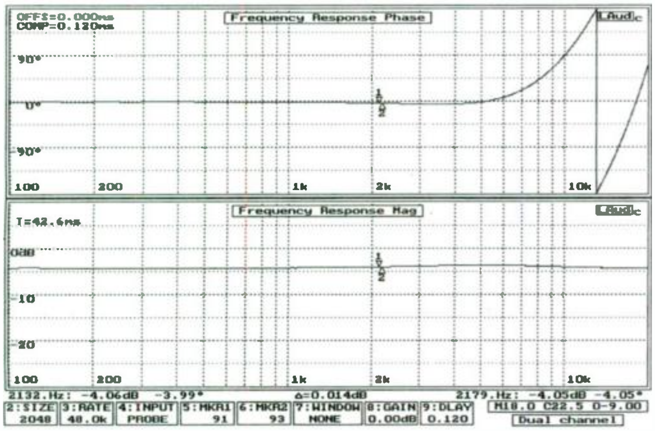

FIGURE 23: Phase response of a half-delay cell (120μs).

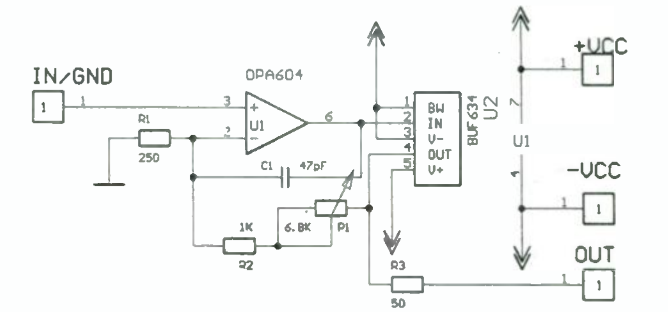

FIGURE 24: Filter output buffer schematic.

is fed to a precious triode, it has travelled through dozens of op amps, I admit that this type of all electronic filtering requires some precautions. The gyrators are especially sensitive to amp voltage and current noise (they feature a noise gain inversely proportional to the series resistor of the simulated inductor), and they also require wide open-loop bandwidth. Therefore, I recommend the excellent OPA604/OPA2604 from Burr-Brown. Analog Devices' AD829 and AD797 are also excellent choices, but they are more expensive. Don't use low-cost TL074s or the like, or you will have excessive noise and poor sound performance.

- Decoupling is also most important. On my prototype, I bypassed each amp with two low-ESR 10μF capacitors. I also recommend a two-layer PCB, with the upper face used as a ground plane.
- All resistors are 1% ¼W metal film.
- Capacitors (except decoupling) are 5% polystyrene for values below 22nF; 1% polystyrene for the delay cell; and Siemens MKT above 22nF.
- All potentiometers are ten-turn Cermets.
- The BUF634 buffers must be cooled with appropriate (7°Kelvin/W) power heatsinks. Take care that the –VCC supply is connected internally to the case. Decouple each BUF634 with two low-ESR 220μF capacitors.

SERVO SUB

In the schematic of the servo loop (*Fig. 25*) U1A is the acceleration-speed converter of the input signal and U1B is that for the mike signal path. U1C is the differential amplifier that feeds the power amplifier centered on U9, a rugged NS LM12.

Box-response compensation is provided by C45, R42, and R43 (for frequencies

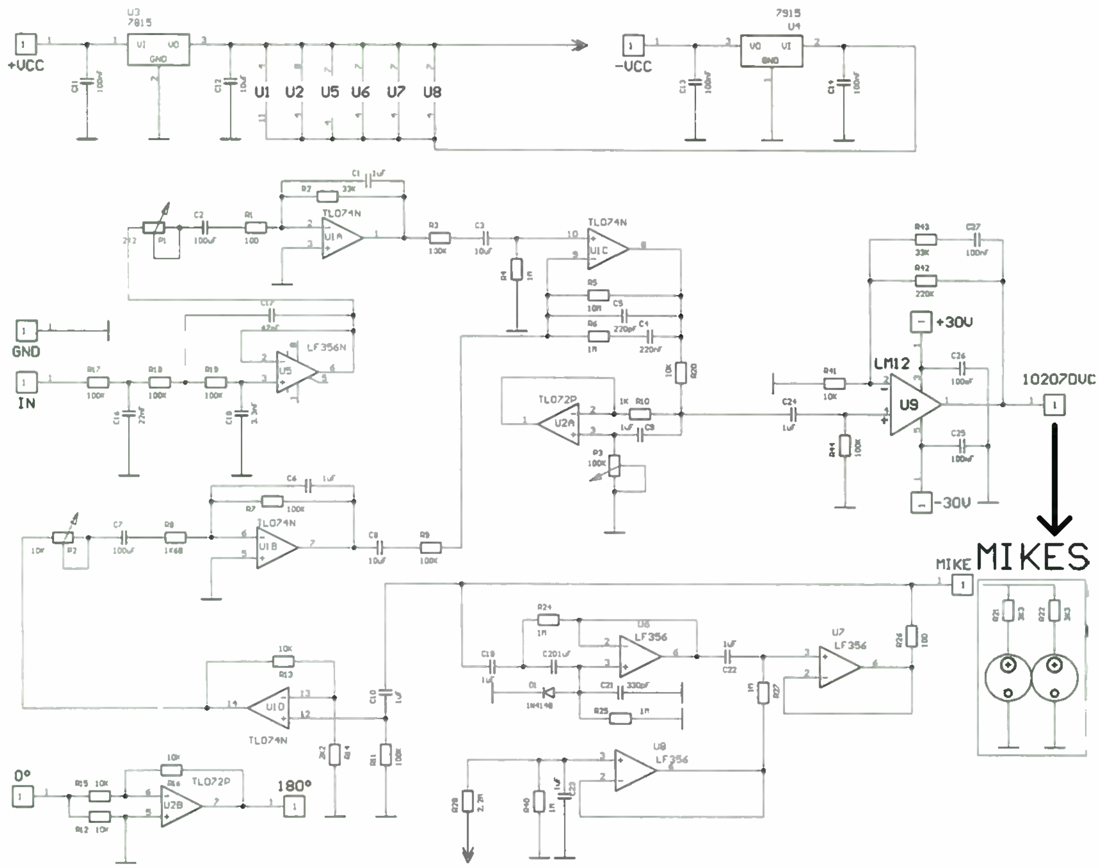

FIGURE 25: Servo control for the subwoofer schematic.

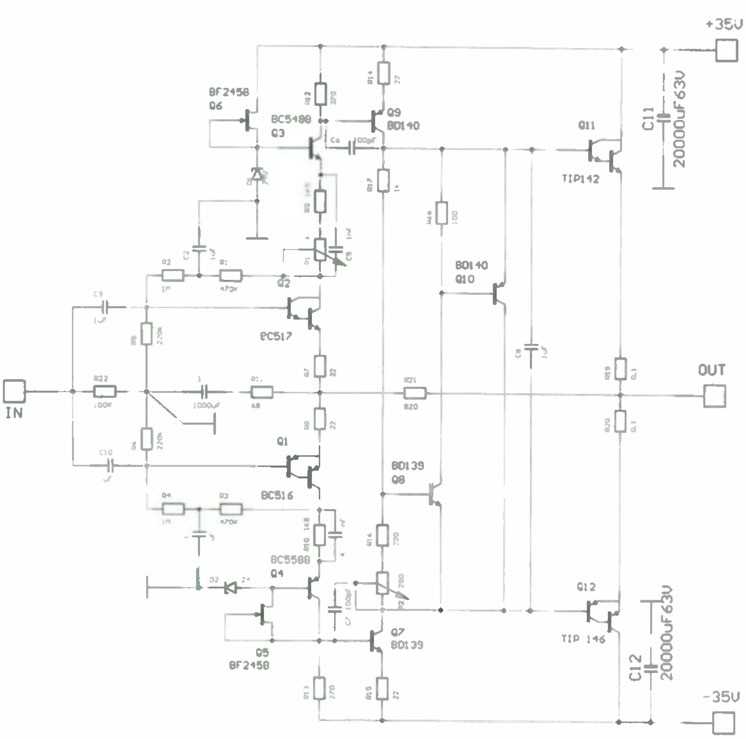

FIGURE 26: Power-amplifier schematic.

PHOTO 2: Closeup showing mike-mounting detail.

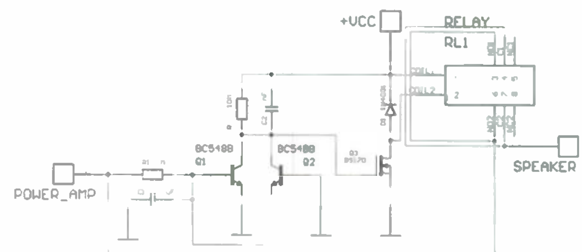

FIGURE 27: Speaker-protection-circuit schematic.

above 100Hz) and phase stability at the low-frequency cutoff of the loop is ensured by the simulated inductor created by U2A (for frequencies below 5Hz).

U6 and U7 form a super-gyrator (simulating a $1/f^3$ impedance) to avoid very low-frequency external signals, such as door and window openings, affecting the loop and making drivers bottom severely. Just

TABLE 4

SIMULATION FOR COMPLETE ARRAY

```
CIRCUIT
!GMA n½1
SPK 1 0 1 0 0 0 negative
!
!GMA n½2
SPK 2 0 1 0 0.31 0 positive
!
!GMA n½3
SPK 3 0 1 0 -0.31 0 positive
!
!Medium du haut
SPK 4 0 1 0.12 0.62 0 positive
!
!Medium du bas
SPK 5 0 1 0.12 -0.62 0 negative
!
!grave du bas
SPK 6 0 1 0.12 -0.46 0 positive
!
!
DRIVER 1
!
SOUND PRESSURE
!
SEN    0    DB
!
FILE impfrd c:\calsod\ber\enceinte\GMA_NM.FRD 0.0 0.0 SECONDS 0.0 DB
!
DRIVER 2
!
SOUND PRESSURE
!
SEN    0    DB
!
FILE impfrd c: \calsod\ber\enceinte\GMA_NM.FRD 0.0 0.0 SECONDS 0.0 DB
!
DRIVER 3
!
SOUND PRESSURE
!
SEN    0    DB
!
FILE impfrd c: \calsod\ber\enceinte\GMA_NM.FRD 0.0 0.0 SECONDS 0.0 DB
!
DRIVER 4
!
SOUND PRESSURE
!
SEN    92    DB
LKR 1100 2 HIGHPASS
LKR 7500 2 LOWPASS
!
!
DRIVER 5
!
SEN    92    DB
LKR 1100 2 HIGHPASS
LKR 7500 2 LOWPASS
!
DRIVER 6
!
SOUND PRESSURE
!
SEN    92    DB
!
BUT 120 2 HIGHPASS
LKR 1100 2 LOWPASS
!
TARGET SPL
!
SEN   92   DB
!
RAB 4 0 0 1 Active
```

(continued at top of column)

```
RAB 4 5 0 1 Active
RAB 4 10 0 1 Active
RAB 4 -5 0 1 Active
RAB 4 -10 0 1 Active
```

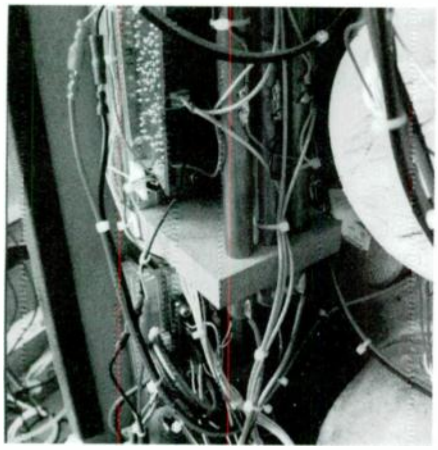

PHOTO 3: Inside The Force.

a fast opening of a door can easily reach 150dB SPL, but you can't hear it at very low frequencies.

U8 provides biasing in case you prefer to use electrets, but can be omitted if you use Shures. You can obtain optimal noise performance by replacing U6 and U7 with an OPA604 (the reason for not having integrated U6, U7, and U8 in a TL074 package).

