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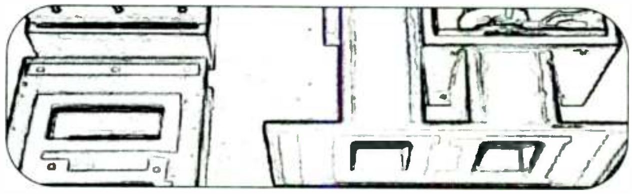

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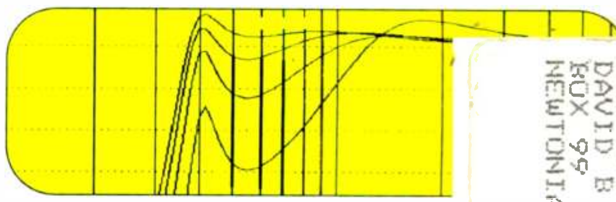

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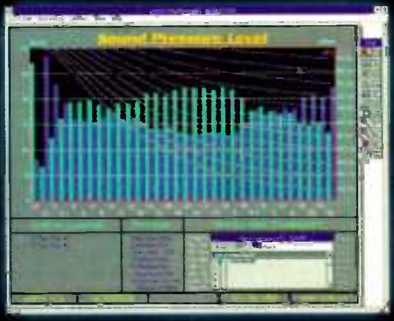

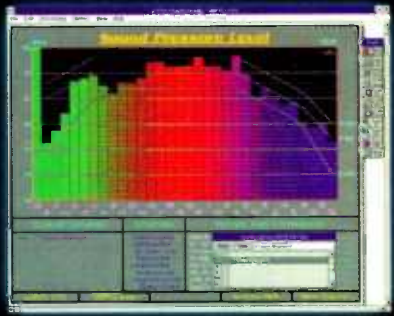

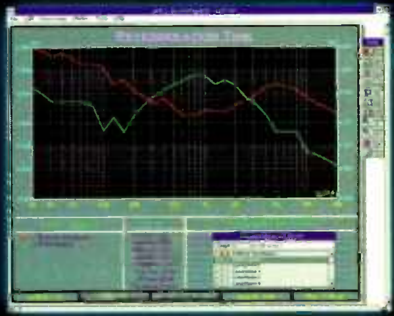

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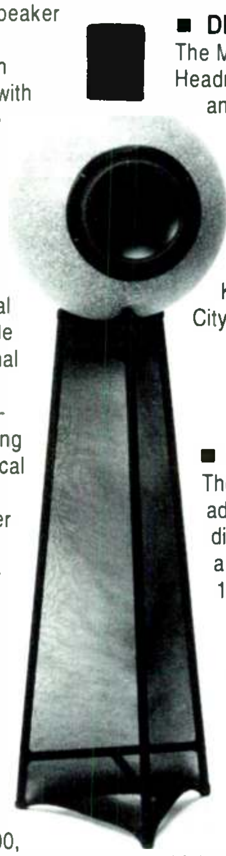

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The MX-125 and MX-145 powered subwoofers utilize the Headroom Maximizer II™ circuit for increased output and full signal dynamics without amplifier clipping. The push-pull dual-driver configuration significantly reduces even-order harmonic distortion and doubles the sound power per watt of amplifier power. The MX-150TX 150W subwoofer is the first in a series of affordable Home THX speaker systems to be introduced over the coming months. Miller & Kreisel Sound Corp., 10391 Jefferson Blvd., Culver City, CA 90232, (310) 204-2854, FAX (310) 202-8782.

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The C803A, an improved version of the C803, was engineered to meet the demands of new and advanced signal sources. The 16W unit offers a frequency response of 60Hz–18kHz, ±6dB, and a 150° dispersion pattern. The 8″ low-frequency reproducer and 3″ high-frequency reproducer are coupled via a built-in crossover network. The C803A is equipped with a post-mounted tweeter. Atlas/Soundolier, 1859 Intertech Dr., Fenton, MO 63026, (314) 349-5702, FAX (314) 349-1251.

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The System 350 is the only THX®-certified home theater speaker system with a low-profile, horizontal center-channel speaker. Two 200W powered subwoofers, three-way front speakers, TwinPolar™ surrounds, and optional pedestal stands are included. The System 220 incorporates a 125W powered subwoofer with optically isolated anticlipping circuitry, a center-channel speaker with timbre control, high-output front speakers, and dipolar surrounds. Atlantic Technology International, 343 Vanderbilt Ave., Norwood, MA 02062, (617) 762-6300.

Reader Service #102

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In Memoriam

PETER J. BAXANDALL 1921-1995

We received word recently that Peter J. Baxandall, 74, died early last month. Only a few biographical details of this British engineer/audiophile's life were available to us at press time:

- He earned an engineering degree from Cardiff Technical College in 1942.
- After graduation, he became an instructor in Fleet Air Arm radio training during the war.
- In 1944 he joined the Telecommunications Research Establishment (later changed to the Royal Signals and Radar Establishment) in Malvern.
- He retired from RSRE in 1971 to become an electroacoustical consultant.

Early in his career, Mr. Baxandall was involved in quality audio circuitry. He was influenced by prominent designers and people whose names became quite well-known in the audio field, such as Paul Voigt, Peter Walker, Cecil Watts, and others.

One of his loudspeaker projects, Mark I, appeared in the early issues of *Audio Amateur*, over 25 years ago (*TAA* 2/3 1970). This small, elliptical driver device was remarkable for providing the best possible sound reproduction for the lowest possible cost.

We recall a pleasant meeting with this distinguished gentleman in 1979. The results of our visit are recorded in the two-part "Conversations with Peter Baxandall" (*TAA* 4/79, p. 12, and 1/80, p. 27).

He has several noteworthy audio designs to his credit. For many years he was associated with *Wireless World* magazine and published some of his work, including a noted amplifier project in January 1948, in that publication.

But Mr. Baxandall will probably best be remembered as the designer of the negative feedback (Baxandall) tone control, which scheme has been widely used in audio. We will miss his talent and contributions.

Speaker Builder

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

❧ John Stuart Mill

Speaker Builder is published eight times a year in the interest of high-quality audio reproduction.

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Guest Editorial

A LAMENT FOR OUR FUTURE

As a professor of acoustics and audio engineering, I routinely receive phone calls from loudspeaker manufacturers and headhunters asking for names of graduating students who would be qualified to interview for a job in this field.

In recent weeks I have received many more phone calls than usual. Each caller seems to have the same story: we need transducer engineers. We need MEs, EEs, manufacturing engineers, and materials engineers who know about transducers, and we need them now. Business is booming. What have you got?

I would not be writing about this if I were not startled by the frequency and the intensity of these phone calls.

Some weeks ago the *Boston Sunday Globe* published a one-third page advertisement from New England's largest consumer loudspeaker manufacturer seeking applicants for these positions. Then I saw another, similar ad in a trade journal from New England's largest professional-audio loudspeaker manufacturer. Is this a crescendo, or what?

Where have all the transducer engineers gone? Retired? Changed jobs? Started their own businesses? And who will replace them?

The last question has no clear answer, even though microphone and loudspeaker manufacturers lament for one. Transducer design is an intriguing engineering discipline in which art plays a role nearly equal to science. That mix conjures up apprenticeship and the image of a dottering engineer holding with shaky hands some fragment of a loudspeaker assembly explaining to the new hire why that little spacer (which, of course, does not show up in any equation) must be there.

The new hire graduated with distinction from engineering school and is now in real-world kindergarten learning how to be useful to the boss. The boss is happy to have this kid on the payroll for a whole bunch of reasons, but the mind reels at the thought of a $90,000 education investment requiring yet another $30,000 education investment from the company coffers to complete "retraining."

You could argue that we educators ought to be doing a better job at this, but there are limits to what can be accomplished in four, or even five, years of engineering education. A graduating student needs to hold a degree in one of the major disciplines to be "marketable," and the school needs to furnish such a degree under the dictum of some Accreditation Council in order, legitimately, to attract students in the first place (this also applies to graduate school).

If the latter accountability allowed for it, what would the "transducer engineering" graduate study in a model program? To begin with, he/she must know about:

- magnets and magnetic interaction;
- how to get a flimsy coil of fine wire to survive years of repeated, exceedingly high peak acceleration at high temperatures;
- heat flow;
- which adhesives will hold the assembly together without creep over a wide temperature range with high mechanical stress;
- aging and fatigue in rubber compounds;
- the longevity of electret foil surface charge;
- producing and analyzing electrical analogs for the moving mechanical parts;
- translating the acoustical requirements into the mechanical design.

Anyone who is knowledgeable in the art can easily add a few more to this list.

There is little opportunity in a "traditional" four- or five-year electrical engineering program to inject any more than two items from the above list, any one of them probably raising faculty eyebrows. Mechanical engineers fare better, except that they usually lack the training to compute electrical analogs, a crucial analytical skill in this field. A "nontraditional" approach is a leap of faith for many schools—an acknowledged impediment.

The socioeconomic forces currently defining the welfare of engineering schools have caused them to discover that diversity, interdisciplinary studies, a big dose of humanities, and flexibility in course selection bring in freshman applications and sustain the student population at a financially acceptable level. Some very creative programs have come into existence as a result.

So perhaps it's time to create a BS or, more likely, an MS in Electrical or Mechanical Engineering with emphasis on Transducer Technology (does it exist anywhere, now?). Such a degree would formalize the intellectual wedding of materials science, chemical engineering, thermodynamics, mechanical engineering, electrical engineering, and acoustics, and would concentrate specifically on electrodynamic and electrostatic transducers. If I had 12 such graduates right now, I could place all of them in jobs in a week.

I asked my department chairman for comments on this editorial, and his answer reflects the thoughts of a man in a higher orbit and with a broader perspective than I:

"...I am frustrated at the situation in which corporations tell universities what they need in engineering graduates, but provide no help in meeting those needs. In other words, corporations treat engineers as their stock of materials. They expect a cheap, plentiful supply that matches their specifications for today's job (and higher education has cooperated because it is in our economic interest to collect tuition from as many students as possible). The profession of engineering... needs to stop accepting that situation as the status quo."

Grist for your mill, Dr. Faustus.

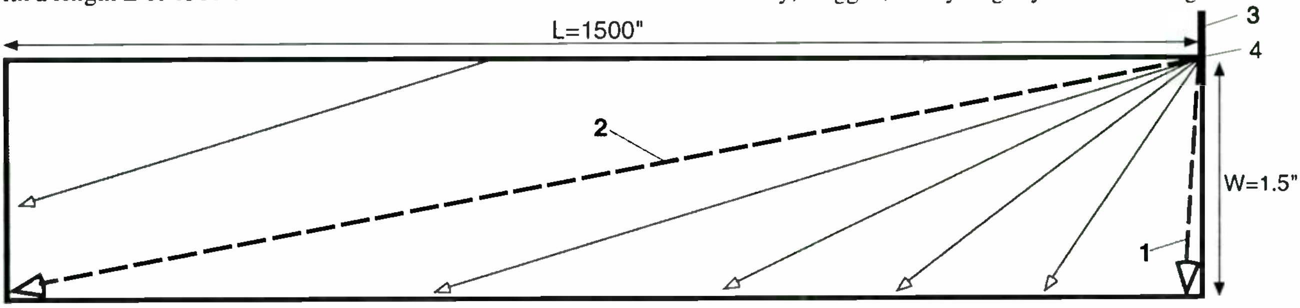
Dick Campbell
Contributing Editor

ADVERTISEMENT

What's Wrong with Your Capacitors?

For any capacitor to work, electric charge must get from the terminal to all parts of the plate. As soon as electric current from terminal 3 (in diagram below) first enters the plate, it begins fanning out from point 4 to all parts of the plate, following various paths including the diagonal paths shown. In your audio system's larger capacitors, the plate might have a width W of 1.5"with a length L of 1500".

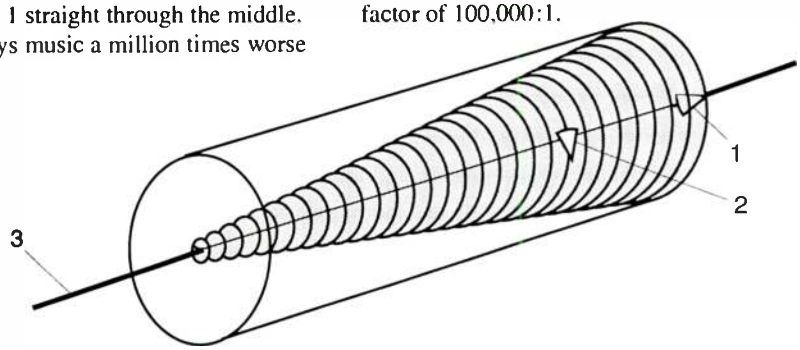
This creates a bad problem. Signal path 2 below is over 1000 times longer than path 1. So your music doesn't all get through the capacitor at the same time. Some music gets through quickly, via path 1, but some of the same music signal takes 1000 times longer, via path 2. This time smears your music, so it sounds fuzzy, defocussed and veiled (especially in the treble) — and muddy, clogged, honky or glary in the midrange.



Your capacitors have another problem that's even worse. They roll up signal path 2 into a tight corkscrew coil, as shown below. The inductance of any coil is multiplied by the number of turns squared, and there can be 1000 turns in an audio capacitor. So the inductance of path 2 can be a million times higher than the inductance of path 1 straight through the middle.

This inductance disparity delays music a million times worse for some paths through the capacitor than for other paths. Some parts of your music arrive 1,000,000 times later than other parts. This capacitor actually smears music by a factor of 1,000,000:1.

Multiple capacitors with 10 coaxial sections slightly reduce these problems, but merely by 10. So they still smear music by a factor of 100,000:1.



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- transparent, open, and airy — instead of veiled or clogged;
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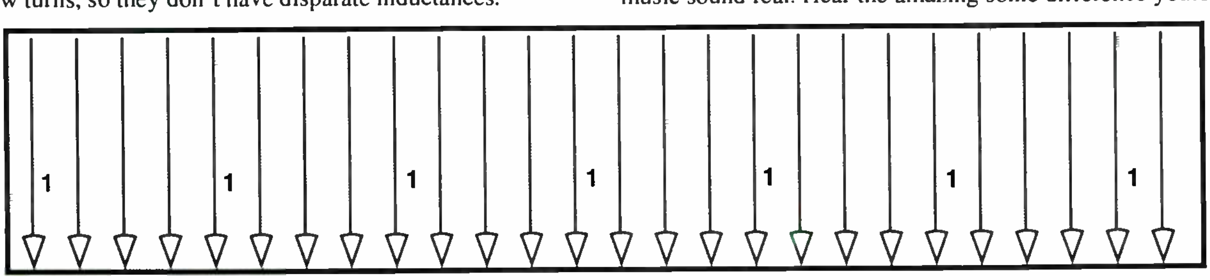

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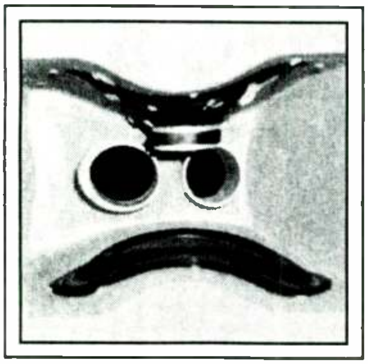

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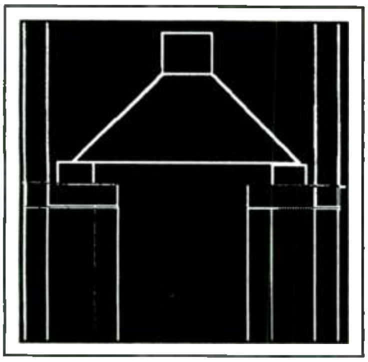

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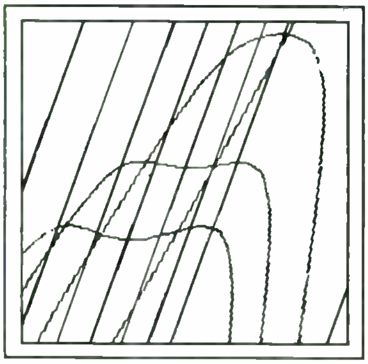

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DEPARTMENTS

A SELF-POWERED SUBWOOFER FOR AUDIO/VIDEO

By Jerry McNutt

As usual, the Enterprise was in a hopelessly desperate situation. Its warp engines were down, and the evil "Q" was about to take his revenge on Capt. Jock Loc LeFroc and crew by sandwiching the ship between two giant slices of buttered bread and feeding the entire mess to a black hole he had befriended.

Down in the engine room, Jordy was munching on some popcorn while trying to figure out the problem with the warp engines. He choked on the last bite and coughed so hard his Fram air filter mask fell off, allowing him to see that the ship's ignition switch was turned to "accessory" instead of "run." He quickly turned the key to start the warp engines and radioed the captain that the ship could go to warp speed. The captain stood and commanded, "Make it so!"

The ship responded and squirted out from between the slices of bread, thus making the black hole's snack a little less tasty and ensuring there would be a ship available for next week's episode. All were happy and jubilant. All but the Q and I.

PHOTO 1: AV-1 with installed ports.

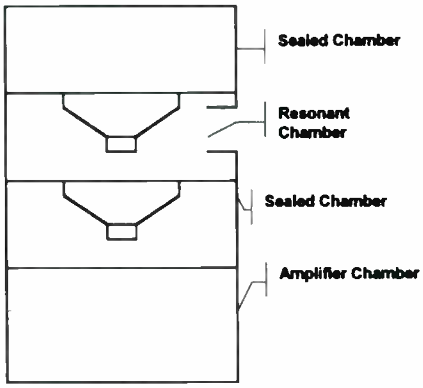

FIGURE 1: VS-1 push-pull configuration.

The Q had a legitimate gripe, so what was mine? Bass! This kind of action and fine scriptwriting deserve decent sound. I commend the producers of the show for its great soundtrack. I needed a subwoofer to bring it out, which is why I decided to build my VS-1.

DESIGN GOALS & CRITERIA

I decided to "give a little to get a little": 3dB down at 38Hz and a sensitivity of at least 94dB for a complete package less than 7ft^3. While 38Hz is not that deep, it is adequate for warp engines, door slams, or even an occasional whale call. Seven cubic feet is large, but the complete package includes woofers, amplifier, and crossover built into one box.

The prospect of building a self-powered sub has always intrigued me. It features no speaker wires and less DC resistance for better dampening. And no, you don't need to build a pretty cabinet for the amplifier.

WOOFER SELECTION & DESIGN

I decided to build a bandpass sub because it offers a variety of parameters to play with and optimize, one of the most beneficial parameters being the upper rolloff point. The 12dB rolloff inherent to bandpass designs decreases the need for a steep electrical crossover. In fact, I was not planning on using an electronic crossover at all. The bandpass design would take care of the unwanted upper frequencies.

After searching for weeks, and running many test equations, I chose the Seas 25FEWX/8, which has an F_S of 26Hz, a Q_T of 0.23, a V_{AS} of 175 liters, and a sensitivity of 91dB. It is incredibly well-made and handles lots of power (rated at 60W, but I have dumped much more and still do) considering its low cost. I decided to use a pair of these woofers in a push-pull configuration (*Fig 1*). Only two woofer parameters change when you run push-pull: the V_{AS} and sensitivity, which jump to 350 liters and 97dB, respectively.

I followed the design steps in Jean Margerand's *SB* 6/88 article, "The Third Dimension: Symmetrically Loaded." After checking the equations (see "Math" sidebar for details), I found a design that would satisfy my goals: 3dB down at 38Hz, speaker enclosure volume of 3.4ft^3, and a sensitivity of at least 94dB. At my request Ledale Reynolds ran the design through LEAP, and confirmed my results (*Fig. 2*).

ABOUT THE AUTHOR

Jerry McNutt is a 1990 Auburn University electrical engineer who currently manages a Radio Shack dealership on the cutting edge of high-end car audio. He has also run his own loudspeaker enclosure business, been a production engineer at an OEM speaker plant, and is the proud first-time father of a baby boy.

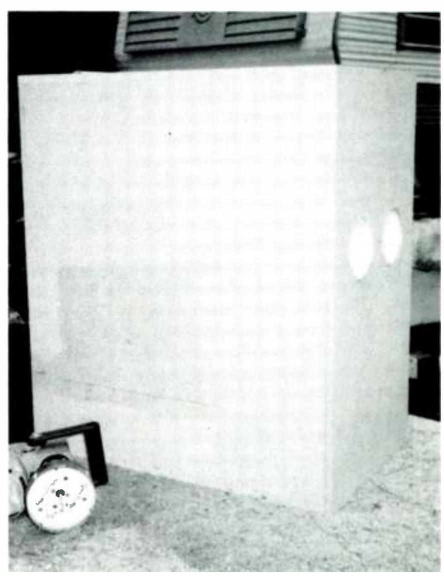

PHOTO 2: Sanded box ready for covering.

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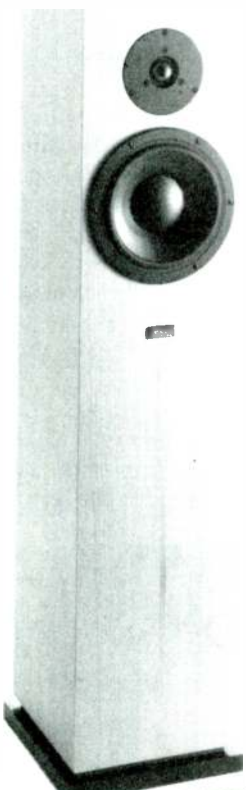
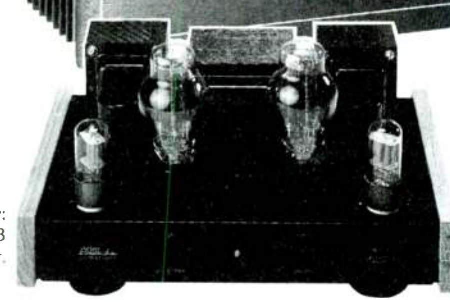


The Aries system showcases the newest Dynaudio loudspeakers: the Esotec 20W75 8 inch woofer, and the Esotec D260 tweeter, in a compact tower design.

The woofer uses a 75 mm voice coil, cast frame, rubber suround, and the tweeter is a chambered, textile 28mm dome unit. In combination with the 6 dB, 13 element filter, this system offers quick response, wide dynamic range, and transients charateristic of live performance.

The Aries system is solid, musical, and the prefinished oak cabinets are a pleaure to behold. It has the performance of systems priced at several thousand dollars per pair.

Audio Electronic Supply: SET-II, single-ended class A, solid state, 20 Watt/Channel Stereo Amplifier.



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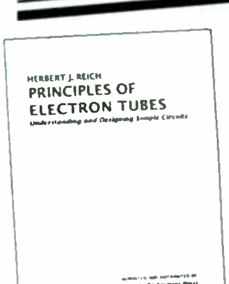

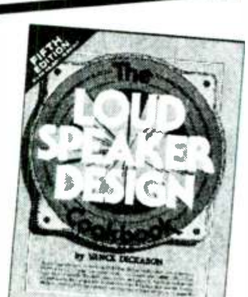

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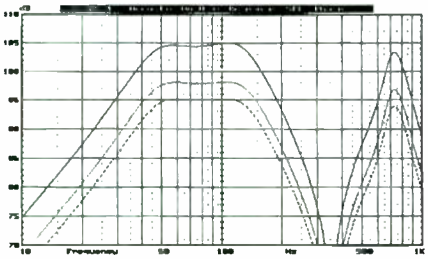

FIGURE 2: VS-1 measurements (solid = 10W input, dotted = 2W input, dash = 1W input).

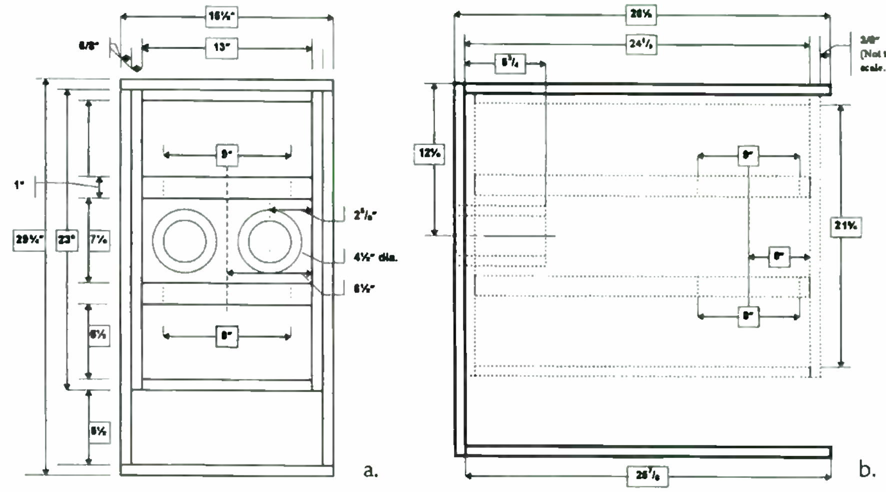

FIGURE 3: Cabinet dimensions a: rear view, back removed; b: side view, back installed.

CONSTRUCTION & FINISHING

I prefer to use ¾″ Medite™ for all my speaker projects, but I did not have enough on hand. The cabinet supply stores were closed during that Thanksgiving weekend, so I was forced to use (gulp!) 5/8″ floor underlayment grade particleboard, which I don't like, but it is better than plywood. Because of the cheaper wood, I decided to make double-thick cabinet walls in places. Luckily, I had enough 1″ Medite for the driver boards. *Figure 3* shows the final cabinet dimensions and details.

