CLIO: A NEW SPEAKER TEST SYSTEM

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Speaker Builder

THE LOUDSPEAKER JOURNAL

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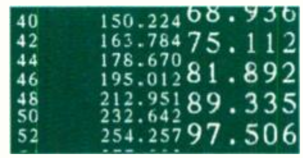
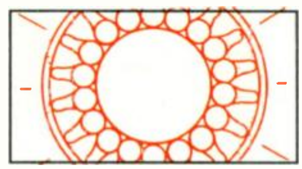


THE FREELINE: A TINY TL



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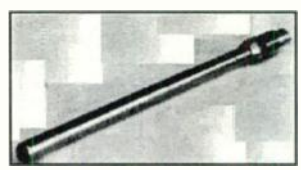
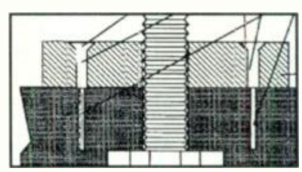
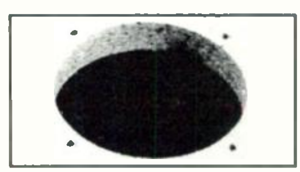


SPEAKER STANDS YOU CAN MAKE



MAKING ODD-BALL DRIVER CUTOUTS



Nuggets from Madisound's BBS

Good News

■ CIRCUIT SIMULATION

Electronics Workbench® version 4.0 offers true mixed-mode simulation of analog, digital, and mixed-mode electronics circuits. Users can design and verify circuit configurations through an intuitive interface which supports drag-and-drop techniques for adding parts, running wires, and hooking up simulated test instruments. More components are available in multiple parts bins, and over 350 real-world analog and digital models allow creation of complex circuits. Interactive Image Technologies, (800) 263-5552, (416) 977-5550, FAX (416) 977-1818, ewb@interactiv.com.

Reader Service #107

CENTER-CHANNEL SPEAKER

The Omnisurround CC66 combines direct and Dolby surround sound. Its patented binaural voice coil technology spreads the soundfield, while the sound source remains fixed. The amount of ambient sound entering the sound stream is controlled by an ambisonic processor located in the speaker enclosure. The binaural voice coil is then fed primary and ambisonic surround signals. The amount of ambience can be adjusted with a center-channel control. Audio-Video Technologies, 60 E. Ida St., Antioch, IL 60002, (708) 395-6321.

Reader Service #110

■ DEEP BASS SUBWOOFER

The V-75 powered subwoofer features a Headroom Maximizer II™ circuit, an 80Hz speaker-level, high-pass filter, a high-performance 12″ driver, and a two-position phase control switch. The internal amplifier is rated at 75W RMS. A "filter" control allows the user to set the upper rolloff to any frequency between 50 and 125Hz. Miller & Kreisel Sound Corp., 10391 Jefferson Blvd., Culver City, CA 90232, (310) 204-2854, FAX (310) 202-8782.

Reader Service #111

BIOCOMPOSITE LOUDSPEAKER

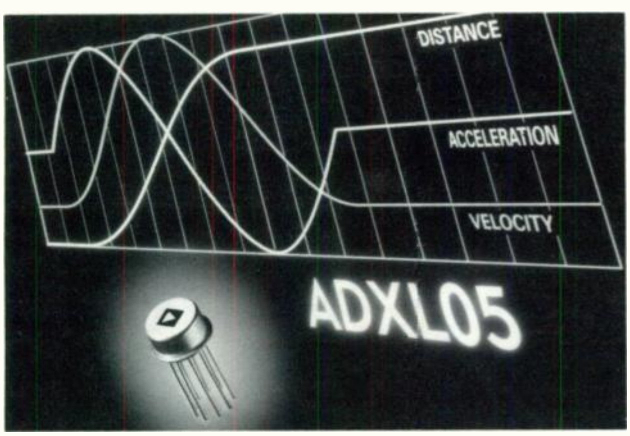
Vision loudspeakers are constructed from a biocomposite material having twice the density of oak. It has a crystalline granite appearance, and is available in several color schemes. Besides being environmentally friendly, the new material eliminates virtually all cabinet vibrations. Additional enhancements include over-sized, gold-plated, five-way binding posts, floor spikes, heavy internal wire, and an acrylic semigloss finish. Curtis Audio, PO Box 61, Lawrence, KS 66044, (913) 865-9991.