U5 is a third-order filter that is really a part of the complete filter described above, but is physically installed on the servo board. P2 is dedicated to gain adjustment of the loop to the mikes; P1 controls the input gain for level adjustment with the other drivers; and you should adjust P3 for optimal transient response.

The sub's power supply, not shown on the schematic, consists of a traditional 50Hz [European] transformer, plus a diode bridge and two 20,000μF 50V capacitors with 20A slow-blow fuses.

A relay (not shown) activated by the mains switch connects the LM12 and the speakers to avoid power-on "chumps." Mike connection is through low-noise shielded wire. All four voice coils are wired in parallel (which results in 2Ω impedance at the LM12 terminals).

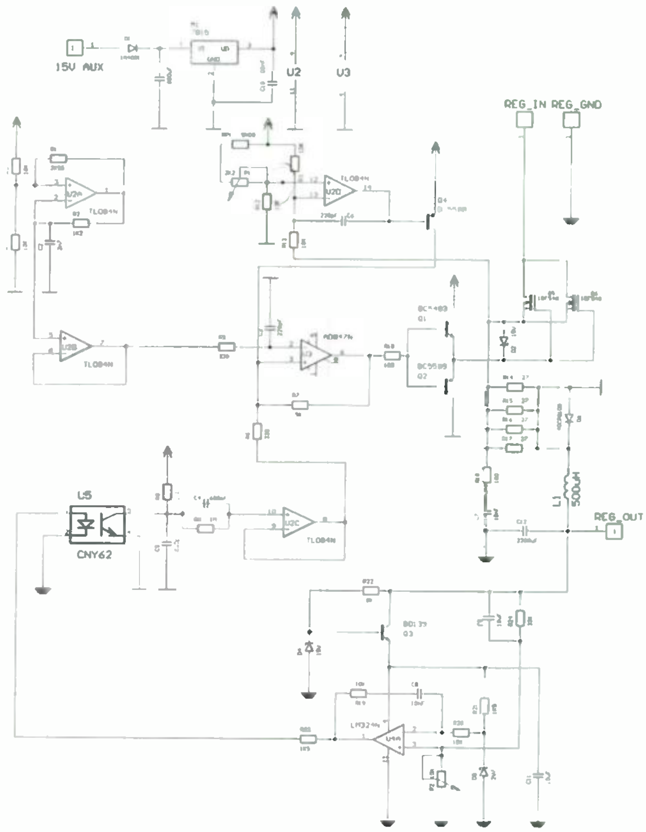

FIGURE 28: 30V switchmode power-supply schematic.

SERVO CONSTRUCTION CONSIDERATIONS

• The components are the same as for the filter, except that a single-sided PCB without a ground plane is sufficient. The LM12 drives a low impedance (2Ω) and requires a large heatsink (2°K/W maximum), as well as a big power-supply transformer and rectifier capable of delivering 15A (see Supply Connections, following).

• The mikes are on an aluminum bar screwed in front of the 10207DVC *(Photo 2)*. If you use electret mikes, to avoid undesirable distortion caused by connection wire movements, you must use very thin wire and make two or three mechanical release turns (see Mitey Mike article in *SB* 4/97, p. 10).

AMPLIFIERS

Figure 26 shows the power amplifiers, with the schematic derived from an *Elektor* article, "Étage de Sortie Audio Robuste," July, August '94, pp. 104/105. These amps have a somewhat unusual structure, but feature low price and very pleasant performance (especially the constant distortion from 10Hz–20kHz rising from 0.006% at 1W to 0.1% at full power). I had built two of them for a previous project and found their sound so good that I build 22 more for The Force.

Q1 and Q2 form the voltage-gain stage. They are loaded by a common base stage made of Q3 and Q4, ensuring a wide open-loop bandwidth (which in turn leads to the frequency-flat distortion curves). Q7 and Q9 are common emitter predriver stages that drive the output transistors Q11 and Q12. Q8 and Q10 act with R19 and R20 to set the idle current. P1 allows fine-tuning of the offset voltage, and P2 is used for idle current adjustment.

POWER-AMPLIFIER CONSTRUCTION CONSIDERATIONS

• All resistors are 1% ¼ W metal film, except R11, which is rated 3W to avoid thermal distortion at low frequencies.

• All capacitors are MKT, except the 100pFs, which are ceramic; C1, which is a high-quality nonpolar (Nichicon); and C11 and C12, which are low-ESR electrolytics.

• Q7 and Q9 are cooled by a single 5°K/W heatsink.

• The 12 amplifiers required for an enclosure are mounted on a single 1°KW, 40″ long heatsink *(Photo 3)*. This means very high thermal time constant, and initially it caused a thermal runout problem. To avoid this, you must mount Q8 and Q10 not on the heatsink, but on top of Q11 and Q12. The mounting screw goes through the holes of both transistors, and the plastic face of Q8/Q10 is in contact with the upper side of the metal plate of Q11/Q12. Naturally, Q11 and Q12 should be insulated from the heatsink with mica insulators coated with thermal grease.

• Ground and ±35V power distribution is ensured by ½″ copper tubes, inserted in three holes through small wood blocks screwed to the heatsink *(Photo 3)*. This provides low-loss power distribution and also helps to fix in place the various cables running to and from the amplifiers. Nevertheless, beware of short circuits! Trying to discharge 12 20,000μF capacitors charged to 30V in 1μs with a pair of pliers can be very dangerous.

SPEAKER PROTECTION

Considering the amount of money represented by the various drivers, I thought it wise to provide some sort of protection in case the power supplies or a power amplifier should fail. It never proved necessary up to now, but you never know.

The speaker-protection cell presented in *Fig. 27* is very simple. R1 and C1 provide an input filter that eliminates audio frequencies, so Q1 and Q2 can detect only a DC component in the amplifier output. This covers 99% of the possible supply and amp failures; it does not, however, cover high-frequency parasitic oscillation within the amplifier.

If the DC component is positive, Q1 conducts; if it's negative, Q2 does so. In either case, Q3 switches off, and the relay opens, protecting the speaker. It is obvious that this can be efficient only if the VCC supply is independent of the main power supply.

The only critical point here is the choice of relay, which must withstand at least 20A, and also provide good low-level contact quality. Siemens and Omron provide excellent models for this application.

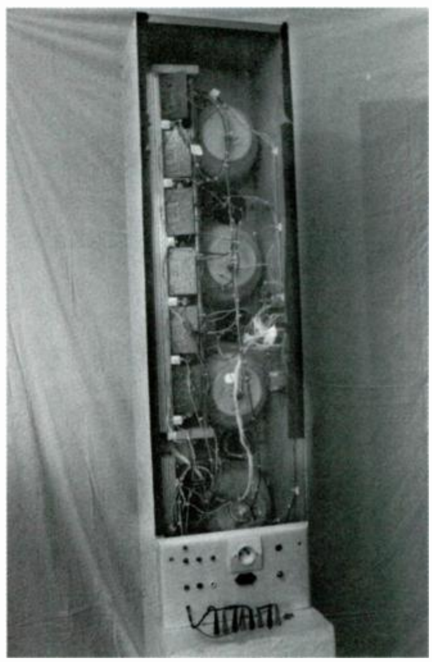

PHOTO 4: Back side of the enclosure.

POWER SUPPLIES

Most of the time, except for sound-reinforcement applications, audio power amplifiers make use of linear—often unregulated—power supplies. For this project, I preferred regulated power supplies, for the following reasons:

- The multiamplification benefits of diminishing amplifier constraints by splitting the reproduced spectrum may be spoiled if the power supplies of one section (low) influence those of other sections (mid and high). You can avoid this either by having separate unregulated supplies or one common well-regulated supply.
- Power amplifiers always sound better when fed by a regulated supply.
- You cannot match unregulated supplies better than roughly 1V between positive and negative sections, and the amplifiers described previously show an offset drift if supplies are not closely matched.

Then comes the following problem: if I had used a linear supply, power dissipation would have been terrific. Let's imagine the average level of each amplifier is 10W for realistic level listening; this means each amplifier draws 1.1A RMS, and that is a total of 13A per enclosure, which means 13 × 30, or about 400W delivered by the supplies. As a rule of thumb, a linear supply would dissipate the same amount of energy that is delivered, roughly 500W. This would lead to unacceptable heatsink size, requiring fans and causing noise.

SWITCHABLE SUPPLY

I decided to go for a switchmode power supply, which has the benefits of low loss (I achieved an efficiency of 97% for 500W output); very low ripple, due to high switching frequency; and very good transient regulation.

I rejected supplies based on switching directly to the mains line, because of breadboarding security concerns (all who have tried know what I mean) and also because of the EC standards on mains spectral pollution caused by switching transients. I chose a traditional 50Hz power transformer with rectifier and capacitors, followed by a step-down converter. The schematic appears in *Fig. 28*.

U2A/U2B is the oscillator, which delivers a triangular waveform at about 50kHz. U3 is a high-speed comparator, which compares the triangular wave with the error signal coming from U2C. The resulting fixed-frequency variable duty-cycle square wave is forwarded to the Q1 and Q2 transistors that drive the gates of the MOSFET switching transistors (IRF540). D2 limits the voltage excursion on the gates.

The sources of the IRF540 are connected through current-sensing resistors (R14–R17) to the inductor and recovery diode (40CPQ100). C12 is a low-ESR output capacitor. C7 and R18 form a snubber network that limits high-frequency spurious emissions.

U4A compares the output voltage to a reference voltage (D5 zener diode), and drives the U5 optocoupler to produce the error signal. You can adjust output voltage with P2. Q3 and D4 extract the U4 supply from the output voltage.

U2D compares the voltage across the current-sensing resistor to a reference value derived from the 15V auxiliary supply, and, in case of current excess, reduces the output voltage by lowering the duty cycle of U3. P1 adjusts the current limit.

POWER-SUPPLY CONSTRUCTION CONSIDERATIONS

All this is quite simple, but requires very careful construction. The main points are the following:

- The ground of the auxiliary 15V power supply is connected to the sources of Q5 and

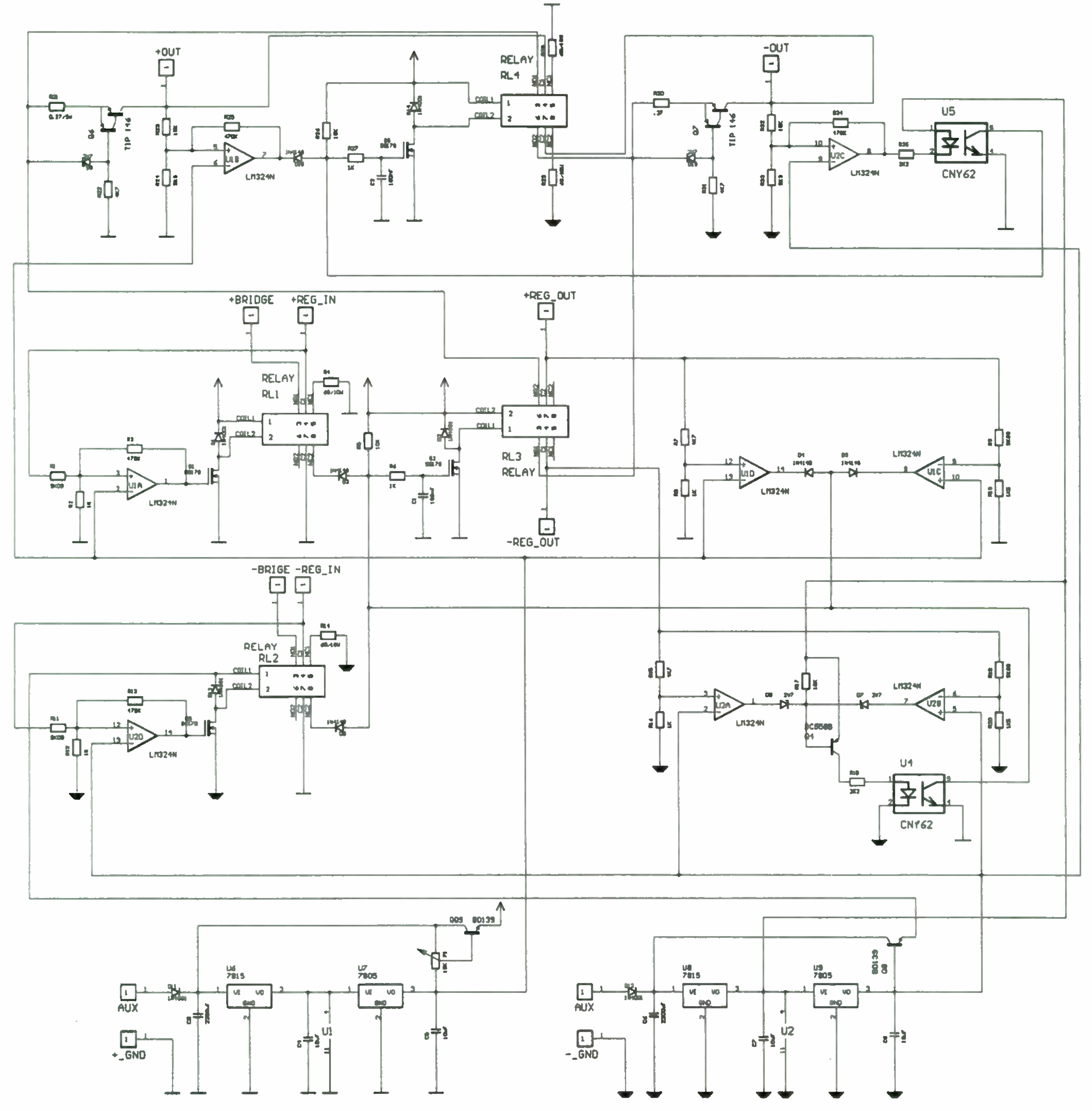

FIGURE 29: Switchmode-supply controller/sequencer.