Building a cabinet with double-thick walls is easy if you use the box-around-a-box procedure. Simply build the inner box to the correct air size and basic shape. After rough-sanding all the edges and seams smooth, start building a second box over the first, one panel at a time, using a thin layer of glue between the panels. In this design, the outer box is taller to accommodate the lower compartment for the amplifier section.

Use a 4.5″ hole saw to cut the ports; schedule 40 4″ PVC pipe fits snugly. The ports should be glued in with epoxy. When cutting the ports to length (7″), allow 1/8″ extra to hang out of the cabinet. After the glue has set, use a belt sander with a 60-grit belt to sand the ports flush with the cabinet. *Photo 1* shows the installed ports, the double-layer construction, and woofer test mounting. (Be sure to cut woofer holes *before* cabinet assembly.)

After rough-sanding with 60-grit and fine-sanding with 100-grit (*Photo 2*), I was ready to cover the cabinet with walnut-grained Formica™. Starting with a side, use plenty of contact cement (brush on both cabinet and Formica) and let it dry until it is not tacky. I cut the large sheet of Formica with a small electric hand saw into easier-to-handle pieces, so they overlapped their corresponding panels by an inch or so in each direction to allow for positioning errors.

Place long, thin strips of wood on the cabinet every 4″ to support the Formica about ½″ or so above the surface to be covered. Once you are positive you have the Formica positioned correctly, pull out the centermost strip and press down the covering. Work your way to the edges by pulling out strips and pressing. This method will help prevent trapped air under the Formica.

After using a roller to ensure uniform adhesion, trim the laminate to fit with a Formica trimmer bit and a router. Repeat until the remaining side, top, and front are covered. Clean off any excess glue with acetone. Use a broad, fine file to knock the sharp edges from the Formica by holding the file at a 45° angle and working in only one direction; you will need to make only about two or three passes.

To uncover and dress up the ports, first drill a ½″ hole in the Formica covering the ports. Next, use your trimmer bit and router to rout out the ports. Change your bit to a

PHOTO 3: Finished cabinet.

PHOTO 4: Completed amp.

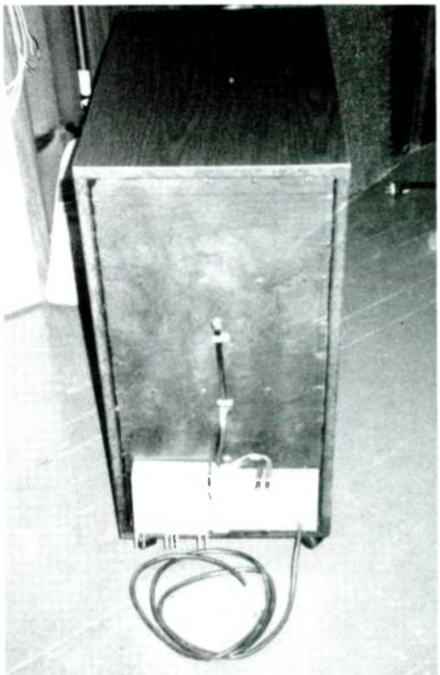

PHOTO 5: Completed VS-1.

Reader Service #52 →


* SEPARATE CAR SPEAKER SYSTEM
* IN-WALL SPEAKER * TWEETERS
Separate/Component Car Speaker System
4", 5 1/4", 6 1/2"
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Mitutoyo
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In-Wall/Ceiling Speaker
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ϕ 13/19/25mm
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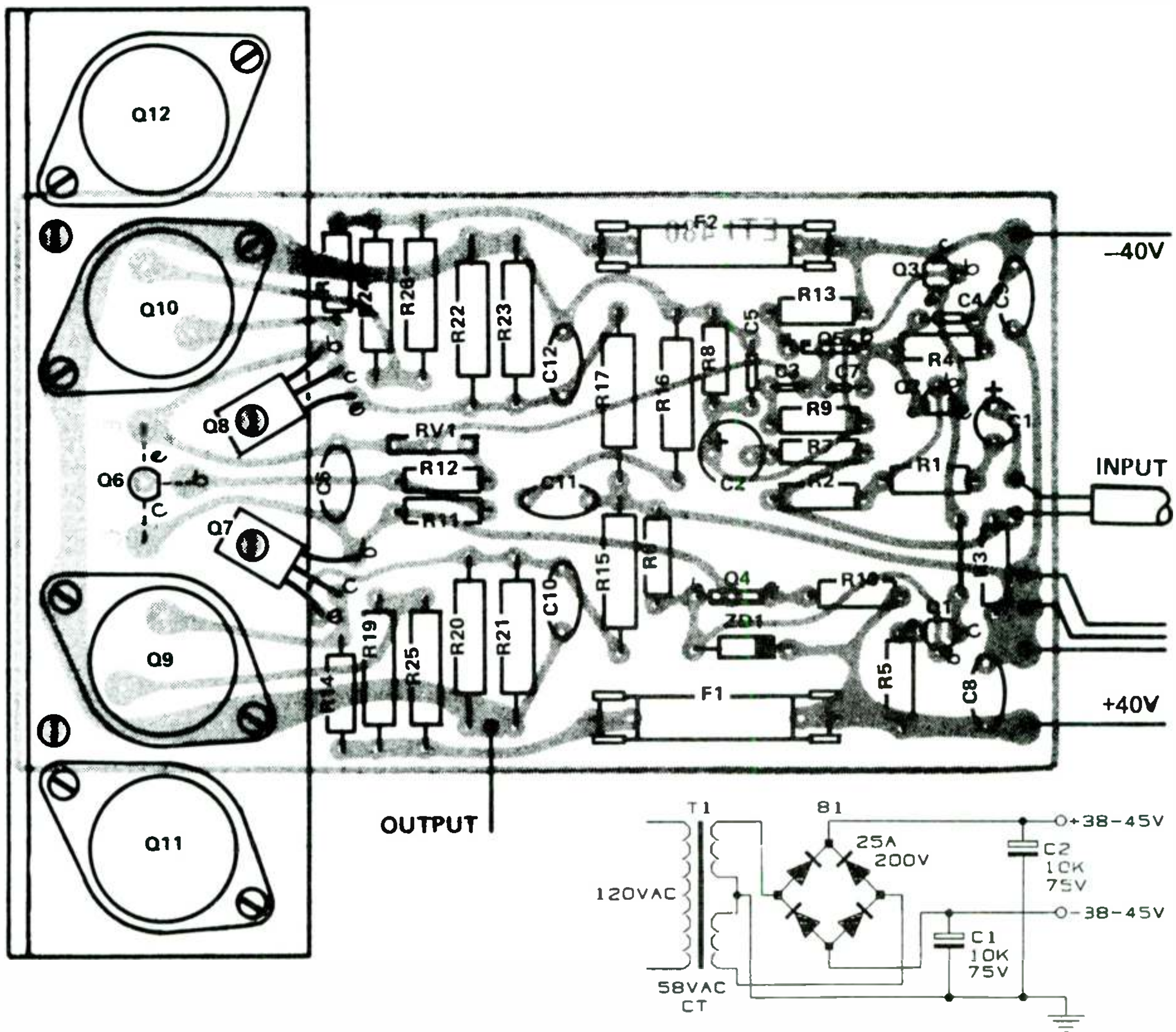


FIGURE 4: K3442 amp details and power supply recommendations (C1, C2 = 10,000µF @ 75V; B1 = 25A @ 200V; T1 = 120V AC Pri, 58V AC CT, 3A).

½″ round-over bit (must have a ball-bearing guide). Set the depth of cut so your round-over almost touches the surface of the Formica. Make several passes around the inside of both ports to achieve a blended-in look.

For a final touch, spray-paint the ports gloss black. Do not worry about getting paint on the covering; in fact, spray it heavily where the ports and Formica meet. When the paint is dry, remove the paint from the Formica around the ports with acetone, but leave the junction painted to give the illusion of the ports being molded into the front panel (*Photo 3*).

SEALING & WOOFER MOUNTING

Mounting the woofers was tricky because of the limited space in which to work. Normally, #10 × 1.25″ Robertson-head wood screws would work, but the cramped quarters required a screw with a hex head, which you can find at most hardware stores; most are of the self-tapping sheet-metal type. The hex head allows you to use a ratchet instead of a screwdriver. Be sure to use a strip of rope caulk between the drivers and the mounting boards; air-tight seals are very important.

Solder on your speaker leads, which should be at least 3′ to give plenty of slack. Drill very small holes for the wires in the back to make them easier to seal. Silicone caulk is a great sealant for these holes.

I used about 10 oz of fluffed-up Poly-fill in each sealed chamber. The exact amount or density is not critical in this design; use just enough to damp out the hollow sound of the sealed chambers. It is not necessary to use stuffing in the reflex chamber, or line the walls.

The back of the cabinet is recessed to allow for a thick layer of sealant. Windshield sealing caulk works the best in this application because it will let you back in if you need to change or fix something. Silicone is much more permanent, so if you need to go back in, expect to break more than just the seal. Screw down the back, apply another bead of windshield caulk around the back of the cabinet and work it into the seams to fill any voids. You can liberally spray-paint the back for a nice, finishing touch.

AMPLIFIER SELECTION

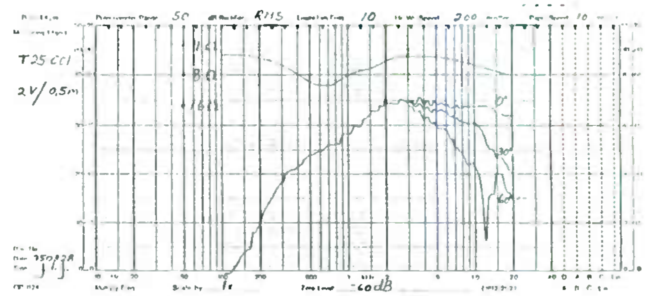
I have built many kit amplifiers, ranging from 14Wpc Class As to 270Wpc powerhouses from companies such as Dynaco, Hafler, Formula, Mark Five, and Dick Smith Electronics. Without hesitation, I prefer Dick Smith's DS K3442 100W module, which, in my opinion, is the best *cheap* bass amp on Earth. Krells, Rolands, and Aragons are better, but for around $45.95, plus a power supply and a heatsink, this little amp from down under is tops at getting down deep for real cheap. I purchased the power supply parts from a local electronic salvage shop.

Figure 4 shows the details of the DS

Madisound Presents..... seas *Excel* Speakers

Seas *Excel* — T25-001

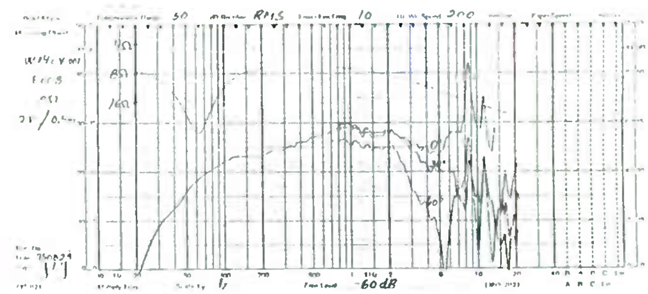
This 25mm dome tweeter features the proprietary SONOTEX fabric dome. SONOTEX is only manufactured by Seas and is precoated 4 times with a damping/sealing material, resulting in excellent acoustic performance and consistency. The T25-001 has a silver wire voice coil, tinsel leads, gold plated terminals and magnetic fluid cooling/lubricant. The T25-001 has a double magnet, one added in reverse for use in A/V systems. The complex shape chamber reduces resonance and interior reflections.



Nominal Impedance	8	Ohms	Voice coil resistance	4.3	Ohms
Recom. frequency range	2k-25k	Hz	Voice coil inductance	0.05	mH
Short term max. power	200	W	Force factor	3.5	N/A
Long term max. power	90	W	Free air resonance	750	Hz
Sensitivity (1W/1m)	90	dB	Moving mass	0.33	g
Voice Coil Diameter	26	mm	Suspension compliance	-	mm/N
Voice coil height	1.5	mm	Suspension mech. resistance	-	Ns/m
Air gap height	2.0	mm	Effective piston area	7	sq. cm
Linear coil travel (p-p)	0.5	mm	Vas	-	Liters
Max. coil travel (p-p)	-	mm	Qms	-	
Magnet weight	0.34	Kg	Qes	-	
Total weight	0.76	Kg	Qts	-	

Seas *Excel* — W14CY-001

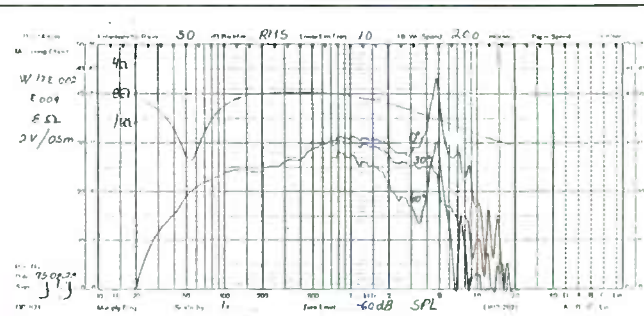
This 5" woofer features a light, yet stiff **magnesium cone** for bass precision and midrange detail. The cone is attached to the magnesium cast frame by a natural rubber surround that shows no sign of midrange (edge) resonance. Heavy copper coils mounted above and below the T-shaped pole piece reduce non linear and intermodulation distortion. A solid copper phase plug and copper plated top and bottom plates, enhance the performance of the copper rings and help conduct heat away from the voice coil. A large magnet system is used for improved sensitivity and transient response.



Nominal Impedance	8	Ohms	Voice coil resistance	5.4	Ohms
Recom. frequency range	50-3000	Hz	Voice coil inductance	0.37	mH
Short term max. power	200	W	Force factor	5.8	N/A
Long term max. power	70	W	Free air resonance	43	Hz
Sensitivity (1W/1m)	87	dB	Moving mass	9.6	g
Voice Coil Diameter	26	mm	Suspension compliance	1.4	mm/N
Voice coil height	14	mm	Suspension mech. resistance	1.5	Ns/m
Air gap height	6.0	mm	Effective piston area	75	sq. cm
Linear coil travel (p-p)	8.0	mm	Vas	10.6	Liters
Max. coil travel (p-p)	14	mm	Qms	1.80	
Magnet weight	0.42	Kg	Qes	0.44	
Total weight	1.14	Kg	Qts	0.36	

Seas *Excel* — W17E-002

This 6.5" woofer features a light, yet stiff **magnesium cone** for bass precision and midrange detail. The cone is attached to the magnesium cast frame by a natural rubber surround that shows no sign of midrange (edge) resonance. Heavy copper coils mounted above and below the T-shaped pole piece reduce non linear and intermodulation distortion. A solid copper phase plug and copper plated top and bottom plates, enhance the performance of the copper rings and help conduct heat away from the voice coil. A large magnet system is used for improved sensitivity and transient response.



Nominal Impedance	8	Ohms	Voice coil resistance	6.4	Ohms
Recom. frequency range	40-2500	Hz	Voice coil inductance	0.4	mH
Short term max. power	250	W	Force factor	7.8	N/A
Long term max. power	100	W	Free air resonance	34	Hz
Sensitivity (1W/1m)	87	dB	Moving mass	15.5	g
Voice Coil Diameter	39	mm	Suspension compliance	1.4	mm/N
Voice coil height	14	mm	Suspension mech. resistance	1.5	Ns/m
Air gap height	6.0	mm	Effective piston area	126	sq. cm
Linear coil travel (p-p)	8.0	mm	Vas	29.5	Liters
Max. coil travel (p-p)	19	mm	Qms	2.4	
Magnet weight	0.42	Kg	Qes	0.37	
Total weight	1.4	Kg	Qts	0.32	

Madisound Speaker Components, P.O. Box 44283, Madison, WI 53744-4283 U.S.A.; Tel: 608-831-3433 Fax: 608-831-3771

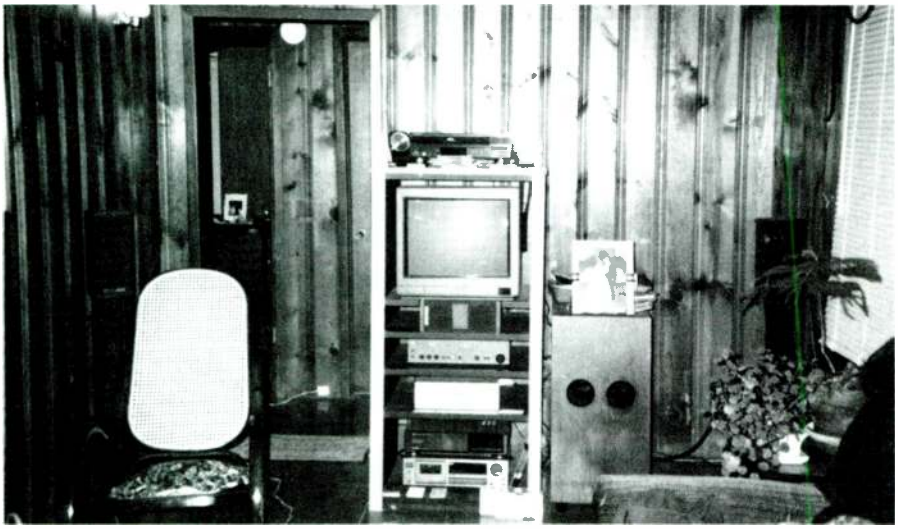


PHOTO 6: Unit installed in author's home.

K3442 (from the K3442 instruction manual) and a recommended power supply. Assembling the amp module is easy; Dick Smith provides everything you need: components, heatsink bracket, PC board, and even solder. Dick Smith sells the main heatsink separately, or you can rebuild a used model (check the sources).

I built the amp on an aluminum plate (1/8″). The back of the amp is taken up by the heatsink, connectors, volume control, and the line cord. *Photo 4* shows the completed amp ready to be slid into the sub cabinet. Once you slide the amp into place, its weight will hold it; no screws or fasteners are needed. The majority of the weight is positioned at the front of the amp's chassis.

Now you can hook up the woofers, remembering to place one of them out of phase with the other due to the push-pull configuration. The completed VS-1 (*Photo 5*) is ready to be tested and evaluated.

SYSTEM INTEGRATION

I connected the VS-1 to my audio/video system (see "System Components") with a second set of preamp outs I added to the NAD control preamp. *Photo 6* shows the installed sub already feeling right at home. After tinkering with the level control while playing a few movies and CDs, I noted a big initial improvement in the sound. Movies now had impact and music had depth.

The sub added a lot of bass and at first it was fun to crank it and shake the house. But the more I listened to music, I noticed two distinct problems the sub had introduced: lack of depth of image and overblown midbass. The sub was doing too much.

I turned down its level and the system regained its imaging somewhat, but I lost the impact and bass extension I wanted. Life is just not simple. The bandpass design was working, but it was still letting too much upper bass information through. And the satellite speakers were running full-range, thus adding to the midbass hump. Back to the drawing board!

SYSTEM COMPONENTS

Mitsubishi 20″ TV
Pioneer Laserdisc Player
Panasonic Hi-Fi VCR
JVC Cassette Deck
Dynaco FM Tuner
NAD Preamp
QUAD 303 Power amps (mono) (2)
Koss M80 Plus speakers (5)
Custom-made passive surround-sound decoder (à la Hafler)
VS-1 Sub

CROSSOVER CONSIDERATIONS

First I relieved the satellites of the low-bass signal information by changing the input capacitor in the satellites' power amps to a value which would roll them off at 100Hz. This improved the system's dynamics considerably. The system easily handled passages of music and movie scenes that used to strain the satellites and their amps. The satellites never sounded better.

But the sub still needed some help. *Figure 5* shows the active electronic crossover designed to tame the sub. It has a gentle 6dB slope at 100Hz and an added gain of four to regain some of the signal level lost by channel summing. The circuit was based on an earlier design, so I was able to use the earlier project's PC board (of which I have about 50 left, if anyone needs one!). The PC board contains room for a bipolar 12V regulated supply and the crossover circuit, which uses a single AD712 op amp.

I pulled the bass amp from the VS-1 and

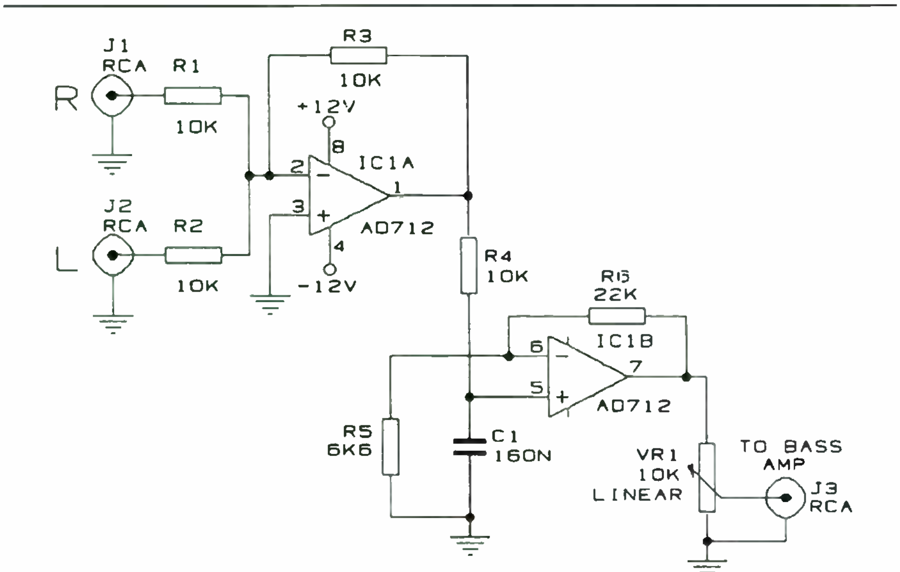


FIGURE 5: AV-1 electronic crossover.

SOURCES

Dick Smith Electronics
PO Box 321
North Ryde, NSW 2113
Australia
(Power amp module, heatsinks, power supplies)

Digi-Key
701 Brooks Ave. So.
Thief River Falls, MN 56701
(800) 344-4539
(Capacitors, bridge rectifiers)

Fair Radio
1016 Eureka St.
Lima, OH 45802
(419) 223-2196
(Transformers, filter caps)

Radio Shack
(800) THE-SHACK
(Capacitors, heatsink grease)

THE MATH BEHIND VS-1

Two Seas 25FEWX/8s (in push/pull configuration):

$F_s = 26Hz$
$Q_T = 0.23$
$V_{AS} = 350$ ltr
SPL = 97dB

1. Choose S = 0.7 (for superior transient performance)
2. $V_f = (2*S*Q_T) * (2*S*Q_T) * V_{AS} = 36.29$ ltr $= 1.3ft^3$
3. Choose Fl = 38Hz
4. $F_S/Q_T = 113.04Hz$, so $Fl/(F_S/Q_T) = 0.3404$
5. $Q'_T = 0.6010$ (from Margerand's Table 3)
6. $F_H = (F_H/(F_S/Q_T)) * (F_S/Q_T)$. Also from Margerand's Table 3, find the corresponding $(F_H/(F_S/Q_T))$ term.
   $F_H = 120Hz$
7. $V_B = V_{AS}/(((Q'_T * Q'_T)/(Q_T * Q_T)) - 1) = 60$ ltr $= 2.1ft^3$. This volume will be made up by the sum of the two sealed chambers' volumes. See *Fig. 1*.
8. $V_T = V_F + V_B = 3.4ft^3$
9. Calculate tuning frequency F_O. $F_O = Q'_T * (F_S/Q_T) = 68Hz$
10. Choose $D_V = 10cm = 4''$. Use two vents, thus $S_V = 81 * 2 = 162cm^2$.
11. $L_V = (30{,}000 * S_V)/(V_F * F_O * F_O) - 0.9 * \sqrt{(S_V)}$
    $L_V \doteq 17.5cm = 6.87''$. Used 7″ port length.
12. SPL' = SPL + PA. Find value of PA again from Table 3.
    SPL' = 97 + (−3) = 94dB

installed the electronic crossover, with just enough room to squeeze it onto the chassis. It looks like I planned for it to go there, but I did not. With the amp reinstalled and hooked up, the VS-1 was ready for further listening tests.

The additional 6dB of rolloff provided imaging, depth, and plenty of bass. The sub never sticks out or draws attention to itself; it is as if the little sats are shaking the house or calling to the whales.

SAFETY FIRST

In any project that involves building a power amp into a wooden cabinet, you should exercise great care to isolate all heat-producing elements from combustible items. Use common sense if you decide to go this route and plan where the hot spots and the flammable items will be. Poly-fill, Formica, and contact cement are very flammable.

In the VS-1, the amp is built on a metal chassis. This helps isolate the transformer, bridge rectifier, and heatsink from the wood. Also, the amp is in its own chamber and does not need to be around any Poly-fill used to damp an acoustic chamber. Thirdly, the back of the sub is painted, and separates the Formica and glue from the amp.

The Dick Smith amp module I used is a Class AB design and does not idle hot. In fact, at room-shaking levels, the heatsink, chassis, and transformer never become very hot, even after several hours of use. I do not recommend Class-A designs or SE amps for use in a built-in design because of the great amount of heat they must constantly dissipate.

YOUR CAR'S (AND LIVING ROOM'S) BASS BOOST

By Tom Nousaine

Ever notice how much easier it is to sing the bass parts when you're in the shower? Wonder why some cars seem to throb so loudly?

The answer is a major advantage of sound reproduction in an enclosed space and one of the reasons we should think twice about "eliminating the room" from the playback equation. In automobile sound systems this phenomenon is the transfer function of the cabin, which also occurs in living rooms. I recently had a chance to quantify the effect.

CAR PERFORMANCE

The transfer function seems to start between 60 and 80Hz for most automobiles, such as those I recently sampled: an '88 Acura Integra, '86 Ford Aerostar, '90 Honda CRX Si, '93 Camaro Z28, and '94 Corvette LT1. The overall pattern includes a 12dB/octave increase with decreasing frequency.