Reader Service #109

■ SPECTRAL ANALYSIS SOFTWARE

Spectra Plus software interfaces with any Windows-compatible sound card to provide real-time spectral analysis, as well as recording, playback, and post-processing capabilities. Frequency response, distortion, and transfer function measurements are also available. The program, which supports FFT sizes through 16,384, can display, export, and print the time series, spectrum, phase, 3-D surface (waterfall) plot, and 256 color/grayscale spectrogram. An audio signal generator utility produces pink/white noise, swept sine, tones, and pulses. A math coprocessor is required. Pioneer Hill Software, (800) 401-3472.

Reader Service #114



LOW-G ACCELEROMETER

The ADXL05 single-chip, integrated, low-g accelerometer resolves minute changes in acceleration (from 0 to ±5g full-scale) with 0.005g resolution. The device combines a micromachined sensor, modulator, demodulator, voltage reference, signal conditioning, amplification, and on-command self-test circuitry on one polysilicon die. The ADXL05 was designed to simplify system complexity, improve performance, and reduce engineering costs in applications where tilt, inertial, shock, vibration, velocity, and acceleration characteristics must be measured. Analog Devices, Inc., 181 Ballardvale St., Wilmington, MA 01887, (617) 937-1428, FAX (617) 821-4273.

Reader Service #112

■ THX UPGRADE

The AV81HT amplifier has been upgraded to THX® and will sell under the AMC brand. The AV81HT-THX incorporates a module with THX-approved circuitry, which used in conjunction with Dolby Pro Logic decoding reequalizes movie soundtracks. The surround-channel information is also decorrelated to minimize the listener's need to be in only one room location. The circuit's timbre matching compensates for natural hearing characteristics by maintaining spectral balance as sound moves from front to rear. Weltronics Corp., PO Box 80584, San Marino, CA 91108, (818) 799-6396, FAX (818) 799-6541.

Reader Service #115

■ MODEL FITTING PROGRAM

Scientist® version 2.0 for Windows fits model equations to experimental data. The program incorporates elements of matrix operations, symbolic manipulation, graphics, and engineering calculations into a worksheet. New features include enhanced equation editing capabilities, graphics interaction via "tabbed" dialog boxes, improved text editor, expanded documentation with sample calculations (including a 500-page manual), and refined error trapping and numerical routines. MicroMath Scientific Software, PO Box 21550, Salt Lake City, UT 84121-0550, (800) 942-MATH, (801) 943-0290, FAX (801) 943-0299, MicroMath@Delphi.com.

Reader Service #108

■ ONE FOR ALL

The Selective Teletransfer Unit connects to single-line telephones (landline) and answering machines. It selectively transfers callers to a cellphone, pager, voice mail, any other number, or an answering machine. The level and ability to transfer can be made from any location by calling the device and entering the proper functions. Cellular and pager phone numbers are programmed into the unit, so they remain secret. Electronic Design Specialists, Inc., 4647 Appalachian St., Boca Raton, FL 33428, (800) 813-1323, (407) 487-6103.

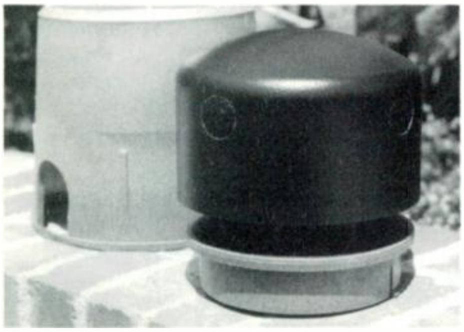
Reader Service #113

← *Reader Service #36*

Good News

GARDEN MINI-SPEAKER

Waterworks Acoustics announces the Soundscaper™, an all-weather speaker with ground-level mounting that is designed to disappear in any landscape plan. The tough plastic enclosure is sealed against the elements and houses a woofer that fires down, eliminating



trapped moisture and debris. The speaker enclosure is removable from its buried base for easy prewire and installation. Sound coloration is eliminated by positioning four tweeters so that they are not obstructed by the enclosure. Frequency response is 80Hz–20kHz ±3dB. Waterworks Acoustics, 3365 Fernside Blvd., Alameda, CA 94501, (510) 522-0374, FAX (510) 769-9636.

Reader Service #102

THREE-WAY DIPOLE SPEAKER SYSTEM

The Jayde three-way, floor-standing speaker system combines dipole technology with a vented bass enclosure. It utilizes an open baffle arrangement, with fabric grille used on the cabinet's front and back so that music is radiated in both directions. Each dipole panel and subwoofer enclosure is constructed as a separate cabinet. The dipole panel snaps into the top of the subwoofer cabinet, and each unit is individually wired. Probe Audio Labs, 10223 NW 53rd St., Sunrise, FL 33351, (305) 749-7344.