Q6, and not to the output ground.

• All high-current paths (DC input, Q5 and Q6, R14–R17, L1 and C12) must be very large traces on the PCB (a minimum of ¼″).

• Do not replace U4 with another amp in which the common-mode voltage does not include ground, or the supply will not start.

• C8, R19, R23, U5, R8, C5, C4, and R9 are carefully chosen for optimal transient response. Do not replace them with other types or values.

• You should mount Q5, Q6, and D6 on proper heatsinks (3°K/W is enough), although they remain cool even at very high output (this is the reason for choosing a low 50kHz switching frequency).

• L1 is an air-core inductor in my prototype (two layers, 80 turns on a 1″ inner diameter). If you use a core, be sure it's one with a very low loss; otherwise the inductor will saturate, and Q5 and Q6 will overheat.

• To avoid RF interference problems, you should follow these high-current, high-frequency rules: ground planes on PCBs; short leads on switching parts; proper grounding (don't forget the heatsinks that easily become efficient antennas); short and large PCB traces for high-current paths; shielding of the whole. To accommodate EC rules for RFI/EMI, my prototype includes a 0.05″-thick steel case, with an additional 0.1″-thick aluminum case for the coreless inductor.

SWITCHMODE CONTROLLER/ SEQUENCER

The high currents of the SMPS required some additional electronics for failure protection and for capacitor-life preservation.

With 12 amplifiers connected in parallel, each supply sees 12 × 20,000 = 240,000µF, and the sudden application of mains voltage would cause damaging current surge and possibly blow the fuse.

Referring to the schematic of *Fig. 29*, the controller contains two more or less identical sections (one for positive supply, and one for negative). At start-up, relay RL1 is open, and the primary DC supply voltage (+bridge input) rises. This +bridge voltage is connected through a 2Ω resistor to the +Regin switchmode regulator input (*Fig. 30*). U1A controls the value of this voltage, with the 5V reference provided by U7. Once it has risen above 52V, RL1 closes. The same mechanism is true for the negative part, with U2D and RL2.

The output of the regulators (+Regout and −Regout) are checked by U1C, U1D, U2A, and U2B. If the output is not between 28V and 35V (short-circuit condition or switching-transistor failure), Q2 remains off, and RL3 prevents any connection to the output. Q4 and U4 provide isolation between the positive and negative sections. When RL1 is closed and the regulator output voltages are correct, Q2 is on and RL3 is closed.

The outputs of the regulators are then connected to Q6 and Q7, current sources that gently charge the power amplifiers' filter capacitors. U1B and U2C control this charging process. When the final voltage is reached, Q5 comes on and RL4 closes, connecting the regulators' outputs to the output terminals (+Out and −Out).

Auxiliary power supplies are used (one for positive, and one for negative). Their grounds are *not* connected, since that of the positive one is connected to the output ground and the ground of the positive regulator, whereas that of the negative one is connected to the output of the negative regulator. This may seem complex, but it is required for maximum failure protection.

There are no special construction considerations, except that some PCB traces should be able to sustain 40A currents (½″ width), and that the relays must also sustain high current values. R4, R14, R21, and R30 are high-power wirewound resistors. You can use almost any optocoupler for U4 and U5, provided the current-transfer ratio is at least 100%.

SUPPLY CONNECTIONS

Figure 30 gives the general connection plan for the various (and numerous) supplies. The important points to note are:

- This schematic is given for one enclosure (left or right).

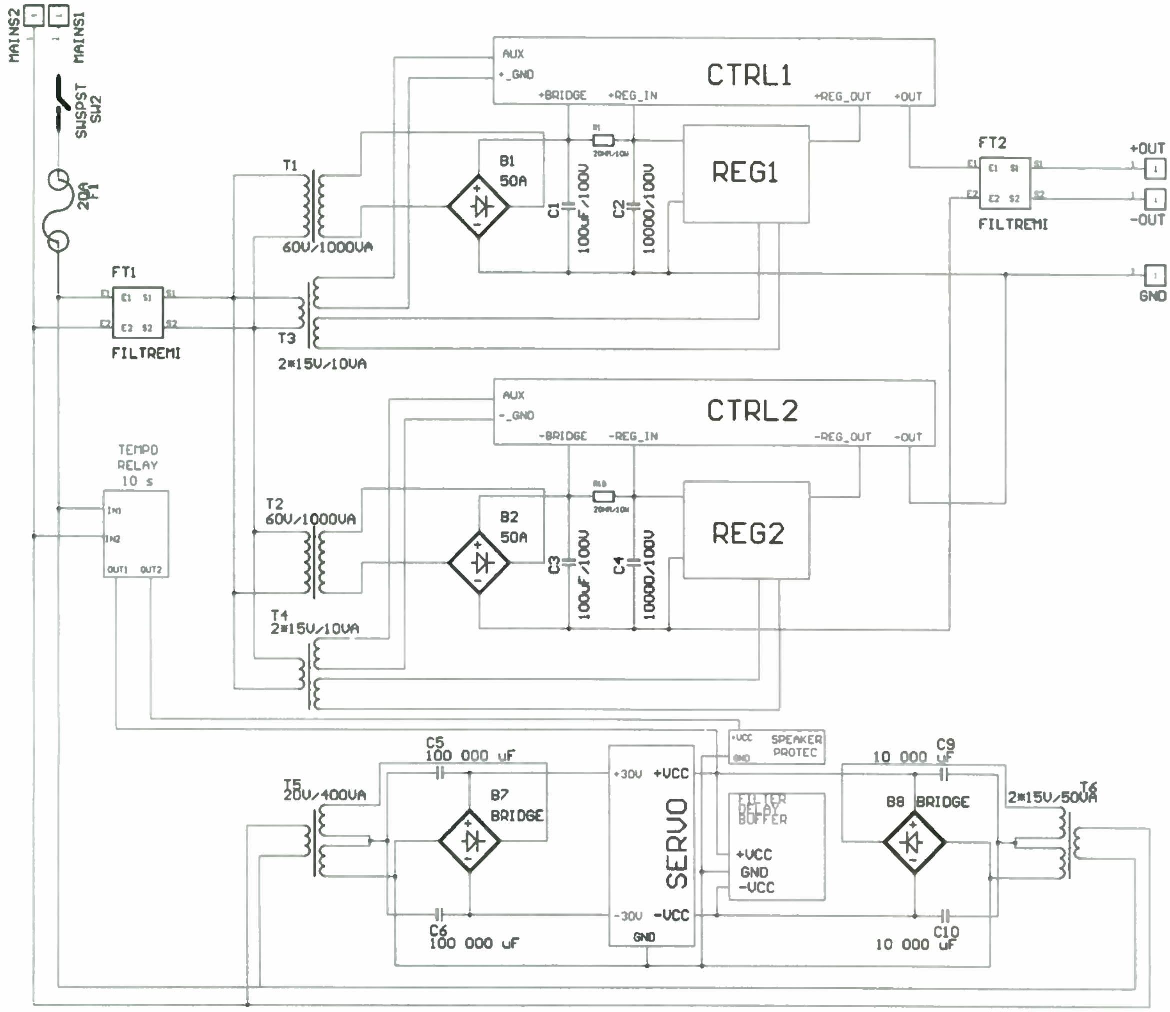

FIGURE 30: Connecting global supplies.

• EMI filters are used both for mains input and output terminals for the SMPS, to provide adequate attenuation of spurious emissions. These filters should sustain 20A for the mains and 50A for the output, with at least 40dB attenuation in the 10MHz–300MHz range. They are not cheap.

• T3 and T4 must feature a secondary-to-secondary voltage isolation of at least 150V; if you can't find such transformers, you can split each of them into two ordinary single secondary 15V transformers.

• C2, C4, C5, and C6 may be made of several 20,000µF items connected in parallel.

• Note the use of a delayed relay for activation of the speaker-protection cells; this avoids any turn-on chumps and is an additional protection against supply failure.

• Make sure to use appropriate heavy-gauge wire where specified, and be careful to ground the metallic cases.

THE SUB ENCLOSURE

Building the enclosure of the sub is quite straightforward. It is a rectangular box of about 2.5ft^3, with the following characteristics:

• made of 2″ MDF;

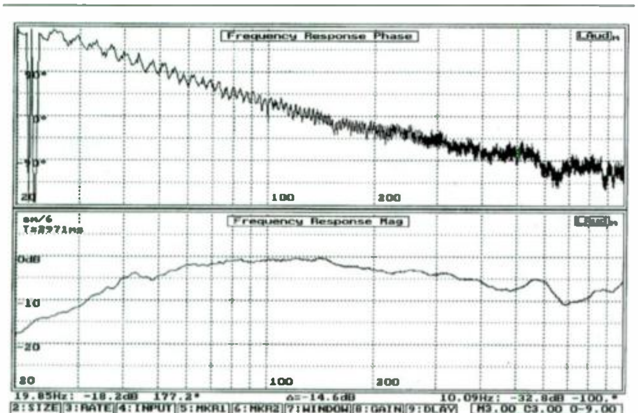

FIGURE 31: 10207 response in 2.5ft^3 closed box, without servo.

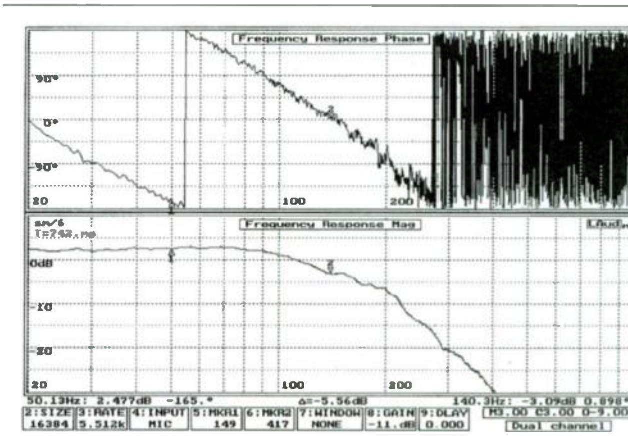

FIGURE 32: 10207 response in 2.5ft^3 closed box, with servo.