Figure 1 shows the close-mike lab response of a Bag End ELF 10″ subwoofer and the response of the woofer in the Corvette. *Figure 2* is the in-car response equalized to the lab response. While some activity exists between 80 and 160Hz, the basic shape is a 12dB/octave rise beginning at 70Hz. A system such as the ELF, which has a half-power point of 25Hz near field, can deliver response to 12Hz in the car. Actually, any good pair of 6 × 9 or larger woofers can deliver 30Hz or better.

According to one rationale, this phenomenon occurs because the car cabin is so small compared to the wavelengths where wave acoustics begin; the speaker thus energizes the interior as though it were a pressure pot. In other words, the speaker cone motion actually changes the size of the cabin. Another theory holds that the driver will be close to all the boundaries, even the opposite corner, so boundary reinforcement is the primary force operating here.

INTO YOUR HOME

This low-frequency transfer function applies to listening rooms, as well. I measured the

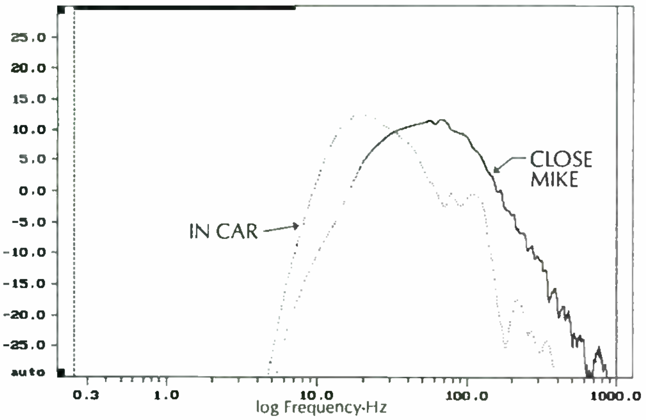


FIGURE 1: Close-mike 10″ woofer versus in-car (dot).

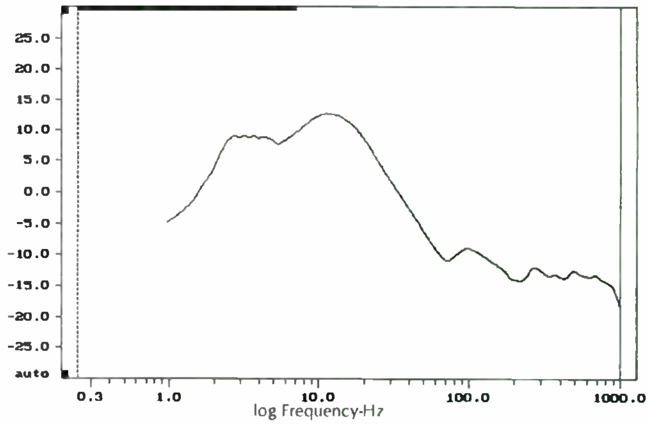


FIGURE 2: Bass transfer function of Corvette.

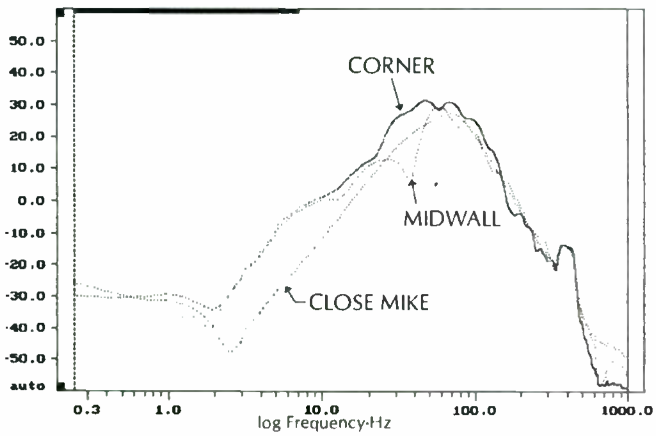


FIGURE 3: Corner location of ADS subwoofer near field (solid) produces smoothest, deepest response.

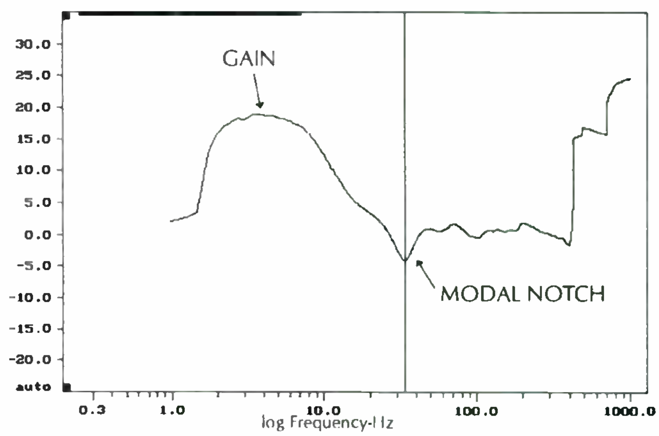


FIGURE 4: Room pressure gain starts at 34Hz (note response notch at 34Hz for non-corner location).

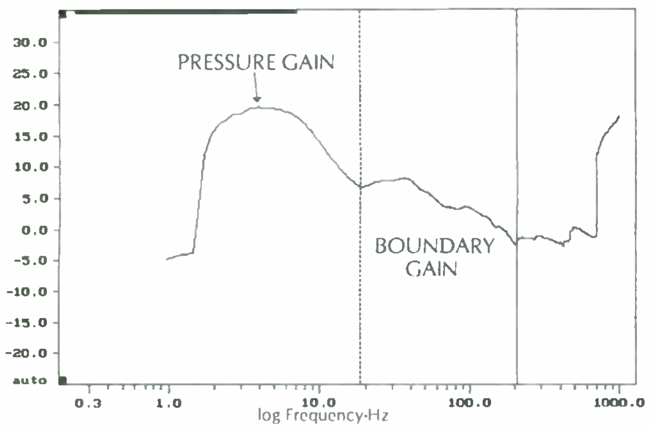


FIGURE 5: Corner placement shows room gain and boundary gain from 20-200Hz.

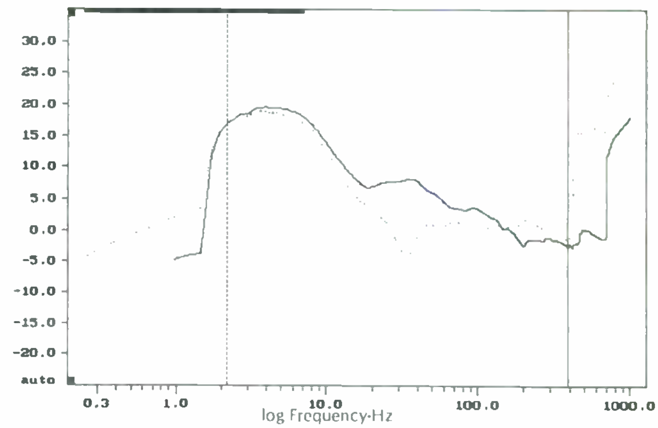


FIGURE 6: Corner location (solid) eliminates 35Hz response notch.

transfer function in my 22.25′ × 12.0′ × 8.0′ listening room in the same way. First I measured an ADS M3 10″ bandpass powered subwoofer with the microphone on the floor directly in front of the port that supplies all the system's output. I then equalized the response of the system—placed at a mid-wall location near an open doorway and in the corner of the room—to the near-field measurement (*Fig. 3*).

The ADS has a half-power point of 39Hz in the near field. Placed in the best corner in the room, the woofer is 3dB down at 30Hz, 6dB down at 25Hz, and 10dB down at 21Hz. But substantial output reinforcement occurs at very low frequencies. When we locate the woofer near a doorway and mid-way along the long wall, there are modal response notches around 30Hz, but significant reinforcement also occurs below 20Hz.

With the mid-wall response equalized to the near-field response, pressure reinforcement begins around 34Hz and output is reinforced at roughly 12dB/octave below this frequency (*Fig. 4*). Equalizing the optimal corner locus shows that roughly 12dB of additional boundary reinforcement is available from the walls between 20 and 200Hz.

This yields roughly 18dB of "room advantage" (not counting the 6dB from the floor included in the lab response) with corner placement to help subwoofer in-room response. The boundary reinforcement gained from corner placement seems doubly important because today there is so much program material with significant content to 25Hz.

Compact disc and home theater have revolutionized bass/low-frequency reproduction. Digital gave us low, low-frequency content down to 5Hz or so, and home theater provided movie sound effects which encouraged—no, ordered—the separate bass channel and separate subs, optimally located in the corner away from the left and right main speakers. We speaker builders won't settle for wimpy 25Hz 10″ subs. We want real bass...the 5Hz stuff.

Part 2

THE WAVEGUIDE PATH TO DEEP BASS

By G.R. Koonce

In Part 1 I described the initial waveguide construction and the tests I conducted with it. From the disappointing results of the listening tests, I concluded that I needed to stiffen the waveguide structure, but what else should be done? I had not clearly resolved whether to couple the drivers closely to the waveguide throats, as with my dual-driver boxes, or to establish some compliance between the waveguide throat and the driver, as I had in one side of my single-driver enclosures.

REVISIONS AND NEW ASSEMBLIES

I stiffened the waveguide structure by first driving ½″ dowels from one face to the other about every 8″. I then glued and nailed/screwed a second layer of 5/8″ particleboard to the original layer. The only boards I did not double in thickness on the basic structure were the one internal between the two sections of the waveguide, which is only 3″ high (and inaccessible), and the face where the attachments mount, as this is doubled when the attachments are installed. I stiffened the dual 8″ driver box internally and doubled the top with external stiffeners.

I decided to build a rubber-throat attachment for the waveguide. This is simply a section bolted between the dual-driver box and the waveguide throats that inserts a compliant volume of air to soften the coupling between the driver and the waveguide input, a technique sometimes used with horns. I made the compliant volumes about one-third the total net volume of the dual 8″ driver box with drivers installed (Table 1 in Part 1). I have no good reason for using this volume; it just made the rubber-throat assembly a reasonably low and stiff structure.

I also built a special structure that would allow the insertion of a single 6.5″ or 8″ driver into the waveguide with the tightest possible throat coupling I could achieve on the back side of the cone and still fit all my test drivers into the structure. This structure (*Fig. 17*) is called the close-coupled single-driver unit. Table 1 listed the gross internal volume of the smallest box I could build that would still accommodate all my test drivers.

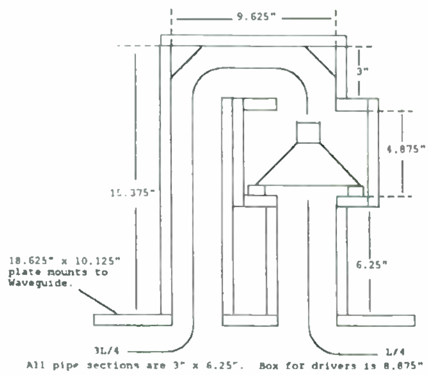


FIGURE 17: Drawing of close-coupled single-driver unit.

Photo 2 shows the close-coupled single-driver unit (right), the rubber throat (left front), and the stiffened dual 8″ driver box with cover (left rear). Photo 1 (Part 1) depicts the basic waveguide structure and ¾L pipe extension after the addition of the second layer of particleboard.

IMP TEST SESSION #2

I returned the modified waveguide structure and new attachments to the back room for more IMP testing. Would IMP show any difference in the waveguide structure performance now that it was stiffer? *Figure 18* shows the near-field (NF) time response from IMP with dual 8″ drivers X on the original waveguide structure; *Fig. 19* is the same data for the modified waveguide structure. Initially, I did not perceive any difference between these two curves, but after talking with Bill Waslo, IMP's developer, I saw that the first positive-going peak is the same height in both curves, but all following peaks in the original waveguide structure curve are reduced in amplitude relative to the modified waveguide structure. It appears that the original waveguide structure is dissipating or storing energy.

Figures 20 and *21* show the waterfall (cumulative spectral decay) plots for the time waveforms in *Figs. 18* and *19*, respectively. These waterfall plots are a bit different from what you normally see in speaker test reports; they show a 25dB vertical range but cover a 200ms time period to look for delayed effects. The modified waveguide structure is surely much cleaner after 50ms. Is this enough improvement? Only listening will tell.

Figure 22 shows the NF response for single 6.5″ driver D in the close-coupled single-driver unit. This now agrees with the expected response shape. *Figure 23* shows the waterfall plot for this configuration, which seems acceptable. *Figure 24* shows the NF response, and *Fig. 25* is the waterfall plot for single 8″ driver X in the close-coupled single-driver unit.

Figure 24 is the expected response shape,

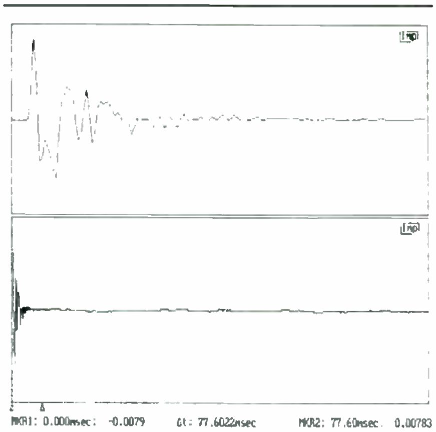


FIGURE 18: IMP near-field time response waveform, dual 8″ drivers X on original waveguide.

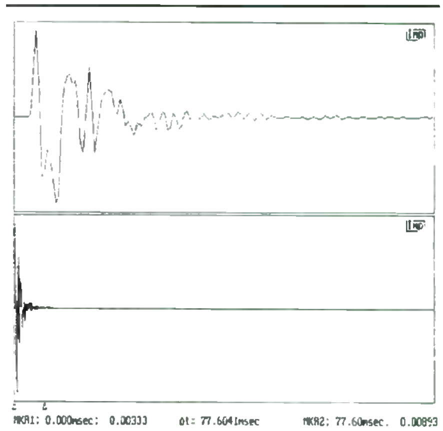


FIGURE 19: IMP near-field time response waveform, dual 8″ drivers X on modified waveguide.

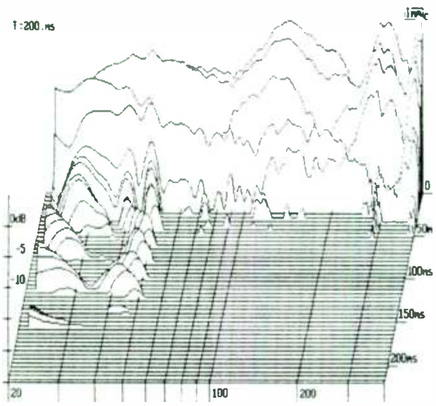


FIGURE 20: Waterfall plot for dual drivers X on original waveguide.

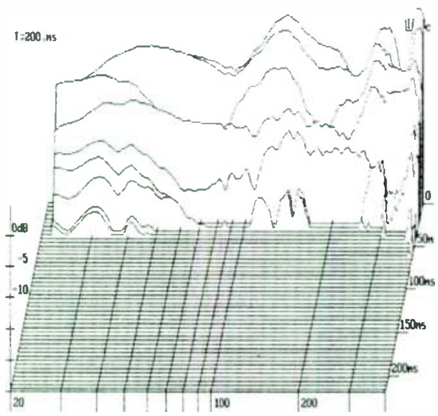


FIGURE 21: Waterfall plot for dual drivers X on modified waveguide.

and, as the cursor shows, the low-end −3dB point is about 35Hz. The waterfall response decay also seems acceptable. It appears that the closer coupling of both sides of a single driver to the waveguide throats works better and provides the expected response shape.

Figure 26 shows Z_{IN} for the dual 8″ drivers X on the modified waveguide structure. Comparison with the same result for the original structure, Fig. 11 (Part 1), shows little change. *Figure 27* shows the NF response for this configuration. If you compare it to Fig. 12 (Part 1), the same configuration with the original waveguide structure, you will note the passband shape is the same, but the passband SPL is higher relative to the trash above the passband.

As the cursor shows in *Fig. 27*, the lower −3dB point is about 34Hz. The waterfall plot for this configuration appeared previously in *Fig. 21*. The close coupling of the dual drivers, driven out of phase, to the waveguide throats appears to work well and produces the same −3dB point as the close-coupled single-driver unit, even though the total waveguide length is shorter. *Table 4* summarizes the waveguide lengths in various configurations.

ANOTHER LESSON LEARNED

I then introduced and tested the rubber-throat structure with dual 8″ drivers X. *Figure 28* shows Z_{IN} for comparison with *Fig. 26* without the rubber throat; peaks have moved lower in frequency, and some have increased in magnitude with the rubber throat. *Figure 29* shows the NF response on the basic modified waveguide structure with the rubber throat.

Three things are evident in this plot. First, the passband response is lower in SPL relative to the trash above the passband; this could indicate a loss of gain with the rubber throat of as much as 5dB. Second, the peak above 100Hz has been "flattened." Third, a bad dip occurs at 80Hz.

I had assumed that the waveguide length would start after the rubber throat, but clearly it does not. The rubber throat has inserted the same increase in length in both sides of the waveguide, when it should insert 3× the increase in ¾L that it does in ¼L. Rule #2: You must maintain the driver's location at the one-quarter point in the total length of the waveguide. *Figure 30* shows the waterfall plot for the above case, and it looks acceptable.

To partially correct the length ratio of the two waveguide sections with the rubber-throat assembly, I attached the unstiffened ¾L extension developed earlier, even though it was somewhat long. *Figure 31* shows the NF response with this configuration, which has removed the 80Hz dip. The lower −3dB point is around 40Hz, but the falloff rate below 40Hz is quite shallow, so the usable passband might be from about 30Hz to 150Hz.

I believe the passband ripples are due to the two pipe sections still not being in quite the required 3:1 ratio. *Figure 32* shows the waterfall plot for this configuration, and it does not look good. I initially dismissed this as an IMP processing anomaly; I would later learn otherwise. I adopted this configuration of rubber throat with full ¾L extension as the "standard" for much of the testing work to follow, but later determined a shorter extension was the better configuration. Remember to maintain the 3:1 length ratio.

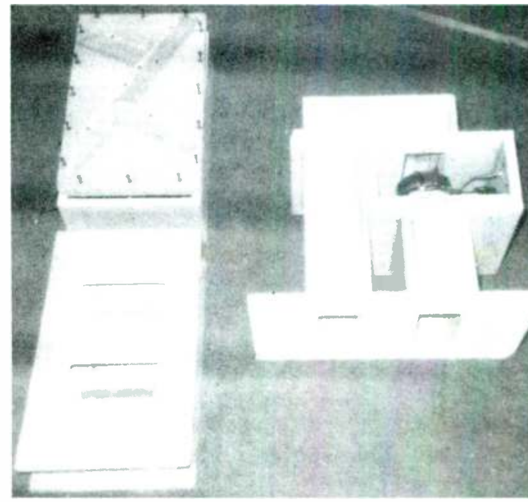


PHOTO 2: Close-coupled single-driver unit, rubber throat and braced dual 8″ box with top.

In summary, the modified waveguide structure seemed to offer response down to about 35Hz with the close-coupled single-driver unit, down to 34Hz with dual drivers coupled tightly to the waveguide, and down to maybe 30–35Hz with the rubber-throat attachment installed. None of these responses are truly flat; they generally rise with increasing frequency and have a peak above 100Hz. I believe the passband variations are slight compared to the effects of the listening room at these frequencies. It was time for more listening.

LISTENING SESSION #2

I moved the waveguide back to the garage and reinstalled it with the upper-end system.

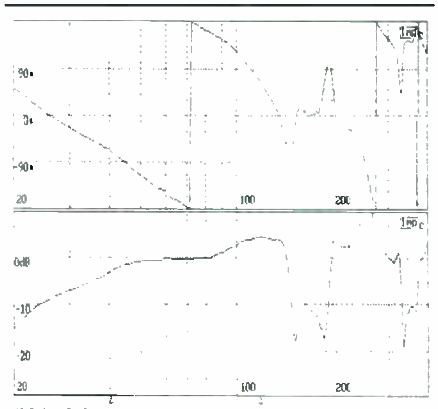


FIGURE 22: Near-field response for driver D in close-coupled single-driver unit.

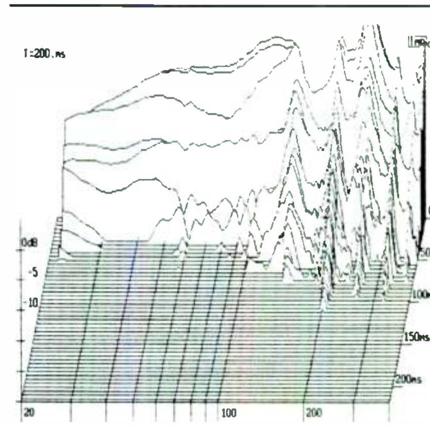


FIGURE 23: Waterfall plot for driver D in close-coupled single-driver unit.

I set the crossover frequency to 110Hz for all tests. This frequency would work with all waveguide configurations, helping to attenuate the trash above the passband, and keeping the motion of the 8″ driver in the upper-end system well-bounded at high playing levels. It was immediately clear the waveguide now worked.

Integration into the system was good, as I could now adjust the relative levels, always by using the same CD cut with kick drum, thus setting the balance for information probably up near the crossover frequency rather than by any deep bass content. Also, all the sound appeared to originate from only one source. Many listeners insisted that all the bass came from the 8″ upper-end system, and were only convinced when I shut off the waveguide.

Starting with 6.5″ driver D in the close-coupled single-driver unit, performance was good. If I ran the waveguide with normal polarity, there appeared to be no bass at all, so I needed to use it with inverted polarity relative to the upper-end system. This may be due to the use of a second-order crossover or my arbitrary definition of normal polarity for the waveguide.

The point is correct polarity was obvious and the incorrect polarity about useless. Among the CDs used in the evaluation, I found one with extreme bass content: Jennifer Warnes' *The Hunter* (BMG Distribution #01005-82089-2). A half-octave RTA monitoring the amplifier inputs showed that certain cuts were flat in frequency content down to the lowest half-octave at 31.5Hz, even though I was playing the waveguide at –6dB relative to the upper-end system, which has a sensitivity of about 88dB/2.83V/m. Average-reading (time constant of about 3s) wattmeters showed that on two cuts (#1 and #8) the total power below 110Hz exceeded the power above 110Hz, even with the 6dB padding below 110Hz. This CD can be destructive to small drivers on these two cuts, so be careful if you have it.

In general, performance of the total system with the waveguide was good. I saw as much as 10W average into the 6.5″ driver in the waveguide without any sign of trouble or apparent harm to the driver. Experience with a peak-reading wattmeter has shown the peak power from a CD can be more than 100× the average wattmeter reading.

Many times there is an integration problem when you have a system composed of direct radiator drivers doing part of the frequency range and nondirect radiators doing another portion. The effect is that the two portions stay matched in level only at one listening volume. None of this was evident between the waveguide and the upper-end system. Several listeners agreed that proper

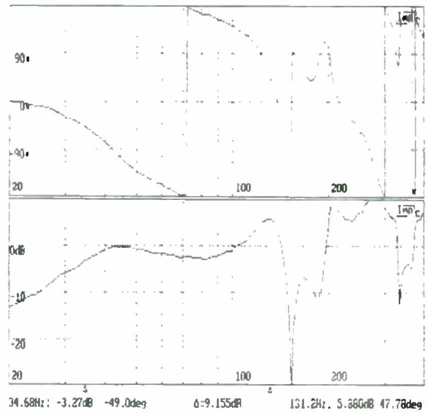


FIGURE 24: Near-field response for driver X in close-coupled single-driver unit.

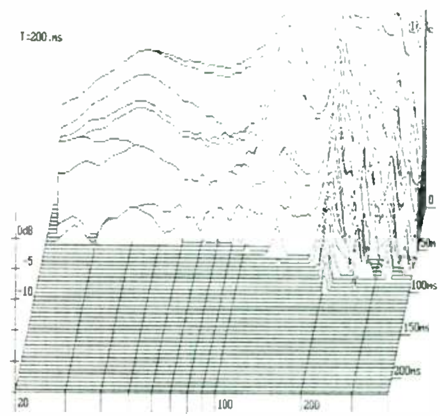


FIGURE 25: Waterfall plot for driver X in close-coupled single-driver unit.

balance was maintained over a wide range of listening level. Possibly the waveguide becomes stronger in relation to the upper-end system as the playing volume is reduced, giving a built-in loudness effect. Whatever the mechanism, the sound quality stayed balanced and fine over a wide range of listening levels.

DUAL DRIVERS

Next I tried the dual 8″ drivers X coupled tightly to the waveguide. Again I needed to use inverted polarity and judged –5dB to be correct for the waveguide level relative to the upper-end system. Both the single-driver waveguide system described above and this dual 8″ system are 4Ω, so the dual drivers seem to output about the same level as a single driver of the same ohms. The bass was tight and the overall sound good. It did sound as though the dual drivers produced more bass extension, but this is difficult to judge.

This configuration seemed able to play at louder levels than the single driver, even though I had suspected the pipe area was way too small for 8″ drivers. I made no attempt to see just how loud the system could play, as I was worried about putting over an average of 10W into the upper-end system! This would indicate the highest average playing level was about 101dB at 1m from only the waveguide. This level was more than adequate for my tastes.

I then inserted the rubber-throat assembly between the modified waveguide and the dual 8″ drivers X mounting box. Starting with the normal waveguide lengths (which should yield an 80Hz dip), I established that a waveguide level of –2dB was correct and the sound was good. This would indicate a loss of about 3dB by having the rubber throat, less than I had expected from my interpretation of the IMP NF data (later work indicates the loss is even less).