Reader Service #103

SOUND SYSTEM FOR PC COMPUTERS

The Thrifty Talker™, from SET, provides a way to get high-quality digital audio from all IBM-compatible PCs. It consists of a pass-through parallel-port module, self-powered amplifier module, and stereo headset, and it is compatible with both Windows and DOS software applications that support audio output. This passive device does not interfere with other installed I/O cards such as CD-ROM, FAX/modem, or networking systems. SET, Inc., 3205 SE Spyglass Dr., Vancouver, WA 98684-3724, (360) 944-0333, FAX (360) 260-1660.

Reader Service #104

COMPACT DIGITAL MONITORS

B&W Loudspeakers has released two new compact digital monitors, the CDM1 and CDM2, designed for serious listeners who value traditional sonic virtues, yet need speakers suitable for placement on a shelf or small stand. Both models are crafted in wood veneer, with black or red ash finishes available. Both are two-way systems, using B&W's vented-box enclosure design, and their linear response extends to 30kHz. Each model is rated for use with amplification from 30–120W continuous output. B&W Loudspeakers of America, 54 Concord St., North Reading, MA 01864-2699, (800) 370-3740, (508) 664-2870, FAX (508) 664-4109.

Reader Service #101

LOW-COST DIGITAL MULTIMETER

Fieldpiece Instruments has introduced the LT6 handheld digital multimeter. This DMM measures volts (AC and DC), indicates continuity with a beep, and tests power semiconductors for catastrophic failure. It works with all the other accessories offered by Fieldpiece for its heavy-duty line of meters. Optional test leads (model ADL2) that fit the LT6 connect the accessory heads to the meter, which has both 200mV AC and 200mV DC ranges. Currently available accessory heads include converters for AC current, DC microamps, relative humidity, and dual temperature. Fieldpiece Instruments, Inc., 231 E. Imperial Hwy, Suite 250, Fullerton, CA 92635, (714) 992-1239, FAX (714) 992-6541.

Reader Service #105

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Guest Editorial

DEFINING GOOD DESIGN

What makes a good speaker? If only I had known the answer to that question six years ago when I first started building my own loudspeakers, I would have saved myself years of experimenting and box building.

As a qualified engineer, I knew I needed some test equipment to tell me what I had produced, rather than rely on just my ears, so I invested in a Radio Shack sound level meter and some CD test discs with pink noise and third-octave warble tones. It was so laborious making frequency plots on graph paper, each one taking half an hour, with only enough resolution to show whether the plot was fairly flat or not. In building a three-way speaker kit, I accidentally connected the bass unit the wrong way around, but my third-octave test system did not indicate this.

Then I bought some second-hand B&K equipment and began to make real progress with my speaker building, but I was restricted by the smallness of the room available for testing. The early reflections from floor, ceiling, and walls masked the real response of the speakers. I realized that I needed a test system that could simulate anechoic conditions. By this time, I had purchased a couple of digital meters and, with the help of the *Loudspeaker Design Cookbook*, started to measure T/S parameters.

Thank goodness the IMP came along and rescued me from all the drudgery and restrictions of the past few years. Even taking an average of five impulses and only running on a 1M 286 PC with a 30M hard disk drive, I produced frequency plots in 25s. With the upgrade to the MLS version IMP/m, significant progress with crossover design and development became possible, and I positively rushed up the learning curve when I bought a 486DX2 PC with 8M and a 420M hard drive. Now I produced frequency plots in less than a second. The test equipment was certainly not holding me up now.

I produced and optimized crossovers in no time at all. In fact, I spent most of my time just changing components. I have a collection of short wires with crocodile clips soldered at each end. I run in drivers overnight using a moderate level of 25Hz sine-wave stimulus, then after letting the driver settle for an hour, plot the Thiele/Small parameters. (For more accurate results, I use the closed box method to obtain V_{AS}, as the frequency is raised rather than lowered when using added mass.)

For reflex cabinets, the theory really does work when you use circular ports properly positioned as recommended in the *Cookbook*. On several occasions, the calculated port length and the actual one needed from measurement have been within an eighth of an inch...remarkable.

With accurate and repeatable audio measurement and analysis systems such as IMP/m and the new Liberty Audiosuite, we can all measure the results of our speaker building efforts for a relatively small financial outlay, less, in fact, than what we may have paid for our last amplifier. Now that we can measure the results of our efforts so well, what design parameters should we be aiming for to ensure the speaker we design is a good one? You will find most of this information in published form, but not all in the same place, so let me present my criteria I developed after building over 30 different designs using a range of drivers.