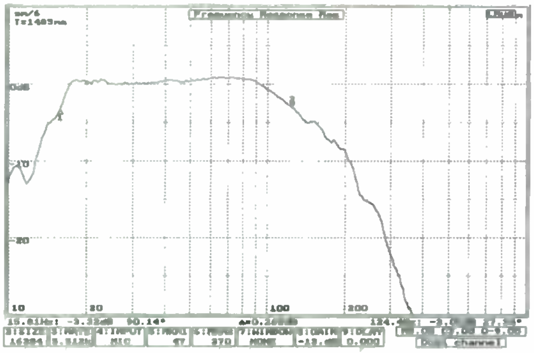

FIGURE 33: 10207 response in 2.5ft^3 closed box, with servo and 20Hz and 150Hz filters.

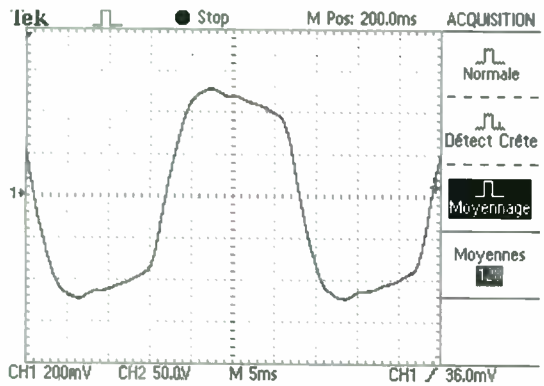

FIGURE 34: Close-field 30Hz square-wave response of the servo sub.

- external dimensions 16″ × 12″ × 24″;
- butt-joint assembly glued and screwed with dry-wall screws;
- corner bracing and heavy face-to-face bracing with 2″ × 2″ solid oak;
- vertical edges rounded with a 1″ quarter-round routing bit;
- polyester filling, 1 oz per ft^3.

Very carefully caulk the enclosure to ensure it is airtight. This is important, first because leakage would change the phase response of the drivers and cause loop instability, and second since the 200W delivered by the LM12 can easily destroy the voice coils by mechanical bottoming. The air spring of the closed box is the best protection against such disaster.

Also, don't forget to caulk the back of the drivers' faceplates before installing them. You can recess the drivers' holes with a router to mount the drivers flush with the front plate, but this is not mandatory.

The voice coils are connected with heavy-gauge wire to female banana plugs located on the rear part of the top face. As already mentioned, the mikes are fastened to a U-shaped aluminum bar screwed in front of the 10207; hence, the mike-to-driver distance is about 3″ with the electrets, and you can reduce this with the much larger Beta 57s.

MAIN ENCLOSURE

The main enclosure is located on top of the sub enclosure, onto which it is attached with two ½″ machine screws fitted in two brass inserts. This allows for easy separation and transport. This enclosure is simply a rectangular box without a rear cover; you can build it in the following way:

- Cut a 54″ × 16″ panel of 1″ MDF for the front face. Locate the drivers' centers according to *Photo 4*, and make a recessed cut of adequate depth with a router fitted with a straight bit so that the various drivers mount flush. Round the edges of this panel (except the lower one that corresponds to the top of the sub enclosure) with a 1″ quarter-round routing bit.

- Prepare the glass bowls by cutting cable passages with a continuous-edge diamond disk, and glue them to the backs of the front plates with special glass silicone.
- Cut two 54″ × 12″ pieces of ½″ MDF for the sides, and two 14″ × 12″ pieces for the top and bottom.
- On the left side panel, prepare holes for the brass inserts that will support the power amplifiers' heatsink screws (two for the main amplifiers and two for the sub amplifier).
- Assemble the enclosure with a butt-joint technique, using glue and drywall screws. I added corner bracing to improve rigidity. If you wish, you can also damp the side panels with Acousta-Stuf.
- Once the box is assembled, you can screw the drivers in place (don't forget the silicone for the P17REX), install the amplifiers, and make the connections.
- The rear view of the prototype is shown in *Photo 4*. You can see the glass bowls of the P17REXs and the large heatsink on the left.

TUNING THE SYSTEM

In the following description of the adjustments to the system, when I mention a potentiometer with the pointer connected to one end, the phrase "turned to minimum" means the pointer is towards the end it is connected to (and this situation corresponds to maximum resistance.)

The first step is to have correctly adjusted power supplies. The only tuning points in all system supplies are the SMPS output voltage and the current limit.

Connect the 60V transformer, bridge, and capacitors to the regulators (check polarity), switch on the mains, and read the output voltage with a voltmeter; adjust P2 to obtain 30V. Then connect a 1.5Ω/300W resistor to the terminals and adjust P1 so that you just reach the trip point at which voltage falls. Do the same for all four regulators. You can then connect the controller boards to check whether everything is OK (all relays should close).

Now, connect the power supplies to the power amplifiers (all with P2 to minimum) and check again. You should see that relay RL3 on the controller board closes only after one or two seconds, due to the presence of the amplifiers' capacitors. If RL3 doesn't close, switch off immediately, for this indicates a possible short-circuit on the supplies.

Once this is OK, you can connect the rest of the supplies. This is rather long and tedious, but quite easy.

SERVO SUB

To adjust the servo sub, put P1 to mid, P2 to maximum, and P3 to minimum. Connect a 30Hz 1V pp square wave to the input, and switch on the mains power. With an oscilloscope, observe the voltage on U1D, pin 14. Slowly turn P2 to decrease its value, up to the point where the servo loop oscillates (you will hear it). Then, turn P3 and adjust for an optimal (small overshoot) waveform on the oscilloscope. You can adjust P2 later if necessary (level-matching with the P17REX). I recommend you let the woofers break in on a 10Hz low-level sine wave for eight hours before making those adjustments.

Figure 31 shows the response of the 10207 in the enclosure, which is very close to what Boxmodel predicted. You can see that in the 20–100Hz zone, the response follows the theoretical 6dB/octave slope, which is compensated by R42/C45.

Figures 32 and *33* show the response of the closed loop, perfectly flat from 2–150Hz, showing efficient loop action. Finally, *Fig. 34* shows the square-wave response of the sub for 30Hz input; little

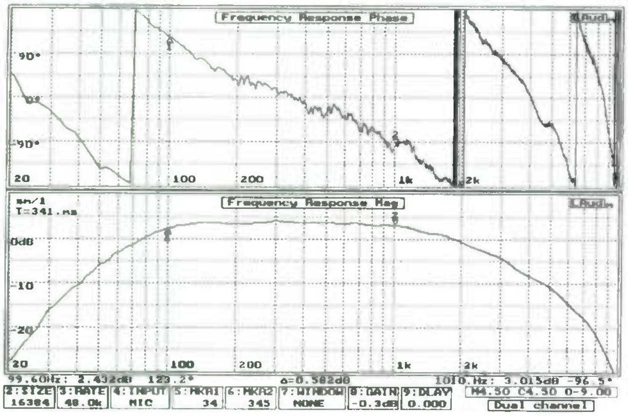

FIGURE 35: P17REX response, with gyrator correction without filters.

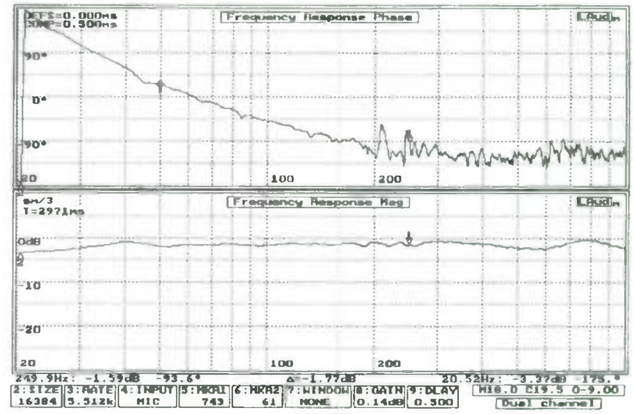

FIGURE 37: P17 and sub combined response.

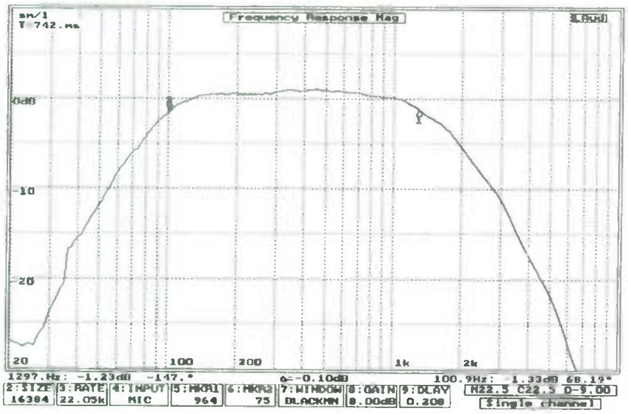

FIGURE 36: P17REX response, with gyrator correction with filters.

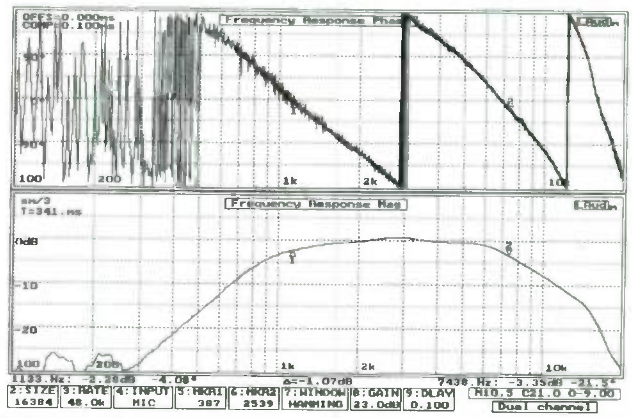

FIGURE 38: LPG50 response, with gyrator correction and filters.

overshoot indicates good loop phase margin. (Note that the main system filter contains a 20Hz high-pass filter, described previously; these figures do not include the influence of this filter.)

POWER AMPLIFIERS

Turn all P2 potentiometers to minimum and set all P1s to mid. Connect the power and switch on. For each amplifier, you should connect a voltmeter across R19 and slowly turn P2 until you read 100mV. Once that's done, connect the voltmeter between output and ground, and adjust P1 to null the offset.

Do the same for the 12 amplifiers, and

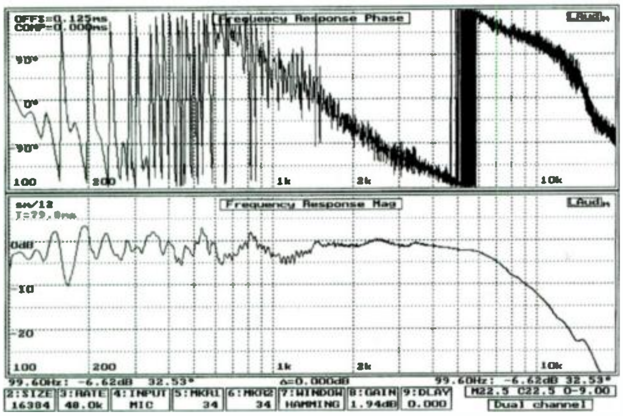

FIGURE 39: P17REX and LPG50 combined response.

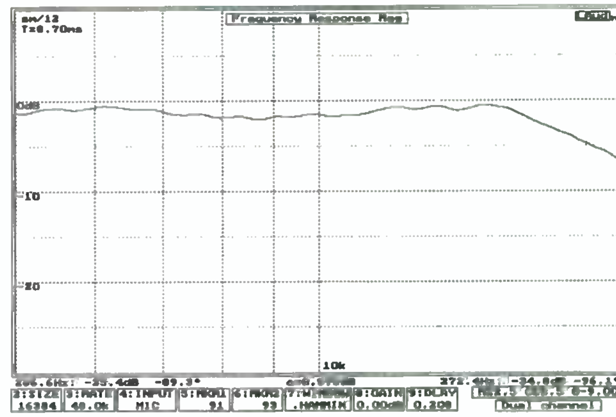

FIGURE 41: D250 1m response, combined with LPG50.