The problem here may be that I set the relative levels with music having a major content in the 80Hz dip region. Again, the inverted polarity was required, the normal polarity clearly useless. I did not really notice the anticipated 80Hz dip, and bass extension seemed about the same as without the rubber throat.

I next added the extension to the ¾L portion of the waveguide and was shocked to discover the waterfall plot was correct (*Fig. 32*), for the sound quality went to pot. Also,

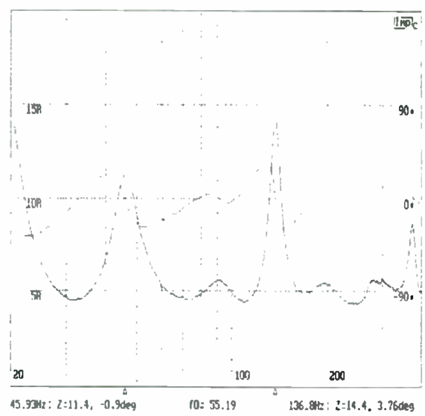


FIGURE 26: Z_{IN} to dual drivers X directly on modified basic waveguide.

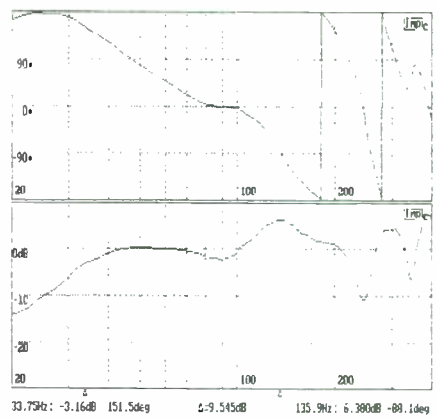


FIGURE 27: Near-field response for dual drivers X on modified waveguide.

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MUSIC POWER	300 W
FREQUENCY RANGE	19-1000 Hz
SENSITIVITY (2.83 V)	90 dB
EFFECTIVE CONE AREA	507 cm2
VOICE COIL RESISTANCE	3.4 ohm
VOICE COIL INDUCTANCE	1.2 mH
VOICE COIL DIAMETER	50 mm
VOICE COIL HEIGHT	34 mm
AIR GAP HEIGHT	8 mm
XMAX	13 mm
FREE AIR RESONANCE	19 Hz
MOVING MASS (MMS)	121 g
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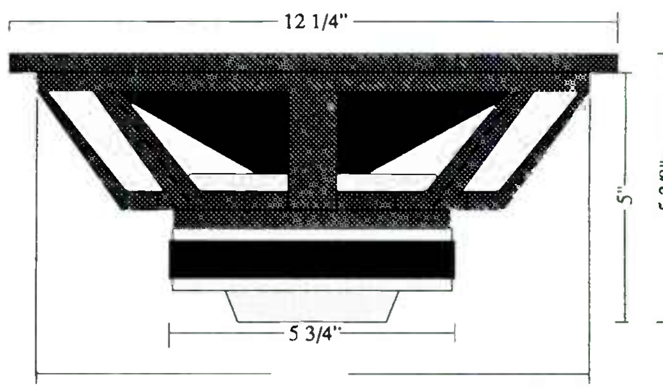

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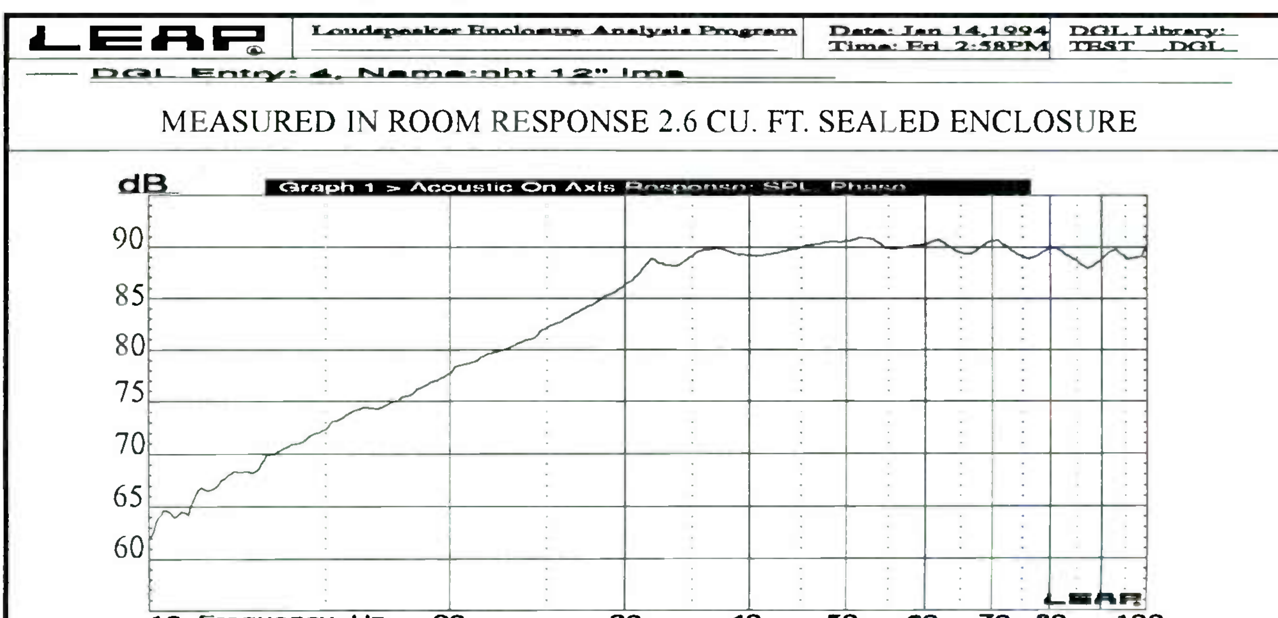

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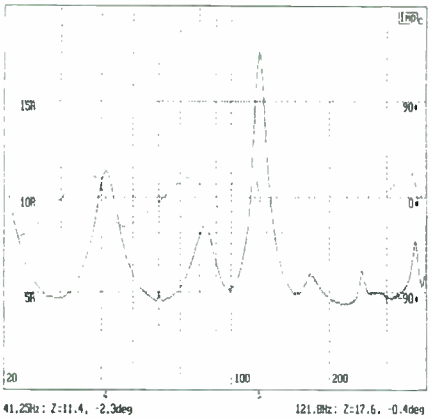

FIGURE 28: Z_{IN} to dual drivers X with rubber throat on modified basic waveguide.

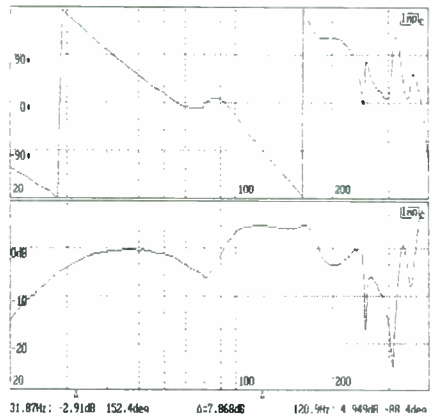

FIGURE 29: Near-field response for dual drivers X with rubber throat, no ¾L extension.

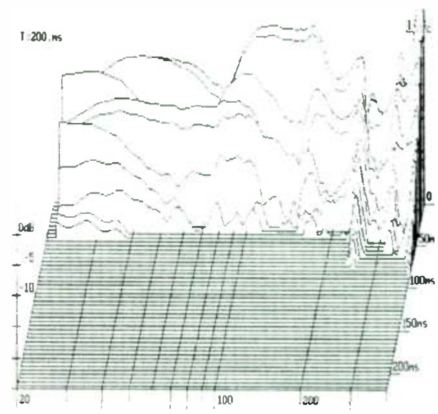

FIGURE 30: Waterfall plot for dual drivers X with rubber throat, no ¾L extension.

although inverted polarity was still best, normal polarity did increase bass; this had not happened with any other configuration when the relative levels were set correctly. The ¾L extension had a 90° turn at the end, and I thought this might be the problem. I removed the extension and sawed it off so it was 9″ in length, which should be correct for inserting the rubber-throat assembly. With this 9″ extension, now with port facing straight out, the sound was still poor.

When I decided to stiffen the waveguide structure, I had not anticipated ever using the extension again, so it was still a single thickness of 5/8″ particleboard and was shaking badly. Could a 9″ section of unstiffened pipe cause trouble? To answer this, I returned to construction mode, stiffened both pieces of the extension, and provided for the installation of either the short 9″ section or the total 24″ length (Photo 1, Part 1).

TABLE 4

SUMMARY OF PIPE LENGTHS (INCHES)

STRUCTURE	¾L	¼L	¾L : ¼L	TOTAL L
Basic waveguide with dual drivers	94.7	32.1	2.95	126.8
Dual drivers, rubber throat (no extension on ¾L)	99.0	36.4	2.72	135.3
Dual drivers, rubber throat (9″ extension on ¾L)	108.0	36.4	2.97	144.3
Dual drivers, rubber throat (full extension on ¾L)	123.0	36.4	3.38	159.3
Close-coupled single-driver unit on basic waveguide	120.1	39.6	3.03	159.7

TABLE 5

NEAR-FIELD RESULTS WITH PINK NOISE

WAVEGUIDE CONFIGURATION	dB DOWN 40Hz	31.5Hz	25Hz
Single 6.5″ driver in the close-coupled unit	2	10	18
Dual 8″ drivers close-coupled	3	11	19
Dual 8″ with rubber throat (no extension)	2	6	20
Dual 8″ with rubber throat (9″ extension)	2	6	18
Dual 8″ with rubber throat (full extension)	2	6	12

I then reevaluated the rubber throat, starting with the full-length extension. I was surprised to find, when I first listened, that I could barely detect when the waveguide section was on or off. Even though lots of air was coming from both ports, it appeared to add nothing to the output of the upper-end system except at very low frequencies.

Checking, I found that I had mistakenly connected the waveguide in normal polarity; it was now behaving as the other configurations had. Connecting the waveguide to the correct inverted polarity gave the system great bass extension, and I needed to reduce the waveguide padding to −4dB. This indicated the rubber-throat loss was less than I originally thought—perhaps only a decibel or two in the portion of the passband I was using.

This configuration, seeming to produce the best bass extension, was now tight and working properly. It is clear that if only a portion of the waveguide structure is not stiff enough, the sound quality will suffer. Remember Rule #1: Keep it stiff! I also tried the system with the short 9″ extension, and the sound was good, but I concluded the full-length extension was better for deep bass extension. At this time I had no IMP data on the configuration using the 9″ extension. The rubber-throat approach definitely has merit and should be considered.

Near the end of testing with the rubber-throat configuration I sensed the bass was not as tight as it was without the rubber throat,

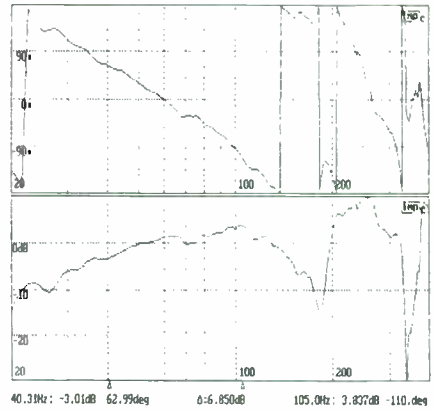

FIGURE 31: Near-field response for dual drivers X with rubber throat, full ¾L extension.

or as it had been earlier with the rubber throat. I started to return to the tightly coupled dual 8″ drivers, but when I opened the dual driver box to start this conversion, I found that two of the machine screws sealing the box to the rubber throat between the drivers had become loose. I retightened these and the rubber-throat configuration returned to tight bass.

TESTING OLD WAYS

I am not experienced in listening for deep bass extension. I shut off the upper-end system and listened to just the waveguide. It was much like listening only to a tweeter, which usually gets less than about 10% of the system's total power, so you don't expect much. The waveguide was getting 50% or more of the power, and by itself still sounded like nothing. However, just as with a tweeter, shutting down the waveguide when the full system was playing made it clearly evident that much information was lost.

If you experiment with any of these waveguide structures, listen to them integrated into a system and not alone, or you will give up immediately. I listened to the dual 8″ drivers tightly coupled to the waveguide with

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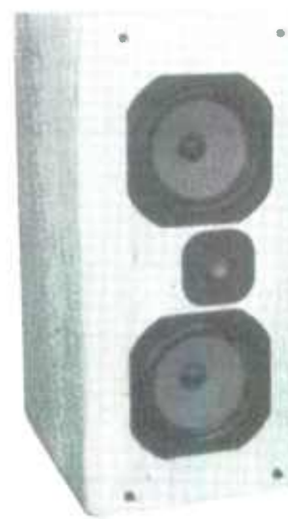

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Qts	0.298
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Re (Ω)	6.5
Power (W)	50
Lvc (mH)	0.5
Xmax (mm)	3.25
Sd (sq. cm)	87
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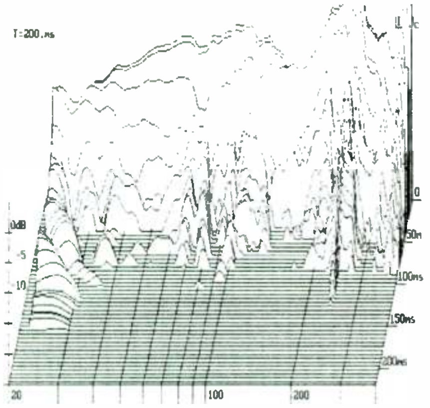

FIGURE 32: Waterfall plot for dual drivers X with rubber throat, full ¾L extension.

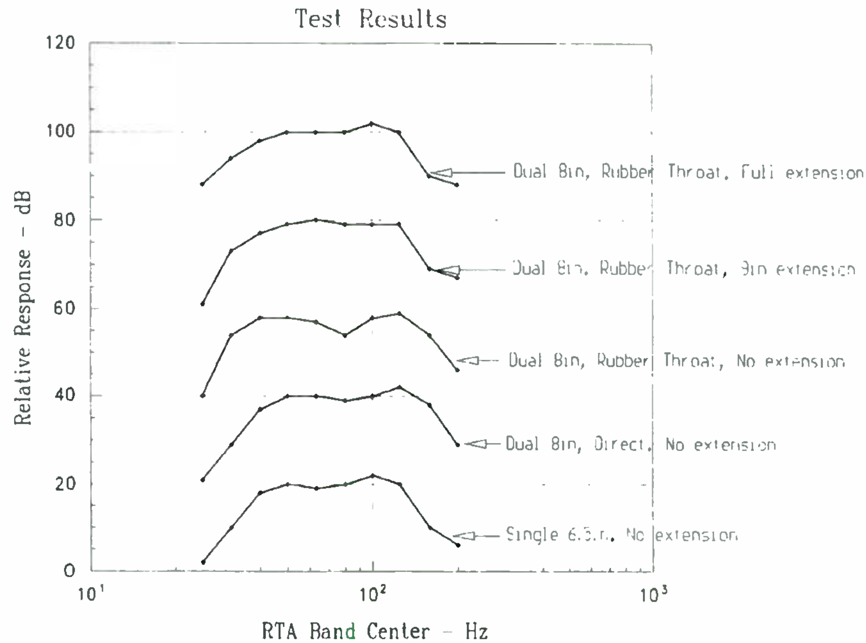

FIGURE 33: Summary of near-field pink noise test results.

1/3-octave warble tones, with my generator going only to 30Hz on the bottom end. At a 2W input level, the apparent loudness started to fade around 50Hz. This is what I had experienced on direct radiator systems that were calculated (anechoic) to go down to the 30Hz range. It may have been room effects or the nonlinearity of the ear with regard to low frequencies, but the system did not sound flat down into the 30Hz range.

Were the various waveguide configurations really doing what the IMP NF tests indicated? The dual tightly coupled drivers seemed to go slightly deeper than the single drivers, and the rubber-throat dual drivers were possibly the lowest of all. I decided to try the old ways, measuring the NF response in the garage with pink noise and the 1/3-octave RTA.

Just as I was setting the noise drive level into only the waveguide at 2W, a jet airliner flew over the house. Something was wrong,

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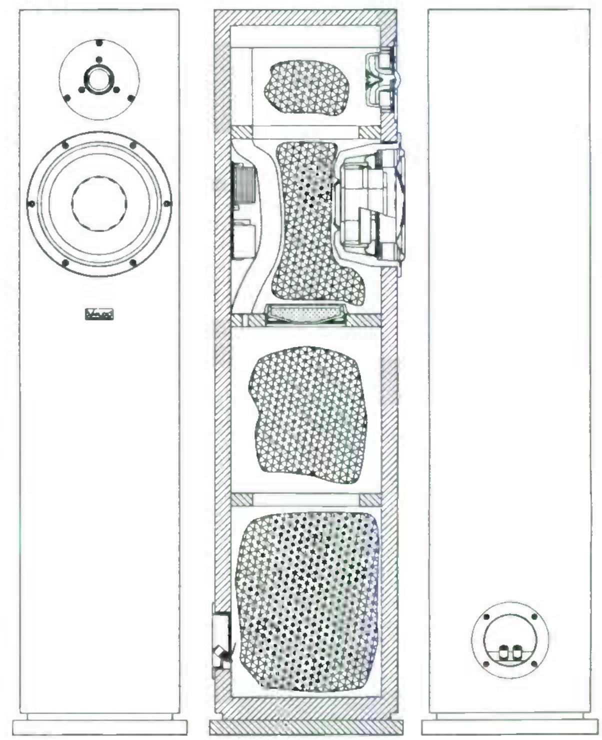

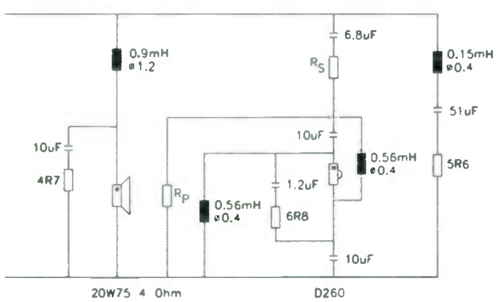

+ Tweeter :	Rs	0.56 Ω	
	Rp	15 Ω	
0 Tweeter :	Rs	1.2 Ω	Default
	Rp	15 Ω	
– Tweeter :	Rs	2.2 Ω	
	Rp	15 Ω	

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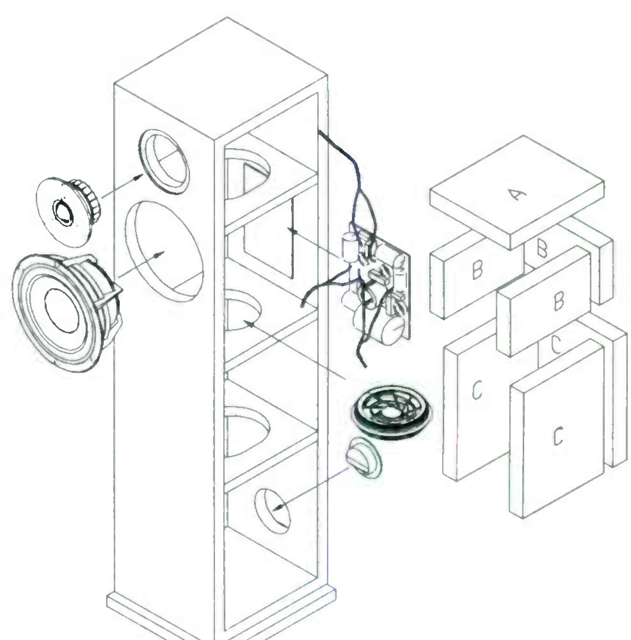

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Madisound Speaker Components; P.O. Box 44283, Madison, WI 53744-4283 USA Tel: 608-831-3433 Fax: 608-831-3771

as I did not hear anything from the waveguide, but that airliner was getting very low. It then dawned on me that the pink noise low-passed to 110Hz into the waveguide sounds just like an airliner. A half-octave RTA monitored the amplifier input and verified the pink noise was flat down to its lowest display band (center at 31.5Hz).

NF RESULTS

All of the standard configurations were tested and are plotted in *Fig. 33*. I offset the curves for the different systems so their shapes can be compared. I ran several curves for each configuration, and the curve plotted is what appeared to be typical. Exact low-frequency measurement with pink noise is not possible with the averaging time used in my RTA.

The bass extension looks about the same for the single and dual close-coupled drivers. The rubber throat does appear to add extension to the bass. *Table 5* summarizes the results of these tests, but note that the frequencies are band centers for 1/3-octave bands, and the passband 0dB level is difficult to establish, as the measurements were made with noise. I do not know if the calibrated mike used with the RTA is really good down to 25Hz.

Figure 33 clearly shows the 80Hz dip for the rubber-throat configuration with no extension, this dip being present for all such cases. I tested this setup because it can be compared directly with the case of the close-coupled dual-driver to get some idea of loss due to the rubber throat. I kept the mike position and all gains constant between the two tests. The results indicate that the loss is in the 0–1dB range. Listening tests indicated a 3dB difference between these two cases, but that may be because the levels were set with drum that might have major energy content in the region of the 80Hz dip.

Taking all the data into account, it appears that the rubber throat does not cause a major loss of efficiency in my passband, but raises the level of the trash above the passband, making for tougher crossover requirements. Note in *Fig. 33* that the cases with the rubber throat and pipe extensions show more 200Hz content than the close-coupled dual drivers. The single close-coupled driver shows this same effect, and IMP NF results (*Fig. 22*) confirm this.

If you plan to use a low-rate crossover (I would not go below second-order), the dual close-coupled drivers seem to have an advantage. Remember that these have the shortest waveguide length, which may be the real cause of this effect. Note in *Fig. 33* that the 200Hz rise does not appear with the rubber throat until the extensions are added. More work is needed to identify the cause.

NEXT TIME

We have covered the waveguide modifications and construction of additional fixtures which looked better in testing. A second listening session showed the waveguide now worked properly and integrated well with a direct radiator system used to cover the upper end of the audio band. I made some additional modifications during the listening session. Next time I will cover testing these modifications and follow-up work with the waveguide structure.

PREVIEW
Audio Amateur

Issue 3, 1995

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- Get On the Bus, Part 1
- Regulators for High-Performance Audio, Part 3
- Marchand's PM2 Amps


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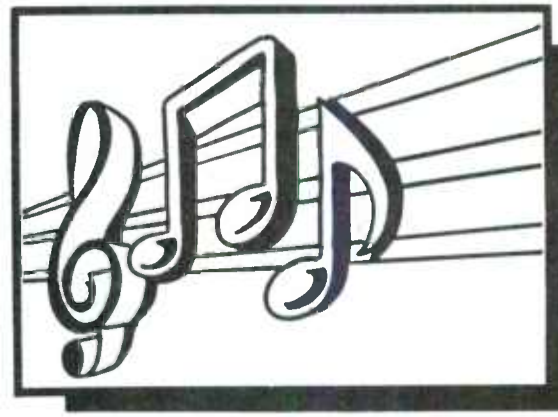

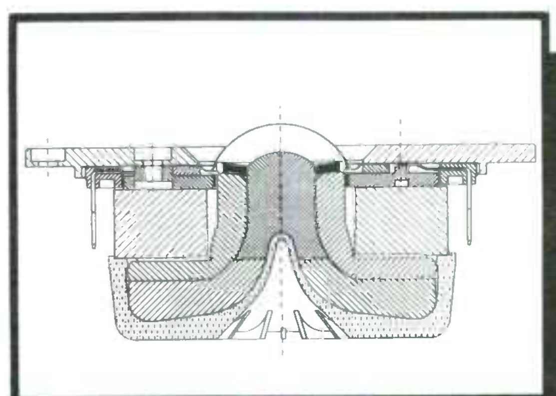

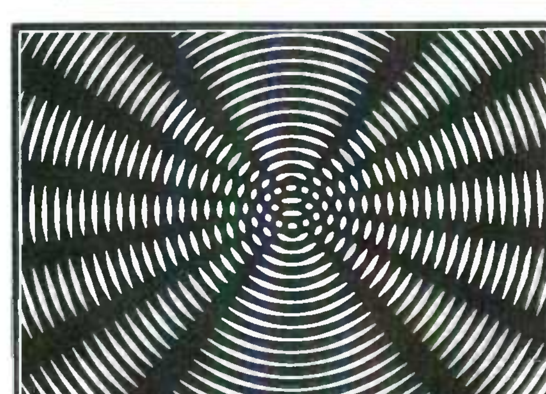

DRIVER TEMP AND T/S

By Mark Rumreich

Have you ever noticed that on a cold winter morning your car stereo seems to lack bass? To understand why, I decided to measure the T/S parameters of a woofer at realistic automotive temperature extremes.

PROCEDURE

Rather than wait for seasonal temperature extremes to occur, I chose to use the controlled environment of a thermal test chamber. The Tenney JR environmental chamber provided a more than adequate temperature range. It has an internal volume of 1.25ft^3, which was satisfactory for the driver I measured. The loudspeaker I used for this experiment is the Foster C130RX255R05, a 4Ω, 5-inch-diameter woofer with a polypropylene cone and foam surround.

For each measurement, the temperature was maintained for 30 minutes to allow the speaker to stabilize. I used an automated measurement procedure, and cone mass loading technique to measure V_{AS} because of the restricted test chamber volume. I repeated the measurement for each temperature to produce three consistent trials, then took the average. The compressor and fan of the test chamber were off during all measurements to prevent electrical or acoustic noise from affecting the results.

RESULTS

From the measurement results (*Table 1*), it's apparent that temperature greatly affects the T/S parameters, but it's difficult to determine what this would do to the sound in your car.