1. Flat, or nearly flat, frequency response, with no peaks and only small low Q dips. Some of my best-sounding speakers have a 1–2dB fall from 200Hz–20kHz.

2. A wide even dispersion of frequency plots either side of the speaker, at angles out to 90°, without the plots crossing each other.

3. Correct phase angle. When this is right, the speakers "disappear" and the imaging becomes very precise, with no hardness or harshness. This is so important and is invariably the one fault with most manufactured speakers I have tested. It is easy to measure phase with the systems mentioned before, and you can adjust the crossover components to achieve the accurate phase. By reversing the connections on the tweeter of a two-way speaker, or a three-way speaker's midrange, I can achieve nulls of 30dB, and sometimes even as high as 45dB. (A fourth-order electronic crossover which incorporates phase correction showed me how correct phase affects the sound and imaging of a loudspeaker.)

4. Choosing the right drivers. This is where some of the "art of speaker building" applies. Drivers *do* sound different, depending on cone material, cone shape, amount, type of damping, and so forth. I prefer bass drivers with curvilinear doped paper cones, a large magnet and voice coil, and a sturdy cast chassis. I like soft dome midrange drivers for ease of integration with bass and tweeter units, with large voice coils for control and power handling. At the high end, I use only soft dome tweeters.

5. Passive crossover designs that give a smooth and even rolloff on either side of the crossover point, and where possible result in as flat an impedance plot as possible with values that do not fall below the driver DCR.

Nearly all my loudspeakers are either transmission line or reflex designs. (I regard a transmission line as an extreme variant of the reflex design; its measurement characteristics are identical.) Of these I prefer the transmission line design in which the complement of the drivers allows that design to be used. From listening tests, and for ease of crossover design, I prefer multiple 6.5" bass drivers to single, larger units.

Tony Seaford
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Speaker Builder

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

— John Stuart Mill

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About This Issue

In this issue, Contributing Editor John Cockroft shares his latest design ("The Freeline," p. 8). At first, his transmission line configuration, with both ends open, may appear a bit unusual, but it works, reducing pressure variations and spreading out any resonances inherent in the line. This simple construction project results in a speaker of smooth, detailed, and accurate sound.

Horn design has always been a complex endeavor. But in this age of computers, it needn't be, as A.L. Senson demonstrates with his program ("Computer-Aided Bass Horn Design," p. 18). This software eliminates some of the tedious calculations, such as throat and mouth areas and back-chamber volume, as well as enclosure dimensions.

If you're using a sound card for speaker measurements, you should protect it with Bill Waslo's latest device ("A Mike/Probe Preamp for Sound-Card Measurements," p. 20). He shows you how to build your own to turn your sound-card/software combo into a high-powered, practical testing system.

Now you can build your own customized speaker stands with Duncan MacArthur's precise instructions ("Four-Poster Speaker Stands," p. 24). His solid, attractive units will enhance the appearance, and function, of any listening room.

Electronic networks and BBSs represent a great untapped wealth of information for speaker builders. In "Mining for Gold on the Madisound BBS" (p. 32), William Hoffman uncovers a motherlode of useful software, and recommends some of the best programs available.

In an imperfect world, Robert Wayland shows us how to achieve near perfection in accommodating irregularly shaped mountings for our speaker projects ("Wayland's Wood World," p. 40).

Plus, be sure to check out the extensive review of the CLIO measurement system, an easy, inexpensive way to test loudspeakers and room placement ("Software Review," p. 44).

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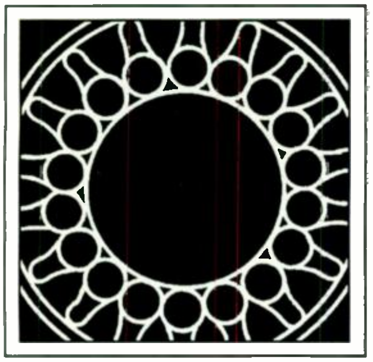

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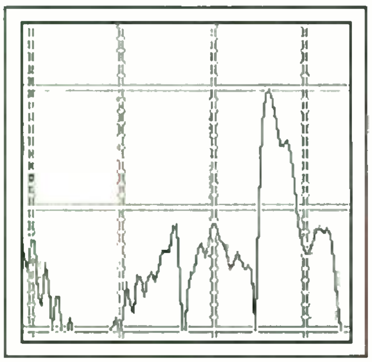