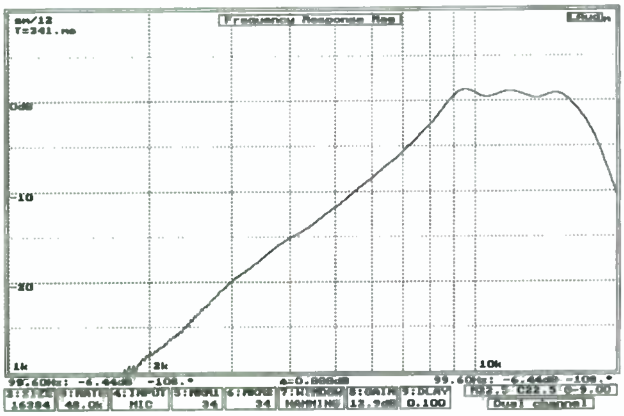

FIGURE 40: D260 response, corrected with filters.

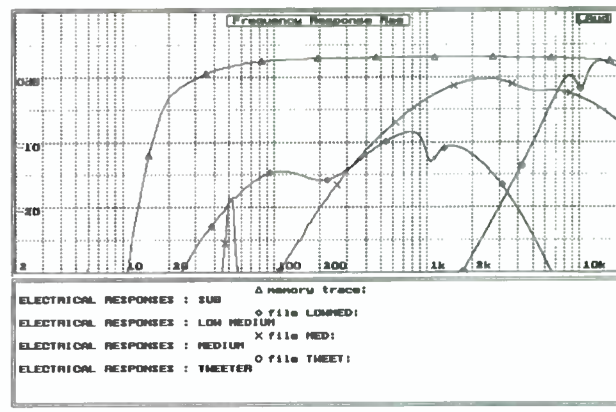

FIGURE 42: Filter electrical response for the four-ways (relative levels not to scale).

let it heat for one hour. Then make a new final adjustment, and it's finished.

FILTERS

Warning: you cannot tune the filters without a measurement tool (IMP, Audiosuite, MLSSA, or the like).

For all the next steps, connect the input of the filter to the signal output probe of the response analyzer, with a level of approximately 1V pp.

For the sub filter, connect the test probe to the "sub" output and measure the frequency response; adjust P1 so that the curve is perfectly flat before rolloff. If you see garbage on your screen, the value of P1 is too high, and the filter oscillates.

P17REX FILTER

Place the test mike 1″ from the P17REX, on-axis. Turn P2 and P3 to minimum, and P4 and P5 to maximum. Set P7, P8, P9, and P10 to mid. First adjust P7, P8, P9, and P10 to obtain the flattest possible response (*Figs. 35* and *36*); all adjustments are coupled together, but after a few minutes, you will be at ease with the process.

Note that P8 and P10 are related to the spectral sharpness of the response anomaly to be corrected, whereas P7 and P9 are linked to the center frequency of the anomaly.

Once the response is flat, you should connect the sub. Place your mike 3m from the enclosure, and measure the frequecy response from 20–200Hz. Adjust P2 on the servo board so that the level in the 50Hz region and in the 200Hz region are the same. When doing this, be careful to avoid any response peak or dip due to room interaction.

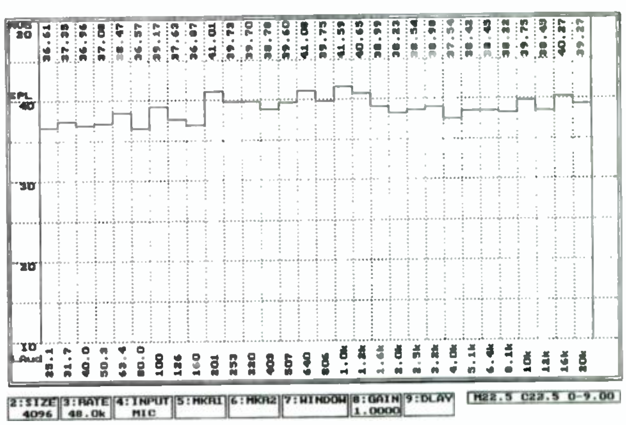

FIGURE 43: In-room ⅓-octave response, pink noise (measured at listening position, 4m).

Adjust the sub level. Now, you need a second mike. Place the first mike at 1″ from the P17REX center and the second in the same vertical plane, on one of the 10207's axes. Unless your measurement system features several mike inputs, you will need to breadboard a small summing amplifier that will add the signal of both mikes to enable double-source, close-field measurement.

Without touching the sub's level adjustment, adjust P2 and P3 until the summed response is flat (*Fig. 37*).

LP50 FILTER

Place the test mike 1″ from the LPG50, on-axis. Set P11, P13, and P14 to mid. Adjust P13 and P14 for maximum flatness in the 1–8kHz band (*Fig. 38*).

Now you must carry out the same summed response process as you did for the sub and P17. First, adjust P11 for equal levels at 400Hz and 3kHz, then adjust P4 and P5 for flat summed response (*Fig. 39*). Placing the two test mikes in the same vertical plane is very important in having correct relative phase.

D260 FILTER

Again, place the test mike 1″ from the D260, on-axis. Adjust P16 and P17 to the flattest possible response from 7–20kHz (*Fig. 40*). Again, if your tweeters are OK, you should be able to remove U13 and the associated components.

For adjusting P15, I didn't use the double-mike setup, since close-field measurement is not as representative of in-room response, mainly because of interaction between the three tweeters. I chose a 1m single-mike measurement, and adjusted P15 to achieve the flattest possible response (*Fig. 41*).

The final response of the tuned filter for my prototype is shown in *Fig. 42*.

IN ROOM

Measuring in-room response of multiple driver systems is always difficult, and is likely to produce strange results not in accordance with your ears' perception. Nevertheless, I have made a spatially averaged (30° horizontal window, 10° vertical) pink-noise measurement: the result is quite satisfying, as can be seen in *Fig. 43*. You should note that below 300Hz, I used a ⅓-octave equalizer to control room response.

It is also important to notice that the CALSOD predictions were quite accurate concerning vertical-angle response variations.

I know that some readers may be disappointed by the absence of 10Hz-resolution in-room response curves, but in my opinion they are almost meaningless for multiple driver systems.

REFERENCE

6. M. Rumreich, "Electronic Time Delay Line for Speakers," *SB* 3/88.

Kit Review

ACI TITAN SUBWOOFER KIT

Reviewed by Thomas Perazella

***Titan Subwoofer Kit,** Audio Concepts, Inc., 901 S. 4th St., La Crosse, WI 54601, $529.*

Audio Concepts (ACI), a company based in La Crosse, Wisconsin, produces both complete speakers and speaker kits. One of those products, the ACI Titan subwoofer kit, is its answer to the need for a high-performance subwoofer that appeals to the home builder. According to Webster, the definition of a titan is "any person or thing of great size or power." How does the ACI Titan stack up to this description? Let's find out.

Audio Concepts' philosophy for its kits is to provide the builder with the drivers and other components such as crossovers and amplifiers necessary to produce a finished product, less the actual housing or housing components. Plans for building a kit enclosure similar to those supplied with the factory-built units are available from ACI, but assembled and finished enclosures are not.

At first, this may appear to be a stumbling block for the inexperienced builder, but in reality, at least with the Titan, the enclosure design is relatively simple and straightforward. The design requires a 2ft^3 sealed box, which makes the construction simpler and more flexible in terms of tailoring the box dimensions to suit your needs.

Being able to design the shape of the box to fit into your decor is one of the main advantages of building your own enclosure. Need a tall shape with a small footprint? Build it that way. Need a low-boy shape to fit behind a couch? No problem. Want to use some exotic veneer for a drop-dead look? Do it. If woodworking is your hobby and you enjoy the time spent in design and construction, you can save money by building your own enclosure.

THE KIT ADVANTAGE

The assembled Titan has a retail price of $1599, but buying directly from the factory will cost you $799. The kit has a suggested price of $529, but my last visit to ACI's Website (www.audioc.com) showed it on sale at $499. Since the two versions are the same except for the enclosure, you can spend

PHOTO 1: Titan 12″ woofer.

the $300 savings on the materials of your choice, with any balance remaining in your bank account.

What do you get for your $499? The kit is built around a 12″, single-voice-coil woofer and a special mono-power amplifier rated at 250W. In addition to the woofer and amp, you get stuffing for the box, sealant tape for the woofer and amp, three RCA plugs with color-coded red and black leads to make your own attenuators for speaker-level amp connections, two spade lug terminals to connect the amp output to the woofer, and instructions for assembling the kit and using the completed sub.

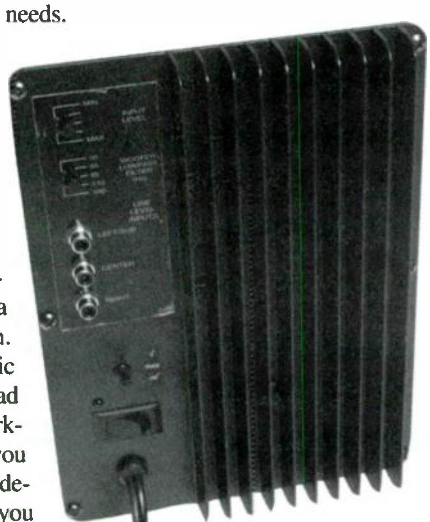

PHOTO 2: Titan power amplifier—front view.

THE DRIVER

The woofer has a stamped frame, rubber surround, large ceramic magnet, and long throw. I made no T/S-parameter measurements, but checked a few key items. Right out of the box, the woofer (*Photo 1*) had an f_s of 19Hz. To break it in, I mounted it to a baffle and drove it with a frequency of 19Hz at a level of 10V for over 40 hours, using a test amplifier (not the unit supplied with the kit). At the end of the break-in period, I rechecked the f_s and found it was 16Hz. Not a bad start.

During the initial tests, it became very clear that this is one tough woofer! Nearly any test signal that was even remotely sane failed to provoke the toe-curling sounds that portend imminent failure.

I noted dynamic offset at different frequencies, using a stroboscope set slightly lower than the test frequency. Viewing the speaker then reveals its movement in "slow motion." Below 30Hz, there was no visible offset. From 40–60Hz, an offset existed in the outward direction. Above 70Hz, it was in the inward direction. Maximum impedance was 31.3Ω at 16Hz, with a minimum of 6.5Ω at 82Hz—results consistent with an 8Ω nominal impedance. DC resistance was 4.8Ω.

THE AMPLIFIER

The amplifier is very interesting (*Photo 2*). It is typical of the newer sub amps in that all components are mounted on a rectangular plate, with all active circuitry on one side (facing in), and the heatsink, connections, and controls on the other side (facing out). Three RCA jacks are provided: left/sub, center, and right. These are line-level inputs for connections to a preamp output. For speaker-level signals, you must make resistive attenuators for connection to the terminals of your main amplifier or speakers.

In addition to the inputs, the amp contains a power switch (there is an auto-on feature as well), a phase-reversal switch, an input-level control, and an adjustable low-pass filter that is calibrated from 50–180Hz.

Major components on the circuit side are a hefty toroidal power transformer and two modestly sized filter capacitors. Two circuit

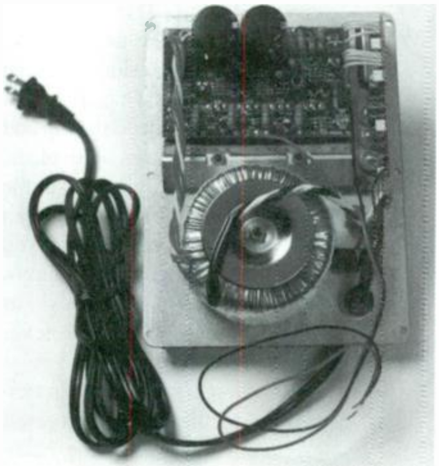

PHOTO 3: Titan power amplifier—rear view.

TABLE 1		
AMP TEST WITH 8Ω RESISTIVE LOAD		
FREQUENCY	INPUT LEVEL	OUTPUT LEVEL
100Hz	60mV	17V
100Hz	300mV	22V
100Hz	2V	23V
20Hz	60mV	12.5V
20Hz	120mV	13.8V
20Hz	300mV	14V
20Hz	2V	14V
10Hz	60mV	9.8V
10Hz	120mV	10V
10Hz	300mV	9.5V
10Hz	2V	8.7V

boards contain most of the active components, with the output devices hidden under a V-shaped bracket attached to the chassis/heatsink (*Photo 3*).