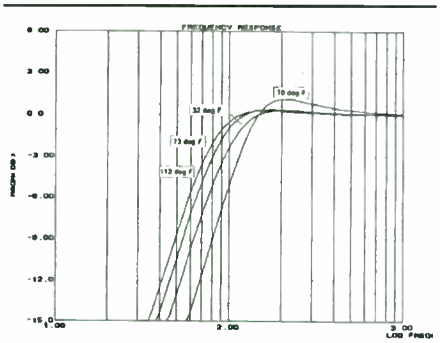

FIGURE 1: 0.5ft^3 sealed box.

TABLE 2

MECHANICAL PROPERTIES VS TEMPERATURE

TEMPERATURE	NORMALIZED COMPLIANCE (V_{AS})	NORMALIZED MASS ($V_{AS}^{-1}f_S^{-2}$)	NORMALIZED MOTOR STRENGTH (f_S/Q_{TS})
10°F	0.21	1.47	1.30
32°F	0.61	1.06	1.18
73°F	1.00	1.00	1.00
112°F	1.31	1.04	0.92

Figures 1–3 show computer-generated frequency response curves for possible woofer applications.

Figure 1 measures the woofer in a 0.5ft^3 sealed box. At the "reference temperature" (73°), f_3 is 80Hz. At 112° this number improves to 73Hz, and at 32° it degrades to 95Hz. As the temperature is reduced to 10°, f_3 worsens to 115Hz. These numbers are consistent with the observation that deep bass seems to disappear as the temperature drops. Notice that the 10° curve shows somewhat of a departure from the trend of the other three curves. This was repeatable, but inexplicable.

Figure 2 shows the woofer in a 2ft^3 ported enclosure with f_B = 41Hz. The general trend is the same as for the sealed box, namely, reduced deep bass as the temperature drops. Unlike the sealed box, the shape of the frequency response curves is quite temperature-dependent, with a pronounced dip slightly above resonance which deepens with reduced temperature. This is not surprising, since ported alignments have higher sensitivities to driver variations than sealed ones do.

In the bandpass system of *Fig. 3*, the box volume is 2ft^3 with the ported subenclosure volume comprising 11% of the total. Here the effects of temperature are truly dramatic. Not only do the response bandwidth and center frequency change with temperature, the efficiency does as well. Like the sealed and ported systems, the lower cutoff frequency is degraded with cold.

TABLE 1

T/S PARAMETERS VS TEMPERATURE

TEMPERATURE	F_S (Hz)	Q_{TS}	V_{AS} (ft^3)
10°F	136.8	1.03	0.052
32°F	94.7	0.79	0.150
73°F	76.4	0.75	0.245
112°F	65.3	0.70	0.322

TEMPERATURE DEPENDENCY

After measuring the changes in T/S parameters with temperature and simulating their behavior in enclosures, I next wished to understand the reasons for the temperature sensitivity. *Table 2* lists the fundamental mechanical properties of the driver, calculated from the T/S parameters. Ignoring the 10° case, the normalized moving mass remained relatively constant with temperature. This makes intuitive sense, since you would not expect mass to be affected by temperature.

The compliance, on the other hand, changed dramatically with temperature. This also makes sense because the stiffness of the polypropylene cone is very temperature sensitive. The motor strength is intermediate in

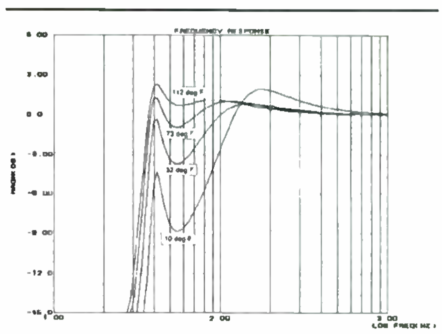

FIGURE 2: 2ft^3 ported box, f_B = 41Hz.

TABLE 3

MEASURED VS PREDICTED MOTOR STRENGTH

TEMPERATURE	NORMALIZED MOTOR STRENGTH (f_S/Q_{TS})	NORMALIZED BARIUM FERRITE B_R	NORMALIZED COPPER CONDUCTIVITY	PREDICTED NORMALIZED MOTOR STRENGTH
10°F	1.30	1.07	1.16	1.24
32°F	1.18	1.04	1.10	1.14
73°F	1.00	1.00	1.00	1.00
112°F	0.92	0.96	0.93	0.89

its temperature sensitivity. Two factors affecting motor strength are the temperature coefficient of the magnet and the temperature dependency of the voice coil impedance.

Table 3 lists the theoretical contributions of each of these two factors. The temperature coefficient for barium ferrite (ceramic) magnets is −0.19%/°C (IEEE Transactions on Magnetics, June 1968, Vol. MAG-4, p. 93.) This means the magnetic field becomes stronger when it is cold. The temperature coefficient for copper wire is approximately −0.4%/°C, so more current flows (for the same applied voltage) when it is cold.

Both these factors contribute to make the motor structure more powerful at low temperatures, with the voice coil conductivity dominating the situation by a factor of roughly 2:1. The combinations are shown in the predicted motor strength column. The results generally agree with the T/S parameter-based numbers.

CONCLUSION

All you, as a consumer, can do about temperature-dependent T/S parameters is boost your bass control until the car warms up. (At least now you can do so with a feeling of authority.)

A woofer manufacturer should be able to reduce the temperature sensitivity of a driver by carefully selecting materials. Numerous magnet materials are available, many with low temperature coefficients. Alnico, for example, has a temperature coefficient of −0.02%/°C.

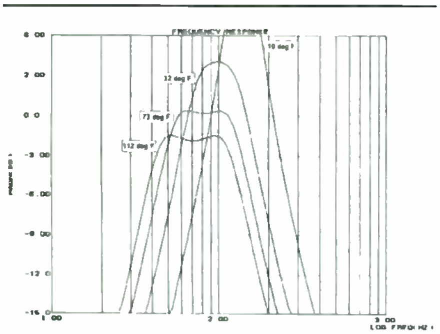

FIGURE 3: 2ft^3 bandpass, ported subsection = 11%.

Unfortunately, conductors suitable for voice coils all have similar temperature coefficients to copper (and most have lower conductivity), so there may not be much room for improvement. There are many cone materials from which to choose, but factors other than temperature sensitivity may dominate the selection decision.

This experiment raises as many questions as it answers. What caused the somewhat anomalous behavior at 10°? Would paper cones show more or less variation with temperature than polypropylene? What about humidity? I'll leave it to other *SB* readers to tackle these questions.

Wayland's Wood World

LAYING OUT AND CUTTING PANELS

By Bob Wayland

After designing your enclosure and choosing the material, you are ready to start construction. The hands-on fun has finally begun! The enclosure material, usually purchased as panels of medium-density fiberboard (MDF) or plywood, leaves you with a feeling of awkwardness: this monster must be cut down to size. The techniques I have found most useful are the subject of this column.

GENERAL CONSIDERATIONS

Before we begin, let's revisit our nemesis, splintering when cutting veneered wood. With a hand circular saw, the teeth of the blade come up from the middle of the plywood and cut towards the upper (top) surface. If your blade is not very sharp and perfectly flat, or if there is the slightest misalignment of the saw blade and/or the piece being cut, the veneer will splinter. The teeth catch on the veneer and pull it up, away from the crossbanding. The tearing occurs on the top, where the teeth come up through the plywood.

Always cut veneered boards so the veneer will be held in place. Use a sharp, flat blade, with the teeth extending through the wood enough that the gullet between them is at the surface.

Place the piece you are cutting with a power hand circular saw so that the good (exposed) side is down, resting on a sheet of scrap plywood or particleboard. You will then have a cleanly cut edge to work with. Or, you can score with a sharp marking knife (i.e., an X-acto) just where you want the cut to be, then carefully align so that you cut down the center of the score. (This assumes that your saw is accurately aligned.)

For more insurance, run masking tape along the cut and remove—again carefully—what is left after you are finished. When you make the cut, slow down, but not enough to cause burning. A good way to judge cutting speed is by listening to the sound of the saw. If the turning speed starts to reduce, you are going too fast.

LAYOUT

When you establish your cutting plan, you naturally want to have as little waste as possible. Of equal importance is the need to minimize the cutting difficulties. The fly in the ointment is which method to use.

There are problems with making *straight* cuts over an 8′ length, as with a 4′ × 8′ panel. We will almost always be dealing with pairs of same-size pieces, which gives us an advantage. Normally, when laying out the pieces, you place the long dimension along the long dimension of the panel. Instead, put the long dimension of the enclosure pieces parallel to the panel's short dimension. This enables you to use more accurate cutting techniques.

Group together like-dimensioned pieces, so that a single cut allows you to create smaller panels which can then be sawed to the dimensions of the enclosure pieces on a table saw. If you don't have a table saw, you can use the following techniques to cut accurately sized panels.

I am assuming that you have at least a power hand circular saw. If you also have a table saw, you are ready to take on the most ambitious projects. There are basically two methods: rough cut to dimension with a circular saw and make the final sizing cuts on your table saw, or carefully cut to exact size with your hand circular saw.

You have more control and a higher chance of success with the first method. Always allow a bit of oversizing if you plan to use a table saw to cut the pieces to size: about a half-inch added to the dimensions for the rough-cut pieces is usually about right. As always, don't forget the kerf of the saw cut when setting up the cutting patterns. The second method saves time, and with care produces noteworthy results. It all depends upon what tools you have and your approach to speaker building.

to page 34

PHOTO 1: Veneered-stock saw blades. I have included a blade that is warped as a result of misuse. It may be possible to have such a blade professionally straightened, but most of the time it should just be discarded. The top blade is carbide-tipped, with a very small set on the teeth.

PHOTO 2: Setting the cutting depth on a circular saw. Notice how the teeth of the Sears blade just clear the surface of the stock to be cut.

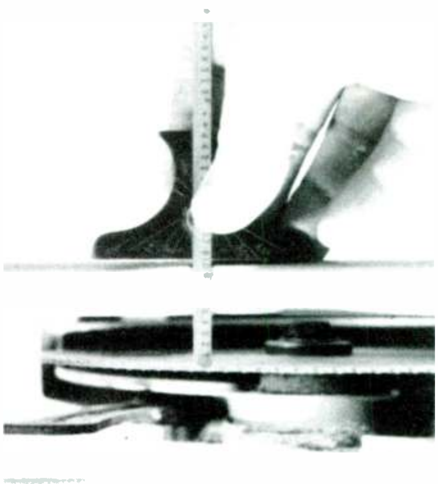

PHOTO 3: Measuring the offset from the edge of the base of the saw blade.

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Value µF	Diameter mm	Length mm	SRP US$	Value µF	Diameter mm	Length mm	SRP US$
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1.5	12	22	1.44	15.0	25	38	4.18
1.8	13	22	1.49	20.0	29	38	5.16
2.2	15	22	1.58	24.0	29	43	5.98
2.7	14	25	1.67	30.0	32	43	7.30
3.0	15	25	1.73	33.0	32	48	7.74
3.3	16	25	1.78	41.0	35	48	9.32
3.9	16	25	1.83	50.0	37	53	10.96
4.7	18	27	1.96	51.0	37	53	11.16
5.6	18	30	2.10	56.0	39	53	12.00
6.0	19	30	2.20	62.0	39	53	12.98
6.8	20	30	2.33	75.0	43	58	15.12
8.0	20	33	2.91	82.0	45	58	16.28
8.2	21	33	2.97	91.0	47	58	17.50
9.1	22	33	3.08	100.0	49	58	18.76
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CUTTING TOOLS

Let's quickly review the characteristics of power cutting tools (*SB* 2/95, p. 54).

Circular Saws: Buy this equipment with the greatest of care. On which side of the motor is the blade located? Is the handle on the top or the back? Is the gear drive straight or worm? The weight is also important: too light is hard to control; too heavy will wear you out. I like a saw that weighs 8–10 lbs, but this is something you must decide for yourself.

Unless you plan to cut stock thicker than 2″, a 6½″ or 7¼″ saw should fill your needs. The Sears plywood and veneer steel blade, if kept very sharp and properly tensioned, is a good choice for producing a satin-smooth cut with minimum splintering (*Photo 1*). Circular saw blades are easily bent or distorted by improper cutting techniques (e.g., forcing the cut, turning the blade in the kerf). The result is a hard-to-control cut with much splintering. Keep your blades sharp and clean, and your cutting unforced.

Table Saws: The advantage table saws have is increased accuracy, as well as the ability to make complex cuts with greater control than circular saws. Along with the disadvantages of size and cost, more woodworking accidents occur on table saws than any other piece of equipment. Be sure there is strict parallelism of table slots, rip fence, and saw blade; the miter gauge is accurately set perpendicular to the saw blade; and the saw has an accurate, repeatable setting of the blade's angularity.

Again, the Sears plywood and veneer blade is a good choice. Excellent Teflon®-coated blades are available with a large number of teeth and a set that will make very smooth, splinter-free cuts. You can add a scoring saw attachment to produce chip-free cuts on laminated boards.

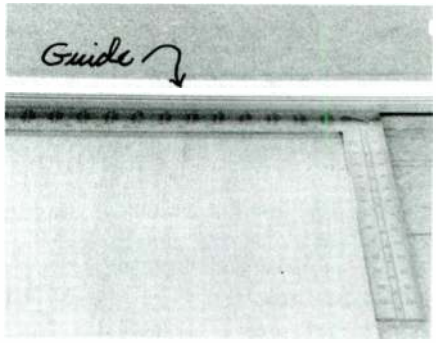

PHOTO 4: Checking to ensure that the guide is square to the edge of the stock.

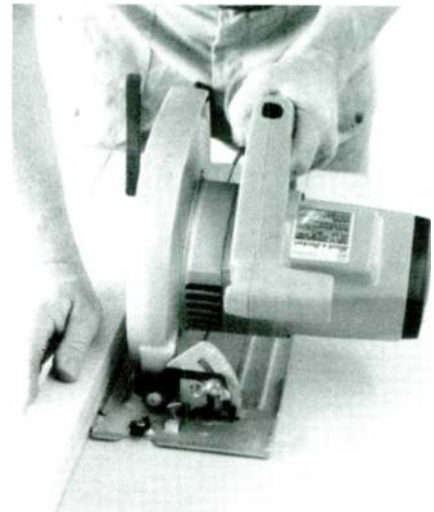

PHOTO 5: Using a straight board as a guide when cutting the stock. If you are cutting square pieces, the guide must be tested to establish that it is square.

SETTING UP

If you don't mind bending over, you can work with the boards on a flat floor which has a suitable protective covering. At my age this is usually a bad idea. The discomfort causes me to be careless, often creating expensive lessons I don't need to relearn.

You can make a good work surface by

PHOTO 6: Using a piece of angle iron/aluminum as a circular saw guide. Note the ear plugs: always wear them when making a cut. Also note how the middle of the guide is bowed up in the center due to clamping pressure.

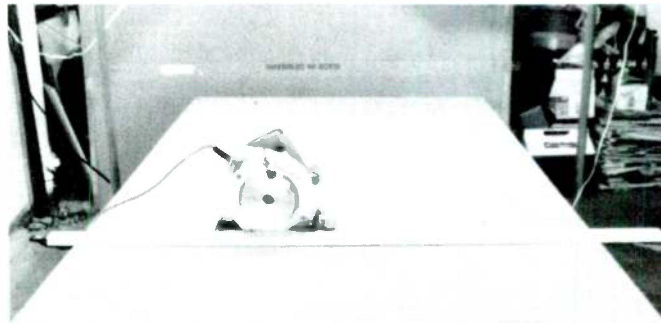

PHOTO 7: The Clamp 'N Tool guide in use. The clamping action is accomplished with the handle on the left-hand end of the guide.

laying two 2″ × 8″, 8-foot-long boards on top of two saw horses placed about 3.5′ apart. A usable spacing between the 2-bys is 20–24″. On top of the boards I lay an old piece of 4′ × 8′ ¾″ particleboard. Set the cutting depth of your circular saw to be about 1/8″ more than the thickness of the panel stock, as shown in *Photo 2*.

GUIDES & THEIR USE

To set the guides accurately you must know the offset from the edge of the saw base to the blade (*Photo 3*). If you are using a blade with teeth that have a set, measure to one of the teeth that is bent in the direction of the base edge that will be against the sawing guide (or, if the saw kerf is on the side of the cutting line next to the guide, to the other side tooth). I usually write the cutting offset on the guide with an indelible pen. You can also get the measurements from a piece of scrap on which you have made a cut, and establish what the offset of both sides of the kerf is from the guiding edge.

Wooden Guides: The simplest guide is a straight board properly clamped with C-clamps. The problem is that the guide board must be exactly straight. The most accurate way to check for straightness is to place the guide on edge on a flat surface, shine a light

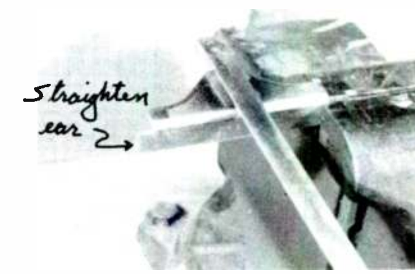

PHOTO 8: Straightening the ears of the circular saw guide.

PHOTO 9: The reconfigured saw guide. You must determine the exact placement and dimensions of the bends for your circular saw.

on one side, then look for light leaks on the other side. For short spans, your table saw top is a good flat surface, but it is often hard to find a flat surface for longer spans. As a test, you can hang a heavy weight on a string, and carefully bring the edge to be tested up next to the string.

When you set up the guide, be very careful to test that it is correctly aligned, i.e., square where it should be (*Photo 4*). Also, when laying out the panels to be cut, I mark an X on the side of the line where the waste is located. This helps keep me from cutting panels that are a kerf-width too small!

Once the guide has been carefully aligned and clamped, you are ready to make the cut (*Photo 5*). Be sure the edge of the saw base is pushed tightly against the guide throughout the cutting process. A trick you can use to help reduce veneer splintering is to pull the saw backwards, thus cutting

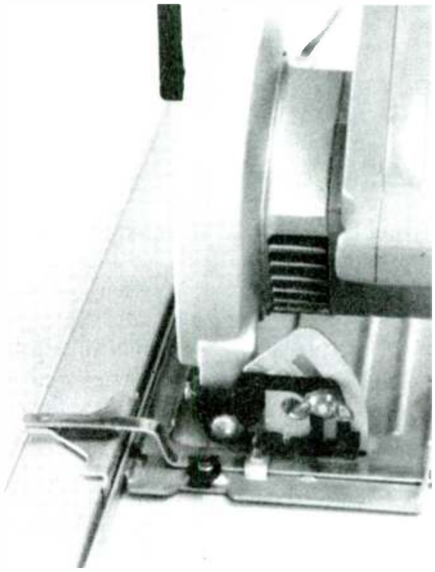

PHOTO 10: Using the bent guide to hold the circular saw base snug to the Clamp 'N Tool guide.

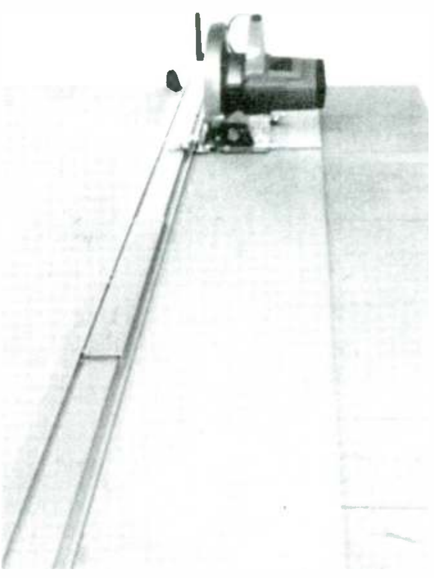

PHOTO 11: The Strate-Cut guide for making extra-long cuts.

SOURCES

Griset Industries
PO Box 10114
Santa Ana, CA 92711
(800) 662-2892
Tru-Grip Clamp 'N Tool

Micro Fence
11100 Cumpston St. #35
North Hollywood, CA 91601
(800) 850-4367
FAX (818) 761-3977

PHOTO 12: The support stand.

PHOTO 13: The new Micro Fence circle-cutting accessory.

downward on the top surface. With the board on the backup particleboard, you should have a splinter-free cut.

The saw blade must remain absolutely parallel to the guide edge. One way to ensure this is to place a second guide parallel to the first, but offset by just the width of the saw base plate.

Angle Iron/Aluminum Guides: Some of the trouble with using boards as guides can be overcome with a length of angle iron or aluminum. Just as with a board, check to be sure the piece is straight. (They seldom are.) The real advantage over wooden guides is, of course, once you have a straight one it will, with reasonable care, remain straight.

In use, as shown in *Photo 6*, the angle iron guide has the advantage of being easy to align. If the circular saw base doesn't slide smoothly along the guide, sand the side of the guide with 400-grit sandpaper, and then rub on a thin coat of soft soap or light oil.

A problem common to both board and angle iron guides is that they will bow in the center if the ends you are clamping extend over the piece you are cutting. In some cases the bow could be enough for the base of your circular saw to slide under the guide! The solution is to use blocks which are just a bit thinner than the stock you will be cutting under the ends. The saw is used in the same way as with the wooden guide.

Clamp 'N Tool Guide: A recent addition to woodworking tools is the Tru-Grip Clamp 'N Tool guide (about $40 for the 50″ clamp from Griset Industries, or most hardware stores and woodworking mail-order companies). This bar clamp eliminates the need for C-clamps to hold it in place, as shown in *Photo 7*. A groove runs along the edge of the guide, making it especially useful for our purposes.

A simple modification is required for a positive attachment of the saw base to the Clamp 'N Tool guide, and thus ensure a smooth cut. You first need to remove the bent ears on the slide, then make a couple of bends (*Photos 8* and *9*). The exact dimensions will depend upon your saw. Insert the modified guide into the normal slots in the saw base, but upside down.

You may need to cut the upper part of the guide slot on the saw base before you can insert the bent guide (*Photo 10*). When using this jig, the saw base is held firmly next to the guide, resulting in much smoother cuts. Clearly, you can make a similar jig for use with angle iron. If the jig binds in the groove of the Clamp 'N Tool guide, sand the slide and apply a light coat of soap or thin oil.

Longer Cutting Guides: You may sometimes be forced to make a cut in the panel's long dimension, but it is hard to find a straight guide for this 8′ span. With care, you can use a length of angle iron/aluminum.

A guide which works well for this purpose, but which is no longer being manufactured, is the Strate-Cut. I have seen this tool at garage sales and flea markets. As you can see from *Photo 11*, the Strate-Cut consists of two 4-foot-long sections of extruded aluminum, with a splice plate holding them together in the center when you need the long reach. It has a groove you can use for the bent guide in the same way as the Clamp 'N Tool. Other long guides are on the market, but they lack this groove.

OTHER SAW CONSIDERATIONS

As previously mentioned, the most accurate way to cut panels is to make slightly oversized pieces with the circular saw, and then make finish cuts on the table saw. While there is nothing special about the cutting on a table saw, it does have one useful addition: an adjustable roller stand.

The stand acts as a catch for your stock as it comes off the table of your saw (*Photo 12*). It is very handy for larger pieces, but there is one problem with its use. If the roller is not perfectly aligned perpendicular to the saw blade, it will pull your stock to the side and can cause a miscue. The stand is also handy to support overhanging stock on the side of your saw table.

When using a circular saw, the main thing to remember is to keep the saw snug against an accurately aligned guide. Personally, I prefer the bent jig with the Clamp 'N Tool guide. You can increase your success with smooth, splinter-free cuts by pulling your circular saw backwards between two parallel guides.

PS ON DRIVER HOLES

Micro Fence has added a circle-cutting guide as an accessory to their routing tool. You can use their new guide ($98 for the complete jig) to make driver holes without the modifications described in this column (*SB* 4/95, p. 40). It looks just as accurate as their fence (*Photo 13*)!

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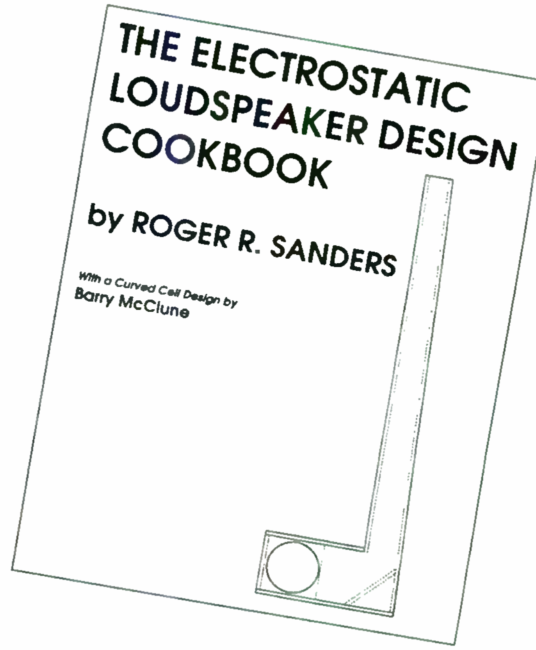

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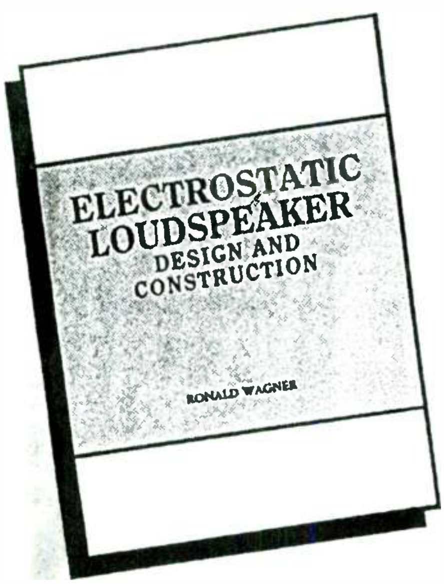

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Software Report

SOUNDBLASTER 16

By D.S. Jenkins

The SoundBlaster 16 digitizer card has generated considerable interest lately for its use in evaluating speaker and amplifier systems. My question is just how useful is a card primarily designed for computer sound reproduction for use in other applications. Used with the Wave Studio software from Creative Labs, does it have the performance needed for critical acoustic systems evaluation requiring quantitative and reproducible results?

TESTS

The following performance data will assist you in evaluating whether this card and software meet your individual requirements. I ran the described tests with a Stanford Research Systems Function Generator Model DS 345 and FFT Spectrum Analyzer Model SR 760. I tested pulse performance with an Hitachi VC-6025 50MHz DSO, jitter data with a Fluke PM 6680 High Resolution Counter/Timer, and data reduction using both instrument software and my own software.

I describe several groups of tests. The first is the straight frequency response of the card I/O, showing data for both the card real-time output and for the digital replay. I measured data for a 10–250Hz sweep, a 200–20kHz sweep, and (limited) for a 200–30kHz sweep. Digital replay makes use of several of the digitizing options of the Wave Studio software. Each response test reflects a 20-second sweep of the frequency limits.

A second group of tests measures total harmonic distortion in both the real-time output and in the digitized replay. The third group of tests relates to pulse performance. A final group determines the amount, if any, of "jitter" in the digitized replay.

FREQUENCY RESPONSE

Figure 1 is a "straight through" response of the card for 200–20kHz. Card output is from the audio output connector into an 8Ω resistor. You can see the output volt-

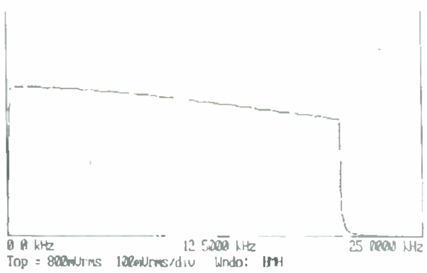

FIGURE 1: Frequency response 200–20kHz, real-time output.

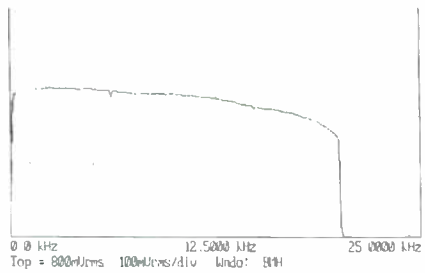

FIGURE 2: Frequency response 200–20kHz, digitized replay stereo, 16 bits, 44kHz.

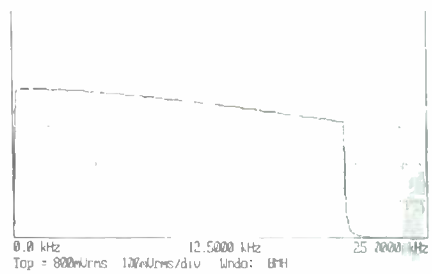

FIGURE 3: Frequency response 200–20kHz, digitized replay stereo, 8 bits, 44kHz.

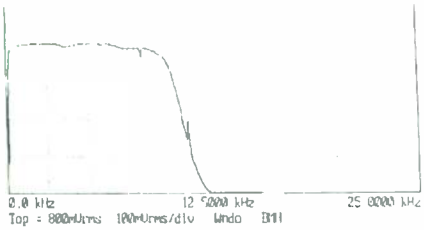

FIGURE 4: Frequency response 200–20kHz, digitized replay stereo, 8 bits, 22kHz.

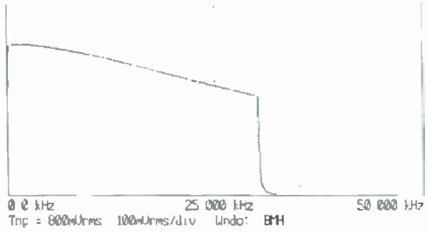

FIGURE 5: Frequency response 200–30kHz, digitized time output stereo, 16 bits, 44kHz.

age level on the various plots. Input voltage for all tests was 100mV RMS except for the pulse input, which was a positive 100mV step for 1ms. I set the gain through the card at ×2 for both recording and play through. This resulted in an output signal level of 400mV RMS with zero card throughput attenuation.

Figure 2 is the digital replay of the 200–20kHz sweep. I used the Wave Studio digitizing options "Stereo, 16 bits, 44kHz;" you'll note some low-frequency and high-frequency fall off.

Figure 3 is a digital replay of the 200–20kHz sweep using the "Stereo, 8 bits, 44kHz" recording option. Note the small difference between these two outputs. In fact, they are identical in the data overlay.

Figure 4 shows the same sweep, 200–20kHz, using the "Stereo, 8 bits, 22kHz" digitizing option. The low-frequency response is the same as the 44kHz rate, but the upper-frequency limit is an expected 9kHz.

Figures 5 and *6* are the straight-through response and the digital replay for a sweep of 200–30kHz. For *Figs. 4* and *6* note the "discontinuity" in the response during the amplitude fall-off part of the curve. An expanded plot is shown in *Fig. 7*. For a narrow band of frequencies above what should be the usable frequency limit of the card, the digitizer has problems, as the real-time output on the scope shows. The amplitude modulation of the output indicates this characteristic, as the digitizer input moves through the critical frequencies, and may be the result of either the internal filter characteristics or possible aliasing in the digitizing circuit.

I repeated this test several times, and the effect is real. If you require data from this type of I/O, beware this characteristic.

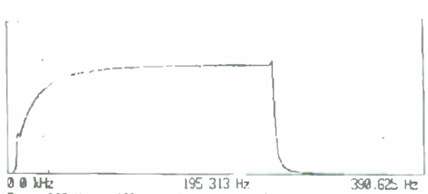

FIGURE 6: Frequency response 200–30kHz, digitized replay stereo, 16 bits, 44kHz.

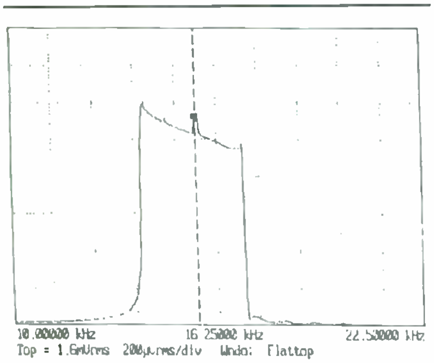

FIGURE 7: Expanded sweep showing amplitude distortion, stereo 16 bits, 44kHz.

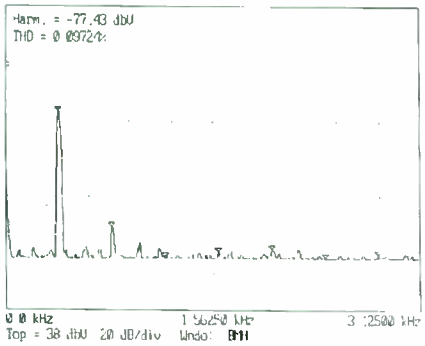

FIGURE 8: Harmonic distortion 400Hz, signal generator direct to FFT analyzer.

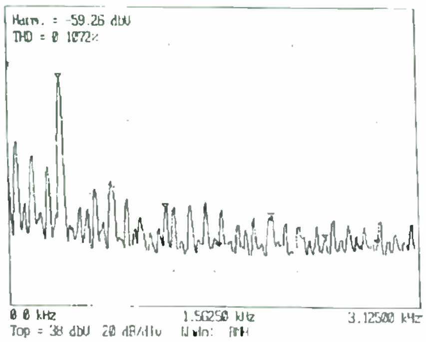

FIGURE 9: Harmonic distortion 400Hz, real-time output.

HARMONIC DISTORTION

Figures 8, 9, 10, 11, and *12* show the harmonic generation for the "straight through" card output and for the digitized replay. All tests included the "Stereo, 16 bit, 44kHz" digitizing option. *Figures 8, 9,* and *10* are for 400Hz.

Figure 8 results from connecting the signal generator directly to the FFT analyzer. You should compare this with the other curves for an idea of the noise generated by the card. *Figure 9* displays the real-time card output, while *Fig. 10* is the digital replay. Not only is the noise level increased, but the harmonic generation in the digital sample goes from about 0.1% to over 5%. *Figures 11* and *12* are for 5kHz and give about the same result, card output THD at 0.1%, while the digital replay is more than 7%.

Figures 13 and *14* show the low-frequency response performance. The sweep here is from 10–250Hz in the same 20-second period. *Figure 13* is for 16 bits, 44kHz, and *Fig. 14* is for 8 bits, 11kHz. Not much difference is noted here—a lot of 60Hz in both outputs with a noticeable second harmonic in the 11kHz record, as seen in real time before the sweep-averaging started, but not noticeable on the plot.

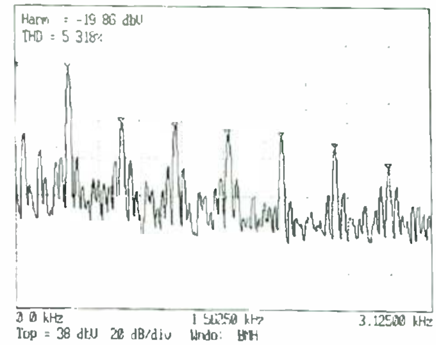

FIGURE 10: Harmonic distortion 400Hz, digital replay stereo, 16 bits, 44kHz.

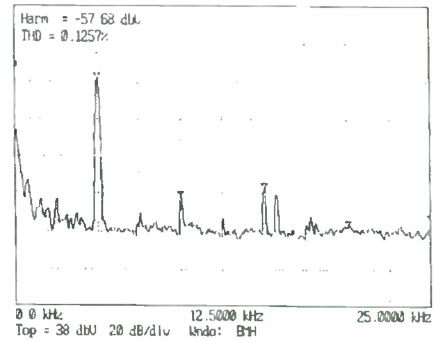

FIGURE 11: Harmonic distortion 5kHz, digital replay stereo, 16 bits, 44kHz.

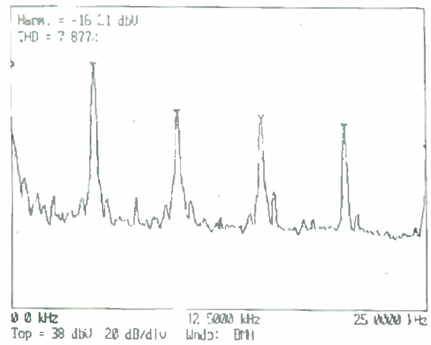

FIGURE 12: Harmonic distortion 5kHz, digital replay stereo, 16 bits, 44kHz.

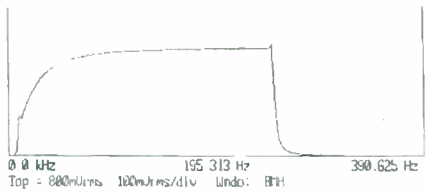

FIGURE 13: Frequency response 10–250Hz, digital replay stereo, 16 bits, 44kHz.

Pulse performance is shown in *Figs. 15, 16,* and *17.* The input was a positive going 1ms 100mV square pulse (*Fig. 15*). The card output is *Fig. 16* and the digital replay is *Fig. 17.* This response is typical of capacitance-coupled circuits with the digital replay showing additional low- and high-frequency drop off. When using this card and software for impulse tests, you need to consider this performance. Also note that the direct output is out-of-phase with the input, while the digital replay is in-phase with the input.

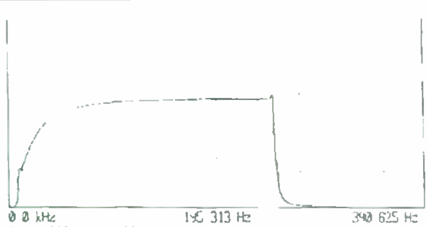

FIGURE 14: Frequency response 10–250Hz, digital replay stereo, 8 bits, 11kHz.

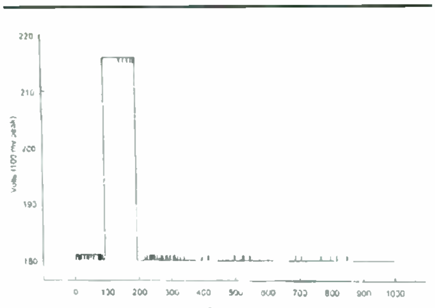

FIGURE 15: 1ms square-wave pulse, input pulse.

JITTER

Figures 18 and *19* show the jitter produced at 100Hz. The card is remarkably stable. While *Fig. 18* shows about ±1Hz, *Fig. 19* shows ±0.1Hz. Interestingly, in *Fig. 19* the input signal of 100Hz is in the "null" section of the data with the jitter values almost equally present on both sides of the 100Hz signal. *Figure 18,* however, shows a different characteristic with all three "side bands" present. In any case, these dispersions are nearly nonexistent. It is difficult to imagine an application in which this would make any difference at all.

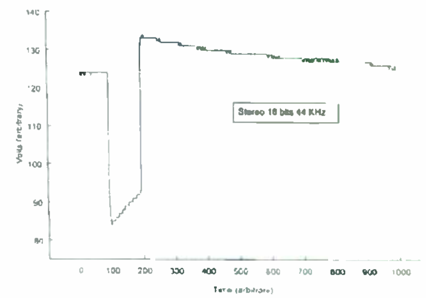

FIGURE 16: 1ms square-wave pulse, real-time output.

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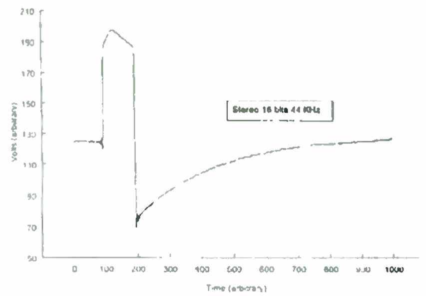

FIGURE 17: 1ms square-wave pulse, digitized replay.

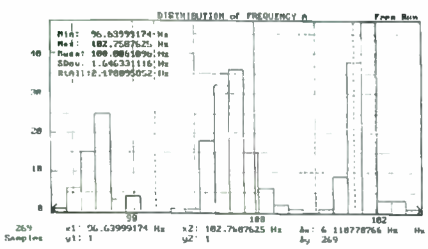

FIGURE 18: Jitter test #1, 100Hz, digital replay stereo, 16 bits, 44kHz.

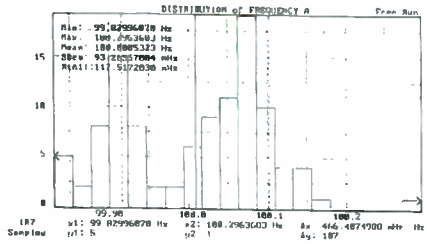

FIGURE 19: Jitter test #2, 100Hz, digital replay stereo, 16 bits, 44kHz.

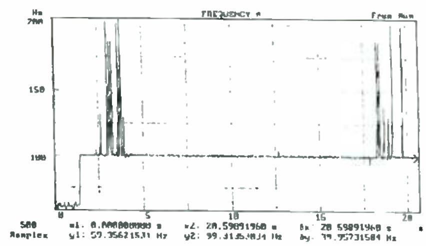

FIGURE 20: Jitter test #2, 100Hz, real-time output.

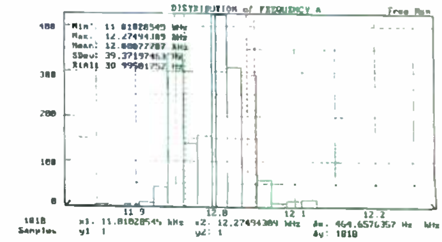

FIGURE 21: Jitter test, 12kHz, digital replay stereo, 16 bits, 44kHz.

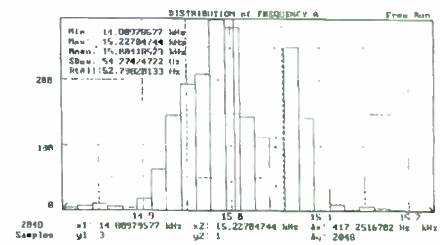

FIGURE 22: Jitter test, 15kHz, digital replay stereo, 16 bits, 44kHz.

Random noise is a problem. It occurs both on straight-through and digital replay. *Figure 20* is a real-time record for the 100Hz jitter test. You can see a lot of random noise spikes in this record. Some of these also can be "heard" over the output. They also appear in the digitized record, but in a reduced number. The card I/O filter used in the digitizing circuit probably suppresses most of these transients.

Figures 21 and *22* are for jitter at 12k and 15kHz, respectively. At most, the jitter at the high-frequency end of the card is 50Hz. This could be related to 60Hz power input, but that is difficult to confirm with these limited tests. In a practical sense, this, like the 100Hz range jitter, seems to be trivial. I performed these tests over a two-week period. It is not apparent why the difference in the 100Hz response occurs. Also, I don't know whether the card uses the computer CPU clock, a CPU timer register, a clock on the card, or some combination. Different computers may present varying jitter characteristics.

One characteristic measured during these tests, for which no documentation is given in this article, is the crosstalk between channels. I measured the crosstalk at about 1.3%, via a 100mV input on one channel and no input on the other. Both channels' gain was set to 8. On the input channel the output was 890mV. The other channel gave 12mV of the same signal (870Hz). In my opinion this is very good performance for a card and system in this price range.

In summary I believe that this card and software perform their intended functions well, serving as an audio tool for computer music and voice reproduction. The software is responsive, and the editing capability seems to work well. However, if you use this card for critical measurements, or to evaluate design options, you need to be aware of possible limitations.

SB16 DIGITIZING & PLAYBACK

During my evaluation of the SB16 hardware and the Wave Studio software, I questioned the actual digitizing rate. If you use the same board for both the digitizing and the playback, the frequency error should be minimized, but what are the conditions when you save a digitized file to disk and play it back on another machine? Is the quoted rate the same for all systems? Or, more interestingly, if the software synthesizes the digital data, will the playback frequency be as expected from the quoted digitizing rate?

The quoted values for the SB16 are 11,025, 22,050, and 44,100 samples per second. Inputs generated for these sample rates, when played back through the system, should hold a precise frequency fidelity. The following tests respond to this question.

To generate digital data for playback, I wrote a program to allow input of any combination of frequencies and sample times, on either or both channels. The enclosed code listing (*Table 1*) will allow you to synthesize such a combination.

The inputs are: sample rate, duration of the sample, relative amplitude of the sample, and frequency. You can input any number of entries for each channel (a maximum of ten per channel, unless you add a dimension statement).

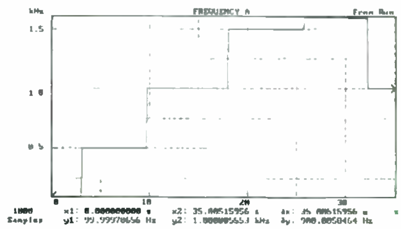

FIGURE A: Calibration results.

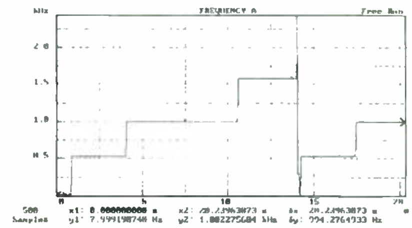

FIGURE D: Synthesized 10kHz rate.

I initiated testing with calibration of the frequency meter, in this case a Fluke PM6680 High Resolution Counter, which comes with the Timeview software that provides IEEE 488 bus interrogation and provides various formats for the data collected. *Figure A* is the calibration record.

For this test I used a Stanford Research Function Generator DS345 to input a chain of 500, 1k, and 1.5kHz sine-wave segments. The results show these frequencies to be "dead nuts" in calibration with the Fluke. In theory these two instruments could be "off calibration" by the same amount, but a check of the Fluke with other sources shows it to be accurate to within at least 10E–5Hz.

In the next test I attempted to synthesize a waveform using the referenced code by inputting the digitizing rate and calculating the sine-wave amplitude for this rate. Using and displaying the code with Wave Studio will show the synthesized waveform. This synthesis is obviously independent of any external frequency reference. If it is played back at the input reference rate, the frequency output of the SB16 D/C converter will be the exact frequency desired.

Figures B, C, D, and *E* give the results from the SB16 system. *Figure B* is a playback using the SB16 11,025 rate. The curve shows that the output frequency is too low, with the 1kHz synthesized input resulting in a 920Hz output. Since the SB16 does not give an option on the digitizing rates, you can synthesize the waveform at other rates, play it back at the SB16 reference 11,025 rate, and, when you achieve frequency correspondence, the playback rate equals the synthesized rate.

Figures C, D, and *E* are for synthesized rates of 9k, 10k, and 13kHz. The 10k rate hit the 1kHz input. However, the 500 and 1.5kHz outputs are slightly high (*Fig. D*). The 1.5kHz playback is about 5% or 78Hz high. This is not a bad result, but interesting because the playback is not linear in the sense that if it is "dead on" at 1kHz, why is it off at both higher and lower frequencies?

In any event, the fidelity of playback from externally synthesized records may have some error. As concluded in the earlier tests on this system, you should recognize these characteristics and evaluate any results from using the SB16 as an instrument for critical measurements with this in mind.

If you use the code provided with this article, you can generate your own data. I am interested in seeing how wide a dispersion exists from board/CPU to board/CPU, and look forward to other readers sharing their results.—**DSJ**

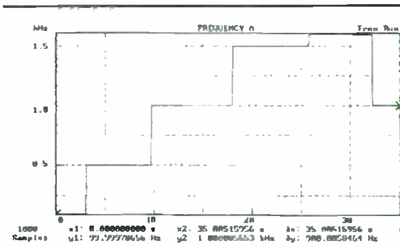

FIGURE B: Synthesized 11,025 rate.

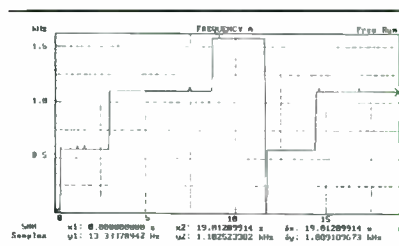

FIGURE C: Synthesized 9kHz rate.

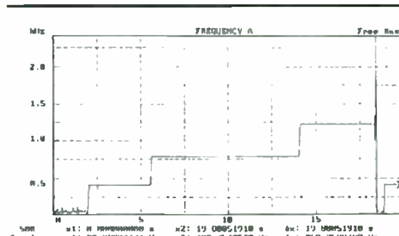

FIGURE E: Synthesized 13kHz rate.

TABLE 1

TWO-CHANNEL PLAYBACK LISTING

```
' this will take a series of frequencies for each channel with the
' playback time. output is sine waveform
COLOR 15
c = 2 * 3.1416
CLS
PRINT "Play this data back with STEREO 8 BITS and the selected digitizing rate"
PRINT "Wave Studio digitizing rates are 11025, 22050, 44100 "
INPUT "Select digitizing rate"; drate
INPUT "How many Left channel frequencies will there be "; numl
FOR i = 1 TO numl
PRINT USING "For Left frequency number ##"; i
INPUT "Input duration (sec), relative amplitude (0 to 1), frequency (Hz)"; dl(i), al(i), fl(i)
pl(i) = drate / fl(i)
timl = timl + dl(i)
NEXT i
PRINT
INPUT "How many Right channel frequencies will there be "; numr
FOR i = 1 TO numr
PRINT USING "For Right channel frequency number ##"; i
INPUT "Input duration (sec), relative amplitude (0 to 1), frequency (Hz)"; dr(i), ar(i), fr(i)
pr(i) = drate / fr(i)
timr = timr + dr(i)
NEXT i
PRINT
IF timr = timl THEN tim = timl: GOTO 10
IF timl > timr THEN tim = timl
IF timr > timl THEN tim = timr
10
bytes = drate * 2 * tim
PRINT USING "Memory requirements will be ######## bytes"; bytes
PRINT
COLOR 11
INPUT "Name of file for play back data "; name$
PRINT
COLOR 15
OPEN name$ FOR BINARY AS #1
il = 1
ir = 1
ctiml = dl(1)
ctimr = dr(1)
100
FOR b = 1 TO bytes / 2
stim = tim * b / drate
IF stim > timl THEN al(il) = 0: pl(il) = 1: GOTO 150
IF stim > ctiml THEN ctiml = ctiml + dl(il + 1): il = il + 1
150 'IF il > numl THEN al(il) = 0: pl(il) = 1
l1 = l1 + 1
IF l1 > pl(il) THEN l1 = 0
x1 = 128 + al(il) * 127 * SIN(l1 * c / pl(il))
x = INT(x1)
a$ = CHR$(x)
PUT #1, , a$
200
IF stim > timr THEN ar(ir) = 0: pr(ir) = 1: GOTO 250
IF stim > ctimr THEN ctimr = ctimr + dr(ir + 1): ir = ir + 1
250 'IF ir > numr THEN ar(ir) = 0: pr(ir) = 1
r1 = r1 + 1
IF r1 > pr(ir) THEN r1 = 0
x2 = 128 + ar(ir) * 127 * SIN(r1 * c / pr(ir))
x = INT(x2)
a$ = CHR$(x)
PUT #1, , a$
300
NEXT b
CLOSE #1
PRINT "Data saved to "; name$

END
```

SB *Mailbox*

MORE ON PSPICE

The Jung/Campbell letters on PSpice SPICE modeling ("SB Mailbox," *SB* 6/94, p. 58) has another chapter, courtesy of the PSpice vendor, MicroSim. I wrote to them to ask if they would allow the JFET models in question to be reproduced here. They agreed, and the models are shown in *Table 1* with the express permission of MicroSim Corp. Interested readers can contact them for further information at 20 Fairbanks, Irvine, CA 92718, (800) 245-3022. Our thanks to MicroSim for this courtesy.