The literature states that the amp will deliver 250W and has an output that is tailored to reduce the drive levels at low frequencies as the average power level rises. This reduction at low frequencies prevents excessive cone excursion at high drive levels and low frequencies, reducing the chances of damaging the driver. Testing the amp confirmed the power reduction at low frequencies, but turned up some rather unusual characteristics.

TESTING THE AMP

I chose an 8Ω resistive load for the first test of the amp. The signal source was a low-distortion sine-wave generator, and I took measurements with an oscilloscope and a true RMS-measuring AC voltmeter. I set the amp's input to a crossover point of 180Hz and gain to maximum. I ran the tests at three frequencies and a range of drive levels. *Table 1* shows the results.

The data makes it clear that there is signal compression at low frequencies at all drive levels, and also that the amp cannot deliver

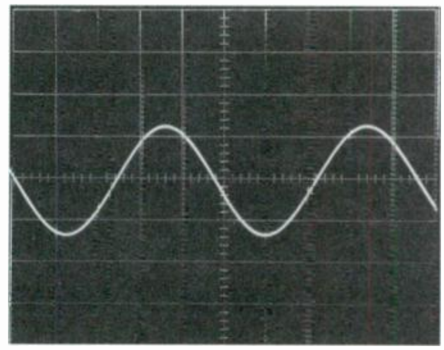

PHOTO 4: Amp output—100Hz, 10V/cm, no limiting.

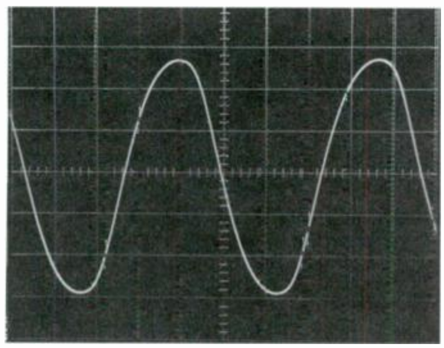

PHOTO 5: Amp output—100Hz, 10V/cm, limiting.

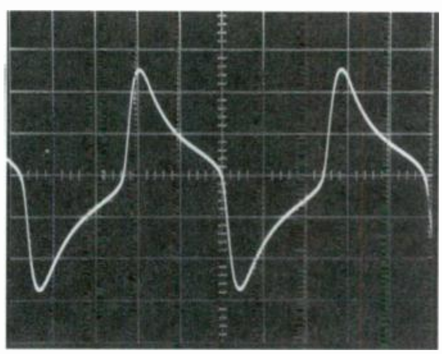

PHOTO 6: Amp output—20Hz, 10V/cm, limiting.

250W into 8Ω at any of the test frequencies. At 100Hz, where the output was highest, the amp still delivered only 23V, which calculates to slightly over 66W.

Photos of the oscilloscope traces show the limiting at 100Hz and 20Hz. *Photo 4* shows the signal at 100Hz before limiting occurs. Note the clean shape of the waveform. *Photo 5* shows the 100Hz signal at a higher drive level. The action of the limiting circuit is clear in the fall-off of the signal voltage as the peak is reached and passed. It appears that there is some sort of time/drive-level integration going on. The good thing is that the amp does not go into hard clipping, but exhibits a rather soft limiting.

Photo 6 shows the 20Hz signal at a drive level high enough to introduce limiting. The limiting is now quite pronounced, and although there is no hard clipping, the sharp bends in the curve at the maximum points indicate the production here of some higher-order harmonics.

Even if the power were rated at 4Ω and the amp were the perfect voltage source (that is, it could maintain the same drive voltage into lower impedance loads), the power into 4Ω would be double, or around 132W, well short of ACI's claim.

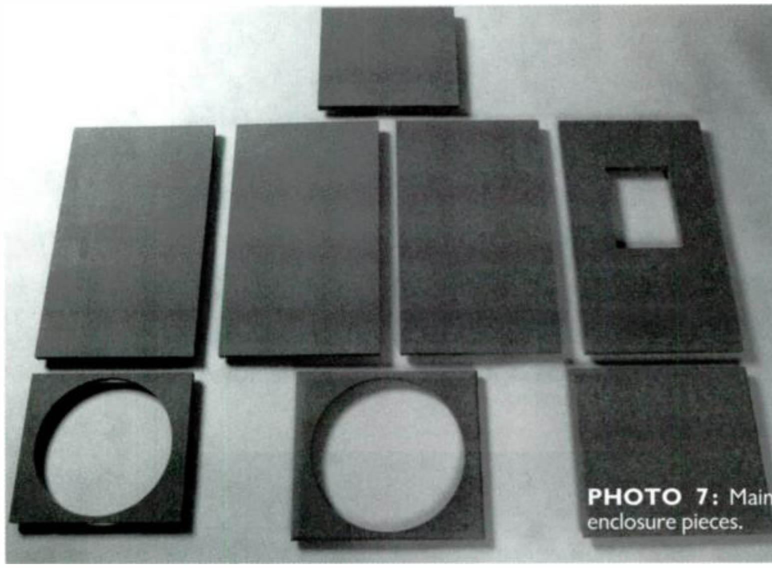

PHOTO 7: Main enclosure pieces.

To test the amp's ability to deliver the same voltage into lower impedances, I then connected it to a 4Ω resistive load. If it had ideal characteristics, the first test at 100Hz should produce the same 23V across this load, for a power level of 132W. Then came the big surprise: the voltage level into the 4Ω load was actually close to 38V at clipping, which represents around 360W. The trace also revealed that hard clipping into this load did occur. It had more the appearance of a conventional amp.

This trend was also apparent at 20Hz and 10Hz with maximum drive levels of 25V and 18V, respectively. Apparently, the control circuitry for the amp has a sensing function that increases the drive as the impedance decreases. Time to reevaluate.

With the speaker's minimum impedance being 6.5Ω at 82Hz, if you change the load bank to 6.5Ω and use 100Hz as a test frequency—since the limiting is less there—can the amp develop 250W? To do that, it would need to provide just over 40V. And the results? Nice try, but no brass ring. The output was 30V, or around 145W. This limiting will certainly help protect the driver, but it will also limit the output level, perhaps more than it should.

BUILDING THE ENCLOSURE

The next step was to build the enclosure. The configuration I used was that suggested by ACI, since it emulates the enclosure provided with the assembled Titan. This would be a good test, not only of the kit results, but also of their correlation with those of the assembled product.

ACI specified ¾″ MDF as the best choice of material, so I used it for the main panels and braces of the enclosure. Orientation is vertical, with the woofer firing downward from the bottom. The main housing is raised from a flat base by four dowel sections, 1″ in diameter and 1½″ long, to provide an exit path for the woofer output. The outside dimensions, less the base and supports, but including a ¾″ trim plate for the woofer, are 24″ H × 15″ D × 13½″ W. The base dimensions match the main housing.

The enclosure comprises eight main pieces, plus internal braces and the dowel spacers for the base. To simplify construction, I made all four walls 13½″ W × 23¼″ H, and the top and bottom pieces also equal in size, at 12″ W × 13½″ L. The base plate and trim plate for the woofer were both 13½″ W × 15″ L.

I cut all pieces, including the internal braces, from a single sheet of MDF. Before assembly, you need to cut holes in three pieces—the bottom, the woofer trim plate, and the back panel. That in the bottom plate, to which the woofer mounts, is 11 1/8″ in diameter, cut in the center. The trim plate, which fits around the outside of the woofer, has a hole 12¼″ in diameter, also cut in the center. For mounting the amplifier, the opening in the back plate is rectangular, 6″ W × 8 ¼″ H. Again to simplify construction, I rounded the edges only of the woofer trim plate and the base plate.

Photo 7 shows the pieces of the enclosure, without the braces, before assembly. They are (from top to bottom and left to right) the top plate, the four body panels, with the back panel on the right, the bottom plate, the woofer trim plate, and the base plate.

GLUING THE PIECES

I started the assembly by gluing both sides onto the front piece. Once the sides had set, I glued the top and bottom plates into place. Assembling even simple enclosures such as this can take a long time if you have only a few clamps. Think of clamps as money—you can never have too many.

ACI's drawings specified the internal braces. I cut them from the MDF sheet in 2″-wide strips, with lengths to match the inside dimensions of the partially completed enclosure, and then glued them into place. *Photos 8* and *9* are two different views of the enclosure at this stage. Note that the braces nearest the driver opening do not extend across the amplifier opening. This allows the amplifier to fit, unobstructed, inside the enclosure.

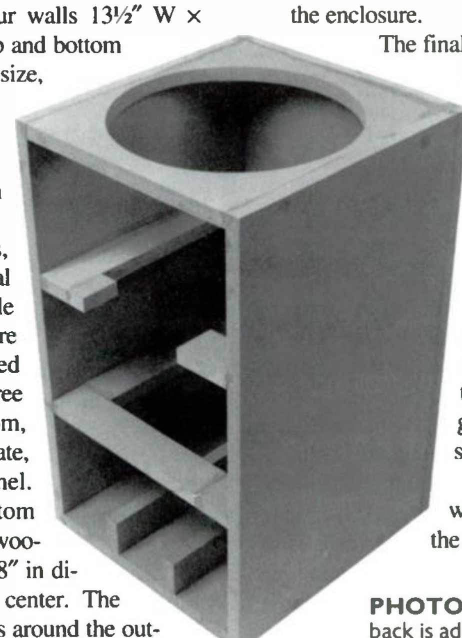

PHOTO 8: Enclosure before back is added.

The final major piece to add was the back. Before fixing it on the enclosure, I temporarily placed the power amp over the opening in the back and marked the mounting-hole positions. I then removed the amp and drilled the pilot holes for the mounting screws, positioned the back, and glued it in place as shown in *Photo 10*.

To make sure the base would fit properly with the enclosure, I aligned it

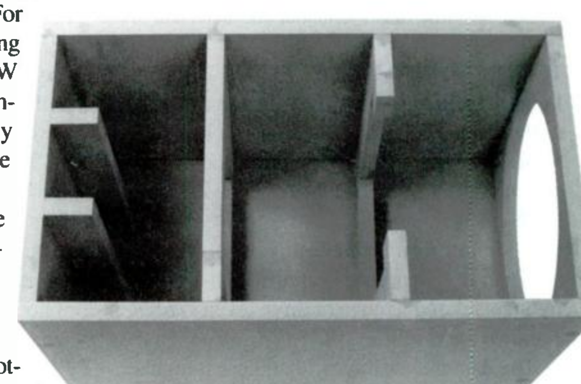

PHOTO 9: View into enclosure.

PHOTO 10: Enclosure without base and trim plate.

PHOTO 11: Enclosure with trim plate and woofer.

with the trim plate before assembly. I marked pilot holes for the base mounting screws and drilled through both pieces at the same time. Separating the pieces, I enlarged the pilot holes for the #10 × 3″ flat-head mounting screws in the base to provide clearance, countersinking them to allow the screw heads to rest slightly below the surface of the base. I also drilled the four dowel spacers with clearance holes for the mounting screws. Since I intended to paint the base and dowels black, I now primed them and set them aside.

FITTING THE TRIM PLATE

Before mounting the driver, I placed it on the bottom of the enclosure and positioned the trim plate to mount flush with all sides. If the driver holes were drilled without the trim plate in place, any misalignment would cause it to overhang one or two sides. Once I'd marked the pilot holes, I removed the trim plate and driver and drilled the pilot holes.

The plans supplied were not clear as to whether the trim plate was to be glued to the bottom or held in place only with the mounting screws. For the sake of additional rigidity, I glued it. The assembly directions also specify mounting the amp in place before the driver. I found it easier to reverse that sequence.

Before mounting the driver, I placed the supplied gasket material in position around the woofer opening. It is important to make sure there are no voids in the gasket material and that its inside edge does not extend past the mounting holes; otherwise there will be air leaks. I placed the stuffing material in the box and mounted the woofer. *Photo 11* shows the woofer mounted in the bottom opening of the enclosure.

I assembled the base to the enclosure by inserting the four mounting screws through the bottom of the base and then through the four dowel spacers. I aligned the screws with the pilot holes previously drilled into the trim plate and tightened them. *Photo 12* shows the mounted base.