Walt Jung
Fallston, MD 21047

TABLE 1

JFET MODELS

```
* Library of Japanese Junction Field-Effect Transistor (JFET) Models

* Copyright 1994 by MicroSim Corporation
* Neither this library nor any part may be copied without the express
* written consent of MicroSim Corporation.

* $Revision:   1.0  $
* $Author:   ANW  $
* $Date:    01 Oct 1993 14:16:02  $

* The parameters in this model library were derived from the data sheets for
* each part.  Each part was characterized using the Parts option.

*$
.model J2sj74    PJF(Beta=92.12m Rs=7.748 Rd=7.748 Betatce=-.5 Lambda=4.464m
+                Vto=-.5428 Vtotc=-2.5m Cgd=85.67p M=.3246 Pb=.3905 Fc=.5
+                Cgs=78.27p Isr=129.8p Nr=2 Is=12.98p N=1 Xti=3 Alpha=10u Vk=100
+                Kf=26.64E-18 Af=1)

*$
.model J2sj103   PJF(Beta=2.197m Rs=76.76 Rd=76.76 Betatce=-.5 Lambda=735.3u
```

REDUCED ROOM EFFECTS

The most critical component in an audio reproduction system is the listening room. This change to Bill Waslo's observations[1] is now certain, due to recent work by Schuck, Olive, et al., on the Athena Project[2], which proved there is a big listening room problem. Claude Fortier and Pierre Côté researched how to fix this in the new listening room (typical, really) at the NRC in Ottawa.[3] Their findings show boundary reflections causing ±6 to ±11dB response errors below 400Hz, ±3dB over 400Hz–2kHz, and less than ±1dB error above that, given excellent loudspeakers and typical listener positions in typical rooms.

Smoothing the overall response to better than ±1dB (±2dB under 200Hz) is easily done with an adaptive DSP filterbank (low-cost in a few years) if the loudspeakers stay put, the room is reasonably damped, and the listener is within a 20″ × 40″ area, 60″ from any wall. This technology will arrive in the studio monitor market, and can be recalibrated for new locations in a few seconds (press the button), with a mike at the listener's head location.

Since this method is not here yet, and the direct/reflected sound balance is not affected either, we need to consider alternatives. Knowing that the big problem is under 400Hz, and that acoustic absorption is cheap and good above that, we can always fall back on absorption. The best way to raise the direct sound percentage is with directional speakers. We can use large PA-type horns. Too ugly? Then use a directional array. But first, let's preserve good stereo imaging by using only two tweeters. That leaves the 20Hz–3.2kHz region.

Now imagine a skinny floor-to-ceiling box, tweeter in the middle, in an MTM arrangement. (The usual MTM has wide vertical directivity because of the T leading the M by 90°, defocusing it.) Recess this box into the wall to avoid reflections and diffraction. A typical MTM array, using 4″ drivers, is about 12″ high, total.

At 3.2kHz that causes a 50° wide main lobe and no serious secondary lobes if the acoustic origins are aligned, and the crossover has 0° phase shift.[4,5] This is perfect for avoiding floor and ceiling reflections at a 10′ listening distance, except that at 6kHz a dome tweeter is omnidirectional. Instead use a horn tweeter with a 60° directivity or a dome tweeter recessed into the baffle to align its origin with the mid driver (about 1″). (Why have I never see this done in *SB*?)

Then you can form a short and wide horn with modeling clay and control the directivity with foam. This works well on column-type PA speakers I build to render them more constant-directivity. Take your time getting the contours right for flat response. Check with 1/3-octave pink noise. This is not really difficult.

Next, to maintain the 50° radiation pattern one octave lower, 1.6kHz, the array must be twice as long, 24″. So add two more 4″ drivers, symmetrically. Double that for 800Hz, 48″ long, or add two more for 400Hz, 96″ long. Now we have near-constant directivity, a 200Hz cutoff five-way array, and four octave-wide bandpass midranges. Finish with a bass section and in-phase crossovers.

This is a properly designed column PA speaker with frequency tapering to control the vertical radiation pattern. Power tapering without frequency rolloff has been tried, but is inadequate. Simple low-pass filter tapering results in 3dB bass gain per section.

Fiberglass pads of varying thickness are used commercially to effect low-pass filtering, causing muddy bass, and imprecise cutoffs. That's why on this continent column speakers have mostly a bad name. There is no inherent problem, using hi-fi drivers and accurate filters. Column speakers have a broad horizontal pattern, due to small drivers in a narrow box.

We've only achieved good focusing down to 200–400Hz with a 96″ high array, so we can curve the array to make the lower frequencies lead the treble. This will focus the pattern more, and we can shrink the height.

Let's try a 45° phase angle lead at each crossover point. This is 0.5″ at 3.2kHz, to cut the inner 12″ MTM array down to 6″. If it's difficult to fit three drivers, try overlapping, or a horn, or a 1.6kHz first crossover. That means a total of about 1.5″ recess at the acoustic origins of a dome or horn tweeter, and additional 1″, 2″, and 4″ offsets for the longer MTM sections of 12″, 24″, and 48″, respectively.

Add a sixth (96″) section, with 8″ offsets to achieve good focus down to 200Hz, and a 100Hz cutoff on focusing. This also accounts for the equidistance curve to the listening position of about 9″ max for a 10′ distance. That totals 15″ offset (1 + 2 + 4 + 8) at the ends of the array. We still need to add a subwoofer, too. I recall two shorter versions in *SB* around ten years ago.[6]

Now we need to make six crossover filters for this seven-way array. Subwoofer slopes can be 6, 12, or 18dB/octave, and the mids are bandpassed at 150, 300, 600, 1,200, and 2,400Hz, with an in-phase crossover of any order. The tweeter high-pass can have a 6dB slope with low-power use (16W).

If the filter slopes are all 6dB/octave, we have a linear-phase system, which should be able to reproduce a square wave reasonably well. This still requires many expensive parts. Better use op amps and cheap and accurate RC filtering in an active crossover array.

Since we're using op amps, why not successively delay each higher crossover output by 45° to replace the mechanical offsets with electrical ones to produce a straight and flat speaker baffle? This can be done with all-pass (RC delay) filters,[7] or some obscure crossover type. I think the limit is 45°, due to response ripple or lobing. If so, this array will average a 60° beam focus, at 6dB down.

I know this works well in the four-way PA lectern systems I build. The column speaker below the lectern top is 42″ high, tweeter at the top, has a 300Hz focus cutoff, and bass and treble controls. All-pass filter delays to the mids and tweeter permit me to play with the focus angle, and even tilt the beam up or down to suit some venues. I run the mids with full bass (9dB gain) to compensate for the focus loss at the low end, 4″ driver rolloff, and to keep it simple. Experimentation is called for, since this is virgin territory.

For your seven power amps per channel, use car-audio amps, some of which (Denon, Earthquake, Hifonics, JBL, Macrom, Precision Power) have six or seven channel outputs at 40–60W each into 4Ω, with quite decent hi-fi specs. You'll need four 14 or 15V, 20A regulated power supplies, too. This is enough power, given the 6dB focus gain (roughly), even if each section gets only 50W.

Since headroom is a problem in 14V circuits, toss in a soft-knee limiter/compressor, set for 2dB below 50W. It costs about $10 per stereo pair, or $60 for 14 channels, for DIY. This makes the system teenager-proof, with 30W max to the tweeters and 15W into the mids. This is also very good for acoustic feedback limiting in PA use or rentals. I predict that in a few years most amps will offer built-in limiters as a courtesy, to save your ears.

Maximum SPL is set by the 4Ω tweeter (no focus gain), which at 96dB/W/m results in 101.5dB (96 + 15 – 9.5), 104.5dB for the stereo pair, or 6–12dB higher if using a horn. The mid drivers should be parallel and 8Ω each. At 90dB/W/m your ears receive 104.5dB per section (90 + 15 + 6 – 6.5)[8], 107.5dB per stereo pair, or 6–12dB higher if using 96–102dB/W/m drivers. Four-inch drivers are excursion limited below 100Hz, so use good ones for at least the 100–200Hz octave with an X_{MAX} of 3mm. The 100–200Hz drivers radiate into ¼ space in an 8′ high room, and should then be 3dB less sensitive. Put the array flush in a wall to avoid back reflections and diffraction.

Below 100Hz you must consider omnidirectional, with a major subwoofer to equal the array capability, using 50W/channel limited to 30W as before. Above 100Hz you have 16 in^2 of nearly excursion limited cones per side, so at 25Hz you need 256 in^2 (16 × 16). In a closed box you need two 15″ per side with X_{MAX} of 3mm, or one 12″ with X_{MAX} of 9mm or more. In a tuned box you could use 12″ with X_{MAX} of 3mm, or 10″ with X_{MAX} of 4mm min. The sub box should be in a corner to excite the fewest room modes.

Subwoofer output is then into 1/8 space, rather than ¼ space for the 100–200Hz drivers, and ½ space for all other drivers. For the sub there is no focus effect as in the array, but 1/8 space versus ½ is good for 6dB also. The sub SPL for 90dB/W/m drivers is then 107.5dB max per side (90 + 15 + 6 – 3.5)[8] in a stiff (low-loss) room of about 3,500 ft^3.

Assuming broadband music, this system can produce 117.5dB (104.5 + 10 + 3), with no single annoying frequency louder than 102–108dB. Using more sensitive drivers (10dB), you can match the loudest acoustic 127dB source material.

You can build 16W/channel amps (limited to 10W) very inexpensively (under $6/channel), using car audio bridging-mode power amp chips. Max SPL goes down by 4dB, or stays the same at a 7′ distance in a small room. The price/performance ratio is right—under $250 for electronic parts, including one 14 or 15V/20A power supply, plus low-power (low-cost) drivers. FM (Doppler) and IM distortion in the array will be lower (although loud) at your chair than anything you can buy now, simply due to the seven-way design.

This array achieves reflection avoidance in the vertical plane only. Extending the tapered array horizontally to avoid side-wall reflections is not plausible below 800Hz, since a stereo image will suffer with 24″ or wider nonuniform (cone) sources in each channel. Human audition is very sensitive to horizontal detail, and insensitive to vertical cues. To remain equidistant to your ears, a semicircular baffle is needed, unless you split the mids and apply the time delay trick again. This is not only too complicated, but the delay accuracy affects the imaging.

As noted earlier, using side wall absorption to fix reflections is easy and changeable, by adding heavy drapes or absorbing panels. Next, put a thick carpet on the floor, acoustic tiles on the ceiling, and heavy-duty anechoic treatment on the wall(s) behind you.

Another, less rigorous approach is to

include a floor-standing (50″ tall), slim, three-way speaker, with an MTM top end. Then replace the single high-power 8″ or 10″ woofer with four series/parallel connected 5.25″ or 6.5″ woofers, all in a row. This will blur the unavoidable floor reflection very nicely, with bass mainly limited by f_S and V_B. This blurring also works on the 96″ array above.

You can achieve excellent focusing from a 2 × 8′, five-way MTM, electrostatic array. Start with a (9″ equidistant) curved baffle, no electronic delays, and a small central tweeter, crossed over at 3.2kHz. The three midrange sections cover an octave each, and are 12″, 24″, 48″ high as before, made of two 4″ × 12″, 6″ × 24″, or 12″ × 24″ panels, and get down to 1,600, 800, and 400Hz, respectively.

Then add two or more 6.5″ cone drivers at the top and bottom for the 100–400Hz range. This array will be well-focused down to 200Hz vertically and 800Hz horizontally. The panels must be angled to face the listener. Martin-Logan already makes large curved multiway electrostatics (not yet MTM) like this.

All the ploys thus far do not have large side lobes, except a horizontally focused cone driver array, so any remaining reflections are predictable. Low-cost drivers can be used where each sees a narrow spectral range, at low power. The electronics are undemanding and easy to build. Therefore, the cost should be equivalent to conventional systems.

In conclusion, directional arrays are feasible and have higher efficiency and less neighbor interference, especially with acoustic absorption; room honk can be eliminated; sound quality is less room dependent; and imaging is better. This setup also features damage resistant overdrive, less piercing-tone annoyance, lowest FM and IM distortion, and excellent driver damping due to direct amp connection. This is a mixed bag, for sure, but I only expect incremental gains in these high-tech times. PA column makers (University, Bose, myself, and others) have built somewhat similar arrays, but not properly designed for hi-fi use.

The final solution? Imagine a globe- or egg-shaped shell chair. Add two small 90dB/W/m MTM arrays at the inside top/front, 2′ apart. Add a modest but proper 96dB/W/m dual voice coil 12″ subwoofer in the seat base, ducts, padding, and a reading light. Then enjoy really great sound and a massage, without bothering anyone. With 60 + 60W of amp power into the 35ft³ "room," and 6dB acoustic gain at the ¼m listening distance, you'd achieve 117dB SPL.

John Fourdraine
Toronto M4B 1B4

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8. A fudge factor, composed of −9.5dB for the loss due to a 10′ listening distance, reduced by 6 and 3dB over the range of 20–200Hz, due to room gain (reflections). These values are less than the 9dB @ 100Hz and 6dB @ 200Hz measured by F.E. Toole and reported in JAES, Vol. 34, pp. 323–348, May 1986, due to an evident need for more than average absorption.

M/PP COMMENTARY

There are several errors in Bill Waslo's mike/probe preamp schematic (Fig. 2, *SB* 5/95, p. 21):

1. The −20dB and −40dB numbers shown next to SPDT switches S2 and S3 are inaccurate when the switches are in certain positions. The value of −20dB *is* accurate for S3 when SPDT switch S4 is in the MIC position (rightwards on the schematic; leftwards in his Fig. 2). Then attenuation between the input of U1C and the output of U1B will be approximately −20dB, or

$$20\log_{10}\left(\frac{.475}{.475+4.32}\right)=20\log_{10}\frac{.475}{4.795}$$
$$=20\log_{10}(.0991)\cong 20\log_{10}(10^{-1})=-20$$

More accurately, the value is −19.9dB, but −20dB is close enough, and Fig. 2 gives −20dB. Furthermore, as the text notes, attenuation from the probe is balanced by amplification from noninverting op amp U1B. So attenuation between signal source and the input of U1C is also −20dB.

Now the errors start. When S4 is thrown to PROBE position, input enters through J2 and presumably an IMP-type probe with its 47.5k resistor. This portion of the circuit lacks an input op amp, and its attenuation is produced only by its 47.5k–2.21k resistive divider. The values are

$$20\log_{10}\left(\frac{2.21}{2.21+47.5}\right)=20\log_{10}\frac{2.21}{49.71}$$
$$=20\log_{10}(.0444)$$
$$=20\log_{10}(4.4\times10^{-2})$$
$$=20[\log_{10}(4.4)+\log_{10}(10^{-2})]$$
$$=-40+20\log_{10}(4.4)$$
$$=-40+20\times(.64787)=-27\text{dB}$$

An attenuation of −27dB does not even approximately equal −20dB. (By the way, Fig. 2 shows S2 in this position.)

Attenuation values in Fig. 2 are also given inaccurately when one uses the −40dB position of either switch S2 or S3. Moreover, attenuations produced from the calibration jack (J3) are also in error for both S2's −20dB and −40dB positions.

From a design viewpoint, the attenuation problem arises because the circuitry following jacks J2 and J3 is not the same as that connected to J1, which leads to op amp U1B; J2 and J3 do not have op amps. One (obvious) solution is to connect J2 and J3 to op amp circuitry that exactly replicates J1 and U1B. However, if precision is truly important—rather than being merely show-offish—then you worry about resistor temperature coefficients and matching the resistors to better than 1%. A true precision analog design would probably involve a differential amplifier using JFET input op amps.

2. Using Mr. Waslo's IMP-type test probe (Fig. 1) connected to MIC input jack J1, the input signal is attenuated through a resistive voltage divider consisting of a 47.5k resistor in the probe and R10, a 2.21k resistor. R10 is returned not to ground but to the output of op amp U1A, which is at 2.5V DC. Whether or not this DC bias affects the signal source depends on its output impedance. R10, like R14 and R15, should be returned to ground.

3. A minor matter comes last. Input jacks J1, J2, and J3 need ground connections between them and the input probe. The small triangles slightly below output jacks J3 and J4 probably are ground connections, but are not labeled.

As published, Mr. Waslo's Mike/Probe Preamp will give seriously inaccurate or misleading results. It is clumsy and unwise to attenuate one signal by −20dB and a calibration signal by −27dB. At worst, the design is merely incompetent and so cheaply made that it is useless. It is also possible that the unequal attenuations serve functions I do not understand, but if so the article fails to discuss them.

Timothy Perper, PhD
Philadelphia, PA 19147

Contributing Editor Bill Waslo responds:

Dr. Perper first points out errors *in the gain scaling circuitry of my Mike/Probe Preamp design which may cause readers "seriously inaccurate or misleading results." It is unfortunate that Dr. Perper chooses to imply that* his *erroneous analysis of the switched attenuator section of the circuit is possible evidence of an "incompetent" design or one which is "cheaply made" (and might be "merely show-offish" for using 1% resistors). Perhaps he should instead have asked for clarification about the operation of the circuit.*

First, some basic analog theory. The dB terminology is used to express a gain or level relative to some 0dB reference value. The reference gain in the case of a relative gain switch as supplied on the Mike/Probe Preamp is rather obviously that which results in its 0dB position. In the case of S3, this is the center-off position in which the wiper does not connect to either R21 or R20. The gain in this position (for a given input) must be used as the reference when analyzing what the attenuator switches do in other positions.

Dr. Perper seems to follow this technique when analyzing the circuit for S4 in the MIC position. When S3 is in the center-off 0dB position, there is very little attenuation (about 0.037dB) between U1B Pin 7 and U1C Pin 10, because the only load (R19, 1M) is so much larger than the source impedance (R12, 4.32k). In the audio band, C9 and C10

TABLE 1

VOLTAGE GAINS

	0dB	–20dB	–40dB
MIC Input	+54dB	+34dB	+14dB
MEAS Probe Input (through 47.5k probe resistor)	0dB	–20dB	–40dB
CAL Probe Input (through 47.5k probe resistor)	0dB	–20dB	–40dB

can be assumed to be AC short circuits. Since the output impedance of U1B (a closed-loop op amp) is very nearly zero, the source impedance seen looking into R12 from switch S4 is basically just the value of R12, which is 4.32kΩ. Remember this value, which is the impedance against which the attenuator switch and its associated resistors must work.

Now, when S3 is switched to the –20dB position, the source impedance R12 becomes the top leg in a new voltage divider, with the parallel combination of R19 and R20 the bottom leg. The gain is:

$$(R20 \| R19)/((R20 \| R19) + R12) = (475 \| 1M)/((475 \| 1M) + 4.32k) = 0.09902$$

where $(R20 \| R19) = (R20 \times R19)/(R20 + R19)$

This value is –20.08dB (relative to the level at U1B Pin 7), which, when taken relative to the situation for S3 in the 0dB position, results in an attenuation difference of (–20.08dB) – (–0.04dB) = –20.04dB. This is quite adequately close to the desired –20dB value, and agrees pretty well with Dr. Perper's value (although he figures 0.0991 by justifiably ignoring R19, but then rounds it to 0.10 and determines the result is "more accurately –19.9dB," although his original unrounded 0.0991 is actually an apparently less accurate –20.07dB). For the –40dB position, the calculated result is –40.04dB.

But then, when analyzing the circuit with S4 in the PROBE position (and, I assume, with S3 in the –20dB position), he neglects to consider either a reference gain or the effect of resistors R26 and R20. A correct analysis could begin by finding the 0dB (S3 in the open position) gain for the probe input, which is really all he has calculated in his entire PROBE position analysis. He has found the attenuation level between the probe tip connection and the J2 connection of R14, which, if the slight attenuation and loading effects of R26 and R19 are neglected, gives a reference gain of –27.04dB for this part of the circuit. Including the effects of all resistors, we obtain a reference gain from probe tip to U1C Pin 10 of 0.0443, or –27.08dB, which is nearly the same.

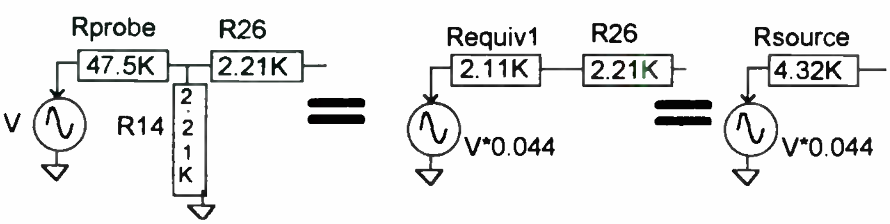


FIGURE 1: M/PP graphic equation for Rsource.

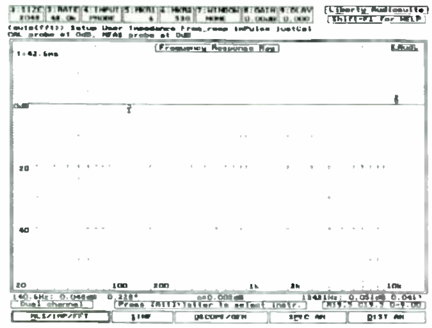


FIGURE 2: 0dB measurement.

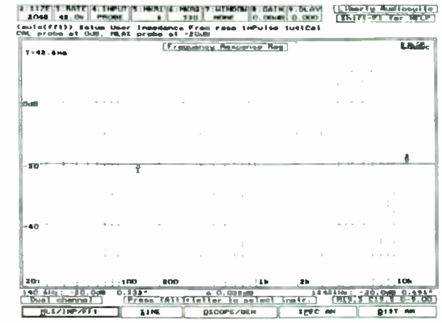


FIGURE 3: –20dB measurement.

To calculate the attenuation steps, you must now determine how the gain changes when the attenuator is switched to its –20dB and –40dB positions. This can easily be done by calculating the impedance from switch S4 looking into the probe network (formed by the 47.5kΩ probe resistor with R26 and R14), much as was done when looking into the mike amp circuitry (U1B through R12) when analyzing the MIC setting of S4. The probe is presumably measuring a voltage signal of interest, so it can be considered at a zero impedance point, per standard Thevenin equivalent analysis.

So the 47.5k probe resistor can be taken in parallel with R14 for impedance analysis purposes, which gives an equivalent resistance for these two parts of

Requiv1 = (RprobeIIR14) = (47.5kII2.21k) = 2.11k

This equivalent is then presented to S4 in series with R26, which results in an impedance of

Rsource = Requiv1 + R26 = 2.11k + 2.21k = 4.32k
(Fig. 1)

Note the eerie similarity of this result with the value of R12. This means that the gain will change with the S3 positions just the same amount when fed from R14, R26 and the probe resistor (with S4 to PROBE) as it does from U1B via R12 (with S4 to MIC). Since the gain steps are correct for the MIC input, they are also correct for the PROBE input, as anyone who completes the calculations will find. Also note that I didn't say the gain will be the same, but that it will change the same amount when the switches are moved.

In both cases additional circuitry contributes to total gain. In actuality, the MIC circuitry operates at a much higher total gain (54.04dB, assuming dead-on resistors and at the 0dB relative gain setting) than does the probe (–0.03dB, at the 0db setting, and using the 47.5k probe resistor). After all, a mike preamp with a gain of 0dB wouldn't be very useful, would it? The total input-to-output voltage gains of the Mike/Probe Preamp for the various inputs and at all three gain settings are tabulated in Table 1.

Dr. Perper's concern about the CAL path not matching the MIC path misses the fact that mike preamps usually need gain, so of course the paths will be different. Measurement condenser microphones measure sound pressure; the CAL probe measures a voltage. A sound pressure cannot equal a voltage, so there is no reason for a mike input gain to match a probe input gain.

The various parameters of the mike preamp stage, the microphone sensitivity, acquisition card gain, and attenuator settings must be considered and calculated within the measuring system software, as with Liberty Audiosuite. Note also that the circuitry of the MEAS probe input matches that of the CAL input when S4 is in the PROBE position, so the probes, both of which measure voltage, can be used correctly in normalization. In short, there is nothing wrong with the attenuator design of the Mike/Probe Preamp.

Figures 2–4 *show a* straight through *measurement made with Liberty Audiosuite, with both probes feeding a Mike/Probe Preamp connected directly (through their 47.5k resistors) to the soundcard output. The MEAS channel is normalized by the CAL channel (in other words, the difference, in dBs, between the two channels is shown). The CAL gain switch is set to 0dB for all plots, and the MEAS gain switch is 0dB for* Fig. 2, *–20dB for* Fig. 3, *and –40dB for* Fig. 4.

Figure 2 *shows, as it should, a line at 0dB because both probes sense the same signal and their circuits have equal gains.* Figure 3 *is a line at –20dB because the –20dB switch setting of S3 is properly doing its job per the analysis.* Figure 4 *shows all is well with the –40dB position also.*

The use of 1% resistors is not merely "show-offish" (notice I didn't use them at R30, R3, R6, R19, R8, or R16 where the precision is unwarranted). On one hand, Dr. Perper takes me to task for using precision resistors, and then complains they aren't precise enough. It reminds me of a person who

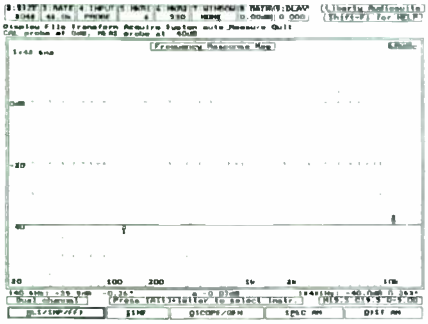


FIGURE 4: −40dB measurement.

complained that a restaurant's food "tasted bad...and there wasn't enough of it."

Perhaps it was the inability to reconcile the supposed 7dB error with my use of 1% resistors that prompted sarcasm. Nonetheless, the precision achieved by the Mike/Probe Preamp is very good for audio measuring equipment (and extremely good at this price). A worst case −1% error in R12 and a +1% error in R21, for instance, will result in an attenuation variation for the −40dB position of only −0.17dB from that using nominal values. To what equipment could he possibly be comparing the M/PP?

Typical 1% metal film resistors are 100ppm maximum per C°. This means that with over a 20°C temperature change due to dissipation and ambient conditions (very unlikely as the circuit is flea-power and will be used almost exclusively at room temperature), the resistor values might change a whopping 0.2% and probably all in the same direction, with little or no effect. Also, the TLC2274 is a quad differential amplifier with CMOS FET inputs; what is it about a JFET amp that is supposed to be more precise? The circuit is AC coupled, so DC offsets are essentially irrelevant. The TLC2274 is a nearly ideal choice.

Dr. Perper's second point has me somewhat baffled. I don't see why he would wish to plug a probe into the microphone input J1. The MIC input J1 is intended for microphones, not for probes, which should connect to J2 and J3, as labeled on the schematic and cover plate. R10 is fed from the 2.5V reference and not returned to ground because it supplies phantom power to the electret microphone capsules (used in Mitey Mike, IMP, or the Girardin microphones). Keep R10 connected as shown, or your microphone input will not work with these capsules (delete it if you have a self-powered microphone and desire a high impedance input).

I have little argument with Dr. Perper's third point. Ground should indeed be shown in the schematic at the inputs and outputs, but this has not caused any real confusion. Most builders are used to the idea that inputs, unless of the balanced type (which these clearly aren't), are almost always referenced to ground. That they are in this case can be easily deduced from the printed circuit pattern. But the schematic is indeed incomplete in this regard; my apologies for this oversight.

But other than agreeing to the need for these few missing triangles, I stand by the M/PP design and the published schematic entirely.

TO KIT OR NOT

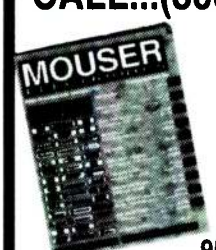
Thank you for the kind words in your "Kit Kick" editorial (*SB* 3/95, p. 5). The Speaker Works, Inc., has offered a variety of loudspeaker kits for the past 15 years and has advertised them in *Speaker Builder* for the last three or four. To our dismay, the kit business "just ain't what we feel it outta be." In a letter, one of your readers stated he held the recipe-book approach to speaker building in some disdain. To that I say different strokes for different speaker hobbyists. Not all of us can afford to fill our basements with mistakes or drivers we can't make work.

The Speaker Works approach to kits is to develop speakers to cover price points on our retail sales floor. We design the very best product we can within the cost constraints required to meet these needs. The primary criterion is to offer a better product for the price than the so-called name brands. Our retail customers and retail sales of finished loudspeakers indicate that we are successful in fulfilling our goal. We then select a portion of our line to offer as kits, with or without cabinets. Our kit prices are well below finished product price, since we don't need to spend hours in a demo room to make the sale.

Our kits offer an excellent value and, when completed, a very good loudspeaker. After assembling the kit, the builder has speakers that work quite well and may choose to stop at that. Or, the builder, with the funds and inclination, may decide to improve upon our kits. We know most of the tricks to pull out those tiny nuances that, if pursued, add to the listening pleasure. Those tweaking experiences can be fun, and a worthwhile learning experience. We are always pleased to work with builders of our kits to effect these upgrades.

Thank you again for the editorial, the magazine's words of wisdom, and the forum for gripes and ideas.

Carl Roberts
President
The Speaker Works, Inc.
Phoenix, AZ 85016



Ask SB

By Joe D'Appolito
Contributing Editor

AUDIO EXAM

In unpublished correspondence with Joe D'Appolito, he asserts that two identical, side-by-side drivers exhibit a system efficiency twice that of their individual efficiencies. This was also observed in a paper Mr. D'Appolito referenced, E.M. Long's "Design Parameters of a Dual Woofer Loudspeaker System." The efficiency gain appears to result from an unanticipated increase of the radiation resistance due to the effect of mutual coupling. The gain occurs for closely mounted drivers over a limited frequency range.

I do not believe this behavior can be inferred from a consideration of the individual drivers, whether using Thiele/Small or more fundamental acoustical models. Two transducers will not ordinarily exhibit twice the efficiency of either, so I must reject Mr. D'Appolito's conclusions to the contrary. As for his example of how to sum SPLs for the correct acoustic sum, I believe he errs here also. If it is a given that the drivers are connected out of phase, then one driver's SPL squared has the opposite sign of the other's, and so the expected zero still results.

On pages 5 and 6 of Vanderkooy and Lipshitz's paper entitled "Power Response of Loudspeakers with Noncoincident Drivers," Equations 8 and 10 show SPL summations for power in which the individual SPLs are first squared and then summed (i.e., notably *not* summed first and then squared). If two drivers are in separate rooms, how can the sum of their acoustic output be otherwise than proportional to the sum of their SPLs squared?

Near the bottom of page 5, they present the case of two identical in-phase sources and show that, at low frequency or small separation, the power response is four times the monopole result. This doesn't surprise me, because Long's paper provides experimental evidence of this phenomenon and offers a physical explanation based on an increase of radiation resistance.

What does surprise me is Vanderkooy and Lipshitz's failure to recognize that their calculations, which imply an extraordinary increase of efficiency, call for a physical—not merely geometrical—basis of explaining this phenomenon. If the geometrical analyses show such an increase (as they do), then something is wrong. The model has no provision for explaining an increase in radiation resistance, which, as I understand it, is the sole reason why closely mounted drivers can exhibit an efficiency increase.

In "An Acoustical and Electrical Interaction in Multidriver Arrays" (JAES, Vol. 31, No. 10, 1983), Greiner and Allie sum for power by first adding the volume velocity of each driver, squaring this sum, and then multiplying by the radiation resistance. By summing in this manner, they are seemingly unaware of the implication that two drivers will output four times the power of one, four will output 16 times the power of one, and all without an increase of radiation resistance! Such a fact, if true, seems to me more important than the conclusions of their paper.

I feel foolish in opposing acoustic luminaries such as Joe D'Appolito and the others, but I think I have raised a valid issue and would be grateful to anyone who would reconcile my thoughts with the truth. When I was in school, I would get 100% if I applied the correct method and calculated the correct answer, 70% if I used the correct method but got the wrong answer because of an arithmetical error, and 0% if I got the correct answer but used the wrong method. On this question, I think Joe D'Appolito got a zero.

David Meraner
Scotia, NY 12302

Joe D'Appolito responds:

As I stated in my [unpublished] letter, I chose not to answer your questions in terms of radiation impedance, but took it as an opportunity to clearly discuss the difference between power efficiency and pressure response sensitivity in the common framework of

Thiele/Small theory. I thought the T/S approach would be more widely appreciated by SB readers. Note that pressure response is not a function of radiation impedance.

Regarding Mr. Long's paper, I believe he (rather than I) arrives at the right answer through the wrong method. His approach has two problems. First, all calculations of radiation impedance he cites are done under the assumption of two closely spaced pistons on an infinite plane, where radiation resistance increases as the square of the area for low frequencies. In a typical listening room this is not the case. Allison and Berkovitz (1970) show that the radiation resistance of a driver at low frequencies in a typical listening room environment is independent of driver area. Second, Long's approach leads to an equivalent driver which is inconsistent with T/S theory. A better alternative is the use of volume velocity.

The acoustic power radiated by a loudspeaker system is proportional to the system's net volume velocity:

$$P_A = R_{AR}\,|U|^2 \qquad (1)$$

where: P_A = *radiated acoustic power,*
R_{AR} = *diaphragm radiation resistance,*
U = *net volume velocity.*

Volume velocity is simply the linear velocity of the cone (v) times the cone area (S_D):

$$U = vS_D \qquad (2)$$

Using the units of (meters)^3/sec, it is the volume of air moved by the diaphragm, and is analogous to current in an electrical circuit. Absolute value bars are placed around U in the first equation, since U is a vector (or phasor) with both magnitude and phase. According to Long, a broad frequency range exists where each driver's radiation resistance roughly doubles. Since there are two drivers, the total radiation resistance is four times larger. So:

$$P_A = 4R_{AR}\,|U|^2 \qquad (3)$$

As Equation 3 indicates, we now have an equivalent driver with four times the radiation resistance of a single driver. But, if the radiated power is to increase by a factor of only 4, the volume velocity must be the same in Equation 3 as in Equation 1. What does the equivalent single driver look like, and does it jibe with T/S theory? Your first examination question worth 40% of your grade is to determine the equivalent driver corresponding to Equation 3, and also determine whether this driver is consistent with Thiele/Small theory and why. (Hint: it isn't.)

An alternative approach: since we have two drivers, the volume velocity is twice that of a single driver (at least at low frequencies, where the two velocities are in phase). Because both volume velocities are at the same frequency, we must add them before squaring (more on this later). If we further assume that the radiation resistance of each driver is unchanged, the total radiated acoustic power is then:

$$P_A = R_{AR}\,|U_1 + U_2|^2 \qquad (4)$$

At low frequencies the Us are equal and in phase, so:

$$P_A = R_{AR}\,|2U|^2 \qquad (5)$$

or

$$P_A = 4R_{AR}\,|U|^2 \qquad (5a)$$

which is the same answer we got before, but by different reasoning.

If we choose an equivalent driver with twice the cone area of the individual drivers, *we will have twice the volume velocity with no increase in radiation resistance (according to Allison and Berkovitz), and thus four times the acoustic power. The volume velocity approach leads to the correct answer and is also consistent with T/S theory. Your second examination question, also worth 40% of your grade, is to determine whether this equivalent driver is consistent with T/S theory and why. (Hint: it is.)*

Let's turn now to the problem of computing power in electrical circuits. Suppose we connect two sine wave generators numbered 1 and 2 in series, and apply the resulting voltage to a resistor, R. The total instantaneous power dissipated in R is:

$$P = (E_1 + E_2)^2/R \qquad (6)$$

where E_1 and E_2 are instantaneous voltages.

Remember that the Es have both magnitude and phase. Expand Equation 6 to get:

$$P = (E_1^2 + E_2^2 + 2E_1E_2)/R \qquad (7)$$

Equation 7 is still instantaneous power. We usually talk about average power and use RMS voltages to compute it. Thus, we must average Equation 7 over time, and here is where you become confused. If E_1 and E_2 are of different frequencies, the cross-product term $2E_1E_2$ in Equation 7 averages to zero, and you arrive at:

$$P_{AVG} = (\underline{E}_1^2 + \underline{E}_2^2)/R = \underline{E}_1^2/R + \underline{E}_2^2/R \qquad (8)$$

The underlines in Equation 8 denote RMS values. We always consider pressures or volume velocities (which are analogous to voltages or currents) that are at the same frequency! OK, suppose $E_1 = E_2 = E$, all having the same frequency, then:

$$P_{AVG} = (\underline{E}^2 + \underline{E}^2 + 2\,\underline{E}^2)/R = 4\underline{E}^2/R \qquad (9)$$

On the other hand, if $E_1 = -E_2$:

$$P_{AVG} = (\underline{E}^2 + \underline{E}^2 - 2\,\underline{E}^2)/R = 0 \qquad (10)$$

Squaring a negative quantity results in a positive number. The cross-product term produces the negative sign and leads to zero net power. Results from Equations 9 and 10 can be obtained by simply adding the two voltages (or currents, pressures, or volume velocities) and then squaring the result. In general, there will be a phase angle between the two voltages which must be taken into account. To repeat, Equation 8 is valid if—and only if—E_1 and E_2 are of different frequencies. Your last examination question worth 20% of your final score is to prove that the cross-product term in Equation 7 averages to zero over time when E_1 and E_2 are of different frequencies. ▶

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Belles DMM stereo preamp, $175; B&K ST-140 power amp, $250; Sumo "Charlie" FM tuner, $150. All mint, with boxes, manuals, etc. Craig Edwards, (407) 984-8640 (Melbourne, FL).

Six Patchbays Switchcraft/ADC ¼" mil type TRS, half-normalled, 52 points each, for studio or audio lab, $69 each or $375 for all; Roland PC sound card, RAP-10, and latest Roland Audio Toolwords (V3.7 for Windows) digital editing software, superior general MIDI sounds as well, $129. Robert, (310) 272-8300, message.

Three Dynaco MK-III amps, two work and are complete, one needs tubes, cap, bottom cover, transformers OK. All have cages, $300. Al, (801) 259-5306.

JBL 2420 horn drivers, 1" throat, Alnico; 3182 500Hz crossovers; 2345 horns; 3110 800Hz crossovers; Altec 802 drivers; 808 horn drivers; 806 horn drivers; 32C horns; Community "zoid" horns for 288 drivers; JBL 2220J woofers; K140 woofers; several mono rackmount integrated tube amps. David, (914) 688-5024.

Four decades record collection, selling all/part—call/write for list of classical/opera; power transformers, get 35V @ 5.5A underrated, $25 each; pair Peerless cc line 811647 8Ω 1" dome tweeters, $40; B&W MR100 4" mid, $25. Jerry Feldstein, PO Box 580163, Bronx, NY 10458, (718) 364-3485, 7-10 p.m. EST, no machine.

Pair KEF B139, SP1044, $150; pair Vifa CM75MX4, $30; pair Polydax HD100D25HR, $20; pair Polydax TPX 1125JSN2CN9, $35; pair Polydax HD20B25H4C12, $35. One pair each. Ron, (912) 925-2311.

Driver pairs: Focal 8K415S, new, $135; T120K, new, $80; Dynaudio 17M75EXT (one needs to be reconed), $25; Audax MHD17B37R, dust caps removed, $40; three-way transmission line cabinets, oak with cocobolo trim, grilles with tile accent (pictures available), $125; Dynaco ST-70, modified, with extra tubes, $200; pair Marchand XM1 with PS10 supply in cabinet, $40; crossover (parts) 17M75EXT to T120K, $15; Kelvin PRO 400 digital multimeter with frequency counter, $40; *Speaker Builder*, 1986 to present, make offer. Steve Hawkins, 1625 SE Nehalam #11, Portland, OR 97202, (503) 234-3608.

Pair Audax HM130CO, $100; pair Audax AWO255I, $60; pair Synchron SYN-519A, $70; pair Woodstyle WS602, $60; pair Woodstyle WS123, $125; pair Peerless 831759, $110; pair Madisound 8252R, $50; Superphon SM-220 amp, $350; Grusin Sheffield, $50; pair Fisher dome midrange, $10. Rick, (919) 662-5253, 7 to 11:30 p.m. (EST).

Altec 811 horns, $125 each; 806A horn drivers, $110 each. Both mint condition. Lance, (206) 744-1252.

Audio Research D-100 amp, needs work, $400; H-P 120B oscilloscope, good condition, $60; Kenwood 7000 tuner, $150; Sony 2251LA turntable, $150; VMPS original subwoofer, $195; Heath IG5256, $75; Eico 369, $50; AR XA, $50; two Viking RP61 tape amps, $75; 10' transmission line cabinet for 18" driver, $300. Tom, (415) 851-2779.

Marchand PM2 50W amp module, $40; Stax SR44 headphones, $75; new Shure M99E cartridge, $15; IDE caching hard drive controller with 1M-byte DRAM, $45; 512K 16-bit ISA video card, $20. Dan Patten, 1768 N. 980 W., Orem, UT 84057, work (801) 224-8080 or home (801) 225-8577.

Pair Audax PR170X0 120dB with mids, new '95, $125 or best offer; pair JBL 2425J 1" drivers, excellent, $250; Threshold T-100, box & manual, excellent, $1,875; Spectral DMA-50, box & manual, excellent, $1,000; Stax Lambda headphones & SR-7 transformer, $250. Steve, (203) 397-3888, 7-10 p.m., machine 4th ring.

Four pairs Gold Ribbon planar magnetic drivers, excellent, $650/pair or best offer; pair dipole wing enclosures for same, free. You pay packaging and shipping. Joe, (609) 397-8315 (NJ, EST).

Nakamichi 580M tape deck with shop manual, needs work or good for parts, best offer; Carver C-4000 Sonic Holography preamp with manuals, good condition, $35. Scott, (815) 965-9180.

Krell PAM-7 preamp, mint, $1,400; Forte 6 power amp, $1,100; Monarchy 22 DAC, $600. Paul, (303) 438-4164, anytime, please leave message.

Dahlquist DQ-LPI electronic crossover, 18dB active low pass, 6dB passive high pass, with boxes and manual photocopy, walnut end panels, excellent condition, $150 + shipping, firm. Paul, work (619) 230-5548, home (619) 632-7463.

Music Reference RM-5 MK-II preamp, custom solid cherry case, wonderful sound, audiophile "A" rated, brand new in box, cost $1,450, will sell for $1,000, what a bargain! Kurt, (805) 961-7590 or 964-9789 eves (CA).

Pair AR-2ax three-way speakers, oiled walnut, six Bolt Alnico woofers, original condition, excellent, $250/pair + shipping; University classic horn components: pair C15W, pair Cobra Flex/T-30, pair HF-206/T60, N-3A crossover, $700/set + shipping; pair University C15HC high-compliance woofers, $400; Jensen PR-4000A, $275; Dynaco ST-70, PAS-3X, FM-3, $375/set. Tom Tyson, (910) 885-9436 (Highpoint, NC).

E-V Georgians, all parts for building a pair of classic E-V K-horns with copies of construction plans and manual, $1,100; Sunn SPL915R, $50; egg crate foam: 34" × 72" × 2", $5/each; pair Altec 511B, rough, $60. Dave, work (216) 922-4816 or home (216) 666-6805 (OH).

Pair 10" x 11" x 15" Grenier veneer cabinets with grilles, 6" woofers, ¾" tweeters, crossover, posts, cabinets cut for drivers, needs assembly, all new, $178 investment, yours for $85 plus 50% of UPS charge. Alan, (704) 333-8436, Mon.-Fri., 8-5 (EST). All calls returned if I'm out.

JBL: 3110 crossovers; 3182 high-power crossovers; 2220J 15" woofers; 2345 horns; Dukane (by JBL) Alnico horn drivers; Altec: N801-8A crossovers; 811 horns; 808-8B drivers; 802D drivers; 806A drivers; two Carver 1201 and one PN1.5 professional 600W/channel stereo power amps. David, (914) 688-5024.

Two outstanding three-way speaker kits with Focal woofers, Eton mids, Esotar T330 tweeters, crossovers, instructions, $480; high-quality Bozak midrange & Dynaudio tweeters, used once in test. Bob, days (310) 331-7700, nights (310) 676-4629.

Norsonic Type 1230 pressure mike capsule (equal to B&K Type 4166), $350; White 4400 1/3-octave EQ, $350; White plug-in octal noise source, $50; TEF BAL-1 for TEF-20 analyzer, $100. All mint. Neil, (703) 533-0717.

Mitey Mike, calibrated by Joe D'Appolito, complete with response curve, $135 or best offer; Hewlett-Packard model 403B wide range AC millivolt, ideal for use with Mitey Mike, $95. Paul, (513) 436-3861, FAX (513) 222-4644.

WANTED

Tube hi-fi, theater amps/preamps and speakers: Altec, Jensen, McIntosh, Marantz, Craftsman, Dynaco, Heath, Eico, etc. ***SUNSHINE SOUNDS***, (405) 737-3312.

Philips 544 speaker or AB8067/mfb4 8" woofer in good condition; Linn Ittok. Tom, (415) 851-2779.

Diagrams, schematics, and manual for Heathkit "Transistor Stereo Amp" model number AA-21D or you may know of mods or improvements that can be made. Lee Meador, Box 3261, Arlington, TX 75007, mornings (817) 467-4964, evenings (214) 235-1474.

To Mr. Rowley (from Alaska): I lost your address & can't send the promised info. Please contact me & I'll send it right away. Marchand BASSIS EQ; dbx 3BX dynamic range expander; Audio Source AMP-1 amp. Angel Rivera, 7804-22nd Ave., West Bradenton, FL 34209-5208, (813) 792-3870.

Urei Model 811A studio monitors with model 829 crossover networks, must be perfect! Robert, (310) 272-8300, message.

150-200W/channel amp. Bob, days (310) 331-7700, nights (310) 676-4629.

JBL 375 drivers, working or not; 537-509 horn lens. Rhett, (517) 487-6691 before 4:00 p.m. EST or leave message.

KLH model 4 bookshelf speakers. Gary, (415) 324-9225.

Pair JBL 2245H; pair Focal 15VX. Good condition. I pay shipping and other costs. C.J. Nam, South Korea, (011) 82-333-52-2257, FAX (011) 82-331-284-0340.

Pair Jordan modules. Larry, (217) 732-6486.

Linn tonearm, any model, must be in mint condition. Jim, (916) 265-2147, 9:00 a.m.-9:00 p.m. (EDT), leave message if I'm not in.

Ad Index

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If you are interested in joining a club or forming one in your area, contact the following:

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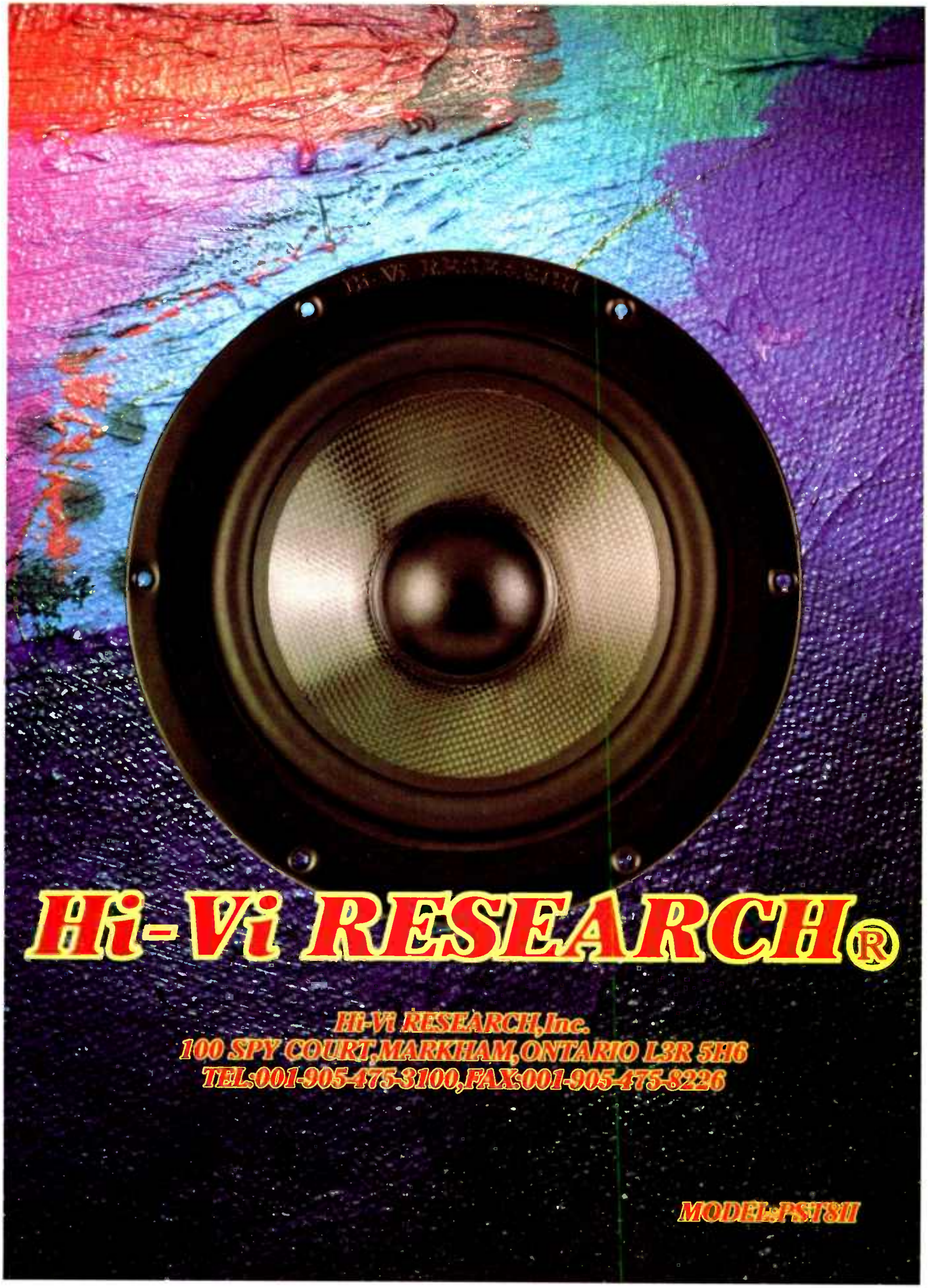

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Parts Express is dedicated to bringing serious speaker builders the finest products currently available. We have recently become a distributor for Linear X and now carry their superb loudspeaker design program, LEAP Version 4.5, plus the outstanding LMS loudspeaker measurement system. We are also very pleased to announce that Parts Express is now an authorized distributor for Morel Acoustics. Morel's unique vented magnet system and large diameter Hexatech voice coils provide smooth frequency response, high power handling, and astonishingly low levels of distortion. We also carry the high performance line of drivers from Audax, featuring the revolutionary High Definition Aerogel and Carbon Fiber cone materials. For high quality crossover networks, we stock Solen polypropylene capacitors, 14 gauge air core inductors from ERSE, Solo copper foil air core inductors, and Monster Cable speaker wire. Our software programs for enclosure design include BassBox 5.1, Quick Box from Sitting Duck Software, and Loudspeaker Architect 2.01 from N.A.M.E. Call now for your copy of our FREE 212 page catalog and to get the details on these and many other quality products.

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