The final assembly step was to connect and mount the amp. I crimped the supplied spade lugs onto the speaker leads from the amp, and then applied gasket material to the inside edge of the amp, being careful to eliminate any spaces that could produce air leaks. Finally, I placed the amp in the opening on the back of the enclosure and fastened it with six #8 × ¾″ screws. *Photo 13* shows the completed sub before the finishing process.

MEASUREMENTS

I made the acoustic measurements with an ACO Pacific model 7012 microphone capsule, an ACO PS9200 power supply/preamp, a custom interface switch/attenuator feeding a Turtle Beach Fiji capture board, and Liberty Instruments' LAUD V2.2 software running on a Pentium PC.

The first tests were nearfield bass response based on an MLS signal. I ran three output levels from approximately 90dB to 105dB maximum to determine the response from 10–400Hz and the effect of the amplifier's limiting circuitry. The amp crossover point and gain were both set to maximum values.

Figure 1 shows the frequency response with a level of approximately 90dB at 70Hz. The response is admirably flat from 30–100Hz, and is down 6dB around 23Hz. Response drops off rapidly below that point, although there is still some useful output below 20Hz.

Figure 2 shows the change in the response curve as the 70Hz level is increased to approximately 97dB. It is clear that the relatively flat response area between 30Hz and 70Hz that existed at the 90dB level is replaced by a more rapidly decreasing output below 70Hz. This is consistent with the stated performance characteristics from ACI.

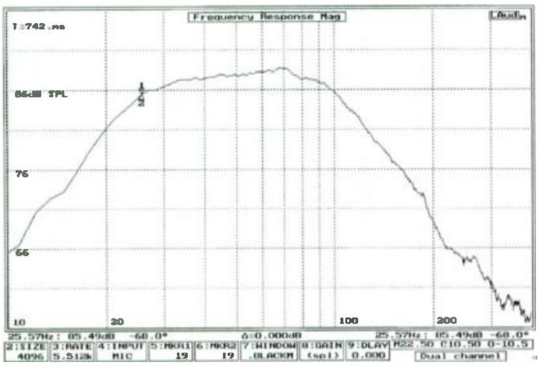

FIGURE 1: Frequency response with 90dB at 70Hz.

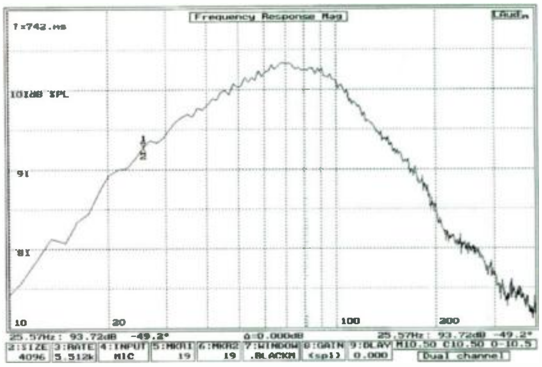

FIGURE 3: Frequency response with level increased to 104dB.

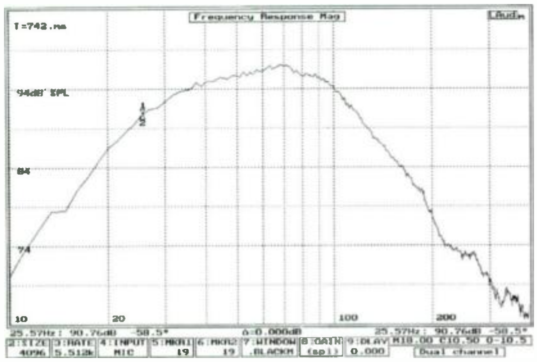

FIGURE 2: Frequency response with level at about 97dB.

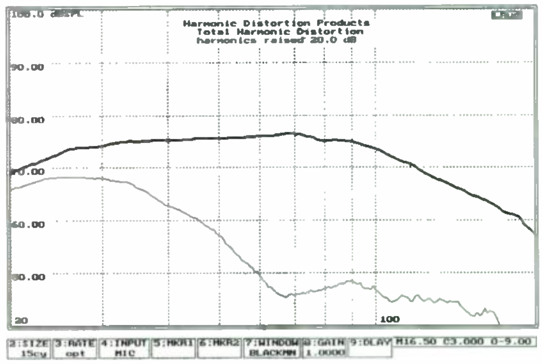

FIGURE 4: Distortion with a peak of 76dB at 70Hz.

In *Fig. 3*, the level at 70Hz was increased to approximately 104dB. The drop in output level below 70Hz makes this curve look almost like a classic second-order high-pass filter.

HARMONIC DISTORTION

I also measured harmonic distortion at three output levels, with a frequency sweep from 20–200Hz for each test. The resulting graphs show the output level as the heavier line and the distortion level as the lighter line. The distortion level is raised on the graph by 20dB, so where the two curves intersect, the distortion is actually down by 20dB, representing 10% distortion.

The first test (*Fig. 4*) produced a peak output of approximately 76dB at 70Hz. Even at 20Hz, where the output level was approximately 70dB, the distortion was below 10%, and at just above 50Hz, it dropped below 1%. These are pretty low output levels, but they are a good base line for comparison.

The story changed quite a bit when I altered the output level to give a peak of 88dB at 70Hz (*Fig. 5*). Because of the limiting action of the design, the low-frequency output increase was not as great as that of the upper bass. However, the distortion increase at 20Hz almost matched the level gain at the upper-bass frequencies. As a result, the percentage of distortion at 20Hz increased, exceeding 10% at 35Hz. Output at that point was around 84dB, so this is quite a bit of distortion for that frequency at that level.

The last test was done with a peak level of just under 100dB (*Fig. 6*). The level at 20Hz increased only a few dB, but the distortion increased markedly. The 10% distortion level now occurred at a relatively high 55Hz, and that at an output of around 95dB. The distortion level also remains fairly close to 10% for the balance of the sweep. This performance indicates that at this level, the Titan is acting more like a woofer than a sub, and not an especially good woofer at that.

LISTENING

Before making any measurements, I performed initial listening tests with a factory-assembled Titan that ACI sent me. I did the tests using the Titan in three different systems: my reference system, in conjunction with a pair of Sequerra Met 7s, and with a pair of Radio Shack Optimus LX5s with Lineaum tweeters.

My reference system consists of a home-made sub (see "True Bass" in *SB* 5/96) that has eight model DV12 12″ woofers from ACI in a 450ft^3 infinite-baffle enclosure (a loft closet), powered by a modified Phase Linear 400. Bass is handled by two 10″ CC line woofers from Peerless, each in a 3ft^3 infinite-baffle enclosure, and is powered by another Phase Linear 400. The midrange is provided by two 7″ Eton drivers mounted on a flat baffle and driven by a modified Hafler DH500 amplifier.

High frequencies are the domain of two Heil Air Motion transformers, mounted on the same baffle as the mids and driven by a Parasound HCA-800II amplifier. All drivers are directly connected to their respective amplifiers, with frequency division done ahead of the amplifiers by a custom 12dB/octave electronic crossover.

When I used the Titan with the reference system, I fed the drive from the electronic crossover to the input of the Titan with its crossover point set to the highest frequency. The main crossover provided the low-pass function, resulting in the best integration with the rest of the system. The Titan amp still provided the power and frequency/level limiting.

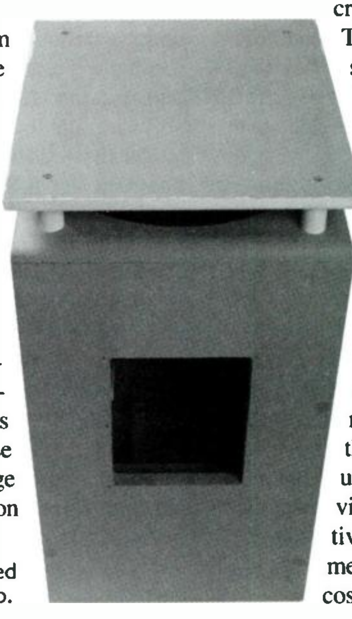

PHOTO 12: Completed enclosure without the amp.

SOME PERSPECTIVE

A bit of perspective is in order. Putting any sub into the reference system in place of the native sub requires you to understand that you should not view the results in a comparative light. Realistically, commercial subs, because of size and cost considerations, cannot pro-

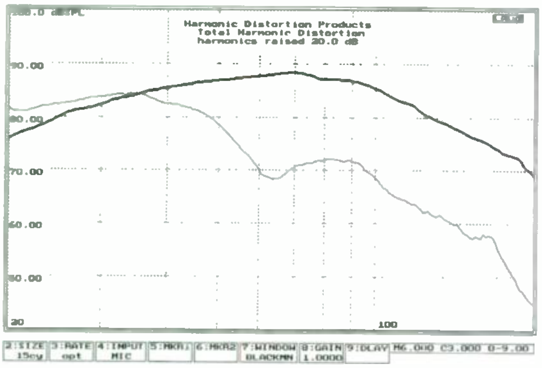

FIGURE 5: Distortion with 88dB at 70Hz.

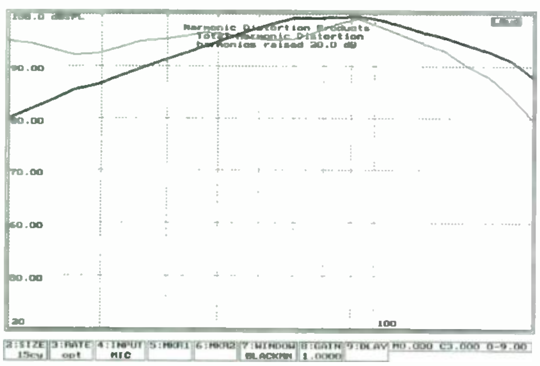

FIGURE 6: Distortion with a peak level of nearly 100dB at 90Hz.

vide the extension, low distortion, or output levels of the reference. The value of the substitution is that performing against a standard can help reveal differences in many parameters that otherwise might not be apparent.

I first auditioned the Titan at relatively low volume levels, similar to those you might use for casual listening while working. I was pleased with the results, since the Titan had a characteristic similar to the reference. There was even a taste of really low bass that I did not expect from such a small enclosure.

As I increased the volume, the Titan began to show its character. The taste of low bass previously present was now just a memory. There was bass, but not the real shuddering kind of low stuff that marks the heavy-duty players in the subwoofer arena. Most of the sins were of omission, rather than commission. There was still a lot of bass, but not as much of the very deep variety.

With volume levels measuring in the low 100dB range from the reference system (and it can play at quite loud levels with low distortion), the low bass really started to go away. The low-pass filter in the reference system was passing only frequencies below 45Hz to the sub, but the Titan attenuated them severely.

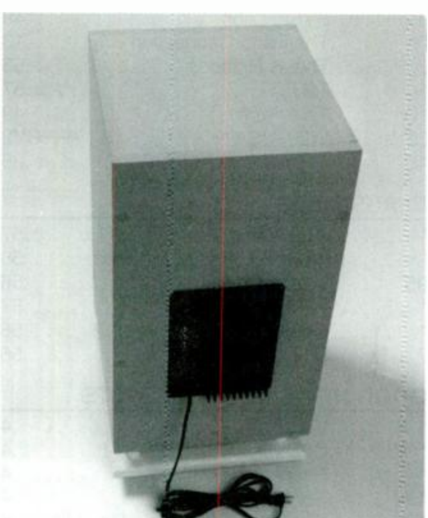

PHOTO 13: Completed kit before final finishing.

KIT TESTING

I conducted the rest of the listening tests with the kit version of the sub, first inserting it back into the reference system. As I recalled, the results were the same as for the factory-built version, but I did not conduct a side-by-side comparison, since I had already returned the factory version. Because most of you would be utilizing the Titan in conjunction with smaller speakers of the monitor variety, I did most of the listening with the Met 7 and Optimus speakers.

To take full advantage of the sub, I first passed the signal to the monitor speakers through a second-order high-pass filter to relieve the smaller speakers of the job of reproducing the low frequencies. For this task, I used a pair of Orban model 672A parametric equalizers having a variable high-pass function, with the output fed to a Parasound HCA-800II to drive the monitors. I used a wide range of CD source material for the auditions.

The Met 7s were the first partner for the Titan. I set the crossover point between the sub and monitor to 80Hz. The overall balance was quite good. When I was listening to the Eagles' *Hotel California*, my notes included "good drum sound," "clean midrange," and "punchy mid-bass."

I played two cuts from the *Brasiliero* album by Sergio Mendes. The first, "Fanfarra," has some explosive drum work. At low levels, the Titan did a good job. At higher levels, where the sound can become uncannily real if reproduced properly, the drums tended to recede. The second cut, "What Is

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from page 55
This?" has some strong low bass about 2½ minutes in. At higher levels, that bass got weak, but there was no sense of overload distortion.

Next, with the Talking Heads' "Psycho Killer," the lower passages were good, but the higher-level passages went flat dynamically. My notes included "no objectionable distortion" and "takes the heart out of the piece."

A very difficult CD for any sub because of wide dynamic range, low bass, and good bass definition is *Ibuki* from Kodo, from which I played two cuts. The first, "Akabanah," has some low drum notes at the beginning that sound almost soft, but very strong. The character of this piece was very different from the reference, with the fundamentals almost totally gone, hardening the sound. This piece was one of several that prevented the monitor/sub from approaching the reference system. The second selection, "The Hunted," does not have bass as low, but is very dynamic. The overall character was there, but the dynamics were limited.

Clark Terry's *Live at the Village Gate* has some good string bass work by Marcus McLauren. With the Met 7/Titan combination, that bass was a little fat. Sax on that album was good, however.

LOW BASS

For really low bass, I use two references, "Jurassic Lunch," from the *Great Fantasy Adventure Album*, and "Patchwork Über Einen Gregorianischen Choral," from *Stereoplay*'s *Avantgarde Edition*. The first has 12Hz bass during the initial T-Rex footsteps that are usually totally absent from most speakers. With the Titan, they were almost totally gone, being heard more as harmonics that felt like fundamentals. The second has a 16Hz organ that was again almost absent with higher harmonics present.

For use with the Optimus speakers, I raised the crossover point to 110Hz. Clean, deep, well-defined bass is not a strong point of that speaker. Unloading the bass duties from the little boomer made quite a considerable improvement.

The first test was the *Knock Out* CD from Charlie Antolini. The improvement in the drums was dramatic. Although the lower bass was a little weak, there was lots of upper-bass detail. The LX5s could never sound like that by themselves.

I next auditioned Garth Brooks' *The Thunder Rolls*. Well, the thunder didn't roll very much, but the bass guitar sounded good. Again, relieving the bass duty from the small LX5 bass/mid driver did wonders for the midrange.

Appropriately enough, the last two cuts were from the Pink Floyd CD, *The Final Cut*. There is a jet-plane flyby and explosion in "Get Your Filthy Hands off My Desert" that sounded OK at very low levels. However, very low levels are not what Pink Floyd is about. When played at higher levels, the deep bass was again shy. The upper bass was good, but a little on the fat side. With "Southampton Dock," the upper bass was again a little fat.

EVALUATION

Forming a single overall opinion of the Titan is difficult. Going back to Webster's definition, the ACI Titan is not a thing of great size. That's good. You can follow the ACI design and take up little floor space, or redesign as you wish, with not many cubic feet needed.

As far as being a thing of great power, certainly at low frequencies, the Titan is no titan. That is not to say that you cannot make significant improvements by matching it with a good-quality monitor, assuming that you relieve the monitor of the bass duties through a high-pass filter, preferably before the main amplifier. This is a generic benefit of a separate woofer or sub. However, the Titan behaves more like a woofer than subwoofer, especially when levels go up.

ACI's limiting philosophy has a very positive aspect. It seems to be nearly impossible to damage the Titan with any kind of even remotely reasonable input. Trying to push it into areas the designer did not intend it to go results in limiting rather than bad noises accompanied by smoke and flames. However, the limiting results in a taste of true sub performance at low levels, and then, alas, a loss of that low bass as you try to get to more acceptable listening levels.

As for the bottom-line benefits, you can build your own relatively small, simple enclosure any way you want, including your own choice of finish. The admission fee is not outlandish, if you don't count your own time. If this is your first attempt at building from scratch, the chance of success is pretty high, and your kids will probably not be able to blow it up.

Drawbacks? Although there are no "killer" subs in the price range of the Titan kit, there are some assembled subs that will do as much or more than the Titan that you can buy at store sales for the same price as the kit. Perhaps the biggest drawback is that once you get a taste of the low frequencies the Titan can produce at low levels, you'll never be satisfied with its performance at high levels. Because of the design, getting that low bass with multiple Titans will not be as cost efficient as either going to another design of your own or purchasing one of the other higher-priced commercial subs.

Despite all this, my time with the Titan was enjoyable, and if the benefits listed outweigh the drawbacks, you might wish to get in touch with the folks at ACI.

Manufacturer's response:

Thanks to Tom Perazella and SB for taking the time and providing the space for a detailed review of the ACI Titan powered sub. Tom does a very thorough job of testing for distortion and output. Unfortunately, I think Tom misses the point of the Titan. The Titan was never designed to be the loudest sub, just the best sounding. If you're looking for a sub that will play loudly in home-theater applications, there are lots of better alternatives. But if you're looking for a sub that will serve the music (for instance piano, or string bass), many have judged the Titan to be of reference quality.

The Titan has been in production for over four years. The assembled Titan is supported by a money-back guarantee, yet of all the assembled Titans sold, less than 1% have been returned for a refund! As for the other 99%, the buyers have carefully compared the Titan to just about every other sub on the market and have decided to keep it.

We've sent out five Titans for review. Tom Perazella and Tom Nousaine of Stereo Review *gave it lukewarm reviews on the basis of output. However,* Stereophile *gave the Titan high marks and placed it in Class C of recommended components (the least expensive sub in Class C). John Potis at* SMR Home Theater *gave the Titan his highest recommendation and bought the review Titan we sent him as his new reference. Greg Smith at* Soundstage! *also loved the Titan and purchased his as well! Anyone with experience in this industry knows reviewers don't buy products unless they really like them.*

Tom doesn't even discuss three aspects of the design that are responsible for the sonic quality of the Titan. First, the limiting system integral to the Titan not only makes it virtually bulletproof, but it is a main reason that the Titan can be seamlessly integrated with "difficult" speakers, such as electrostatics or planar magnetics. Many customers have told us that the Titan is the only sub they could seamlessly transition to speakers with which they'd unsuccessfully auditioned lots of other subs.

Second, the low-Q system design (less than 0.6) means it has less phase shift and greater transient accuracy than subs with higher output and higher Qs. Many power subs use higher Q as a way of gaining extension and output.

Third, the Titan cabinet design has several advantages over that of many subs. It is very solid and nonresonant. Its 70 lbs is more than the weight of many larger subs—just read the subwoofer evaluations in Stereo Review. *This solidity and nonresonance means there is far less out-of-phase "cabinet talk" to muddy the upper bass and transition to the main speakers. Also, its relatively small size, proportions, and construction make it an ideal end table—far easier to integrate aesthetically into a room than a big black box. Finally, you can build it yourself, which will save you money and allow you to customize it to your requirements.*

Is the Titan the sub you want to build? If you want extremely high output and have the room to make a big box, you can probably come up with better alternatives. However, if you're looking for the highest sound quality, particularly on music, the Titan should be a prime candidate.

Ultimately, we believe you should decide for yourself if it's the right sub. But what about a kit for which you've already built a cabinet? Well, we can't take parts kits back, but we can make arrangements for you to try a demo assembled Titan to first confirm it has the sound quality you want. Just E-mail, call, or write.

Mike Dzurko, President, ACI
audioc@lse.fullfeed.com, http://audioc.com

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CROSS MEASUREMENTS

This letter concerns Joe D'Appolito's measurements of cross-modulation distortion on the Aria, *SB* 6/98, p. 53, and on the Audax, *SB* 4/98, p. 48. On the Aria, he measured 0.3%, considered this a high value, and offered the following explanation, "But in practice, cross-coupling in the crossover networks and the common return wire introduce low frequencies into the tweeter, and high frequencies into the woofer. This makes for biwiring. On the Audax, he measured 0.08%, and considered this an acceptable value.

On reading the text describing these two speakers, I note that the Audax has separate high-pass and low-pass filter boards, mounted on opposite sides of the cabinet. The Aria, on the other hand, has one filter board on which all of the crossover components are mounted; neither is described as being biwired.

Of the two reasons offered for the Aria exhibiting a relatively high value of IM distortion, lack of biwiring does not appear to be the cause. The other possible reason offered: cross-coupling in the crossover networks appears to be the more likely. (Related readings include "Cable and Sound Delivery," P. Newell/K. Holland, *SB* 6/91; and "Magnetic Crosstalk in Passive Crossovers," Mike Chin, *SB* 5/90.) Does Mr. D'Appolito have any comments?

David J. Meraner
Scotia, NY

Joe D'Appolito responds:

It is standard practice when biwiring to use separate boards for the low- and high-frequency crossovers and to place these boards well apart. Any other approach would potentially negate the benefits of biwiring. Thus biwiring as practiced eliminates both causes of interdriver coupling.

TWEAKING THE TWEETER

The Reference Monitor project in *Speaker Builder's Loudspeaker Projects #1* (p. 37) is of interest to me; however, the crossover plans do not seem right.

The project offers the option of using either the Scan-speak D2905/9900 or D2905/9700, and provides the different component values of C2 and L2 if the Revelator D2905/9900 is used. Figure 2 (circuit diagram of crossover network) gives the C2 and L2, as well as the R1 and R2 values for the D2905/9700 tweeter. The R1 and R2 resistors form a fixed L-type attenuator in the tweeter circuit to match the output level of the tweeter (89dB for the D2905/9700 and the 91dB D2905/9900) to the woofer, which is 89dB. As you can see, the dB (sensitivity) of the two tweeters is different, with that of the D2905/9700 being the same as the 18W8546/01 woofer. Therefore, the R1 and R2 L-pad would not be needed with the D2905/9700, yet it is included in the diagram.

I found the driver specifications through my own research, as well as what the crossover network's various components represent. The article does not even give the sensitivity for the monitors, something that would be very useful to those trying to match the speakers with their amp. How did the designer intend the crossover?

In addition, the crossover is a single-wire setup, yet Figure 3 shows a biwired binding post, and what looks like a single-wire completed crossover mounted on a PC board utilizing low-cost inductors and caps. I would think that anyone who would use such expensive drivers as in this design would want to use high-quality low DCR air-cored inductors and polywhatever film in foil caps. Hard wired! Yet again, the text makes no reference to any such thing.

In the construction plans (Fig. 1), the size of the tweeter cutout is for the D2905/9700; the Revelator has a larger mounting flange, which would not fit in conjunction with the cutout or the 78mm×100mm axis for such a cutout. Judging from the photo on page 37 of the finished speaker with the Revelator tweeter, the mounting axis is different than that given in Fig. 1, yet again there is no reference to this in the text.

Marc Price
actinic@hotmail.com

Rolph Smulders responds:

1. The reflex port has an outer diameter of 50mm, an inner diameter of 42mm, and must be filled with straws.

2. In the tuning of a BR system, the DC resistance of the series inductance is taken into account. Mr. Price's suggestion of lowering it by 0.16R does not appear to serve any purpose to alter the port dimensions on the basis of computations. This is because the consequent lower value of Q_e will lie within the manufacturing tolerances of the 18W8546. It would become useful only if the units were measured individually and the enclosure tuning optimized on the basis of the T/S parameters so determined.

3. Lowering the DC resistance of the shunt inductor for high frequencies is not advisable, since it would increase the Q of the circuit with the series capacitor, which might lead to spurious oscillations.

4. The quality of the Audyn capacitors is certainly on a par with that of Hovland or Multicap capacitors.

5. A S.E.T. with the 300B delivers relatively low power. Listening tests have shown that the Reference Monitor performs much better with a higher-power amplifier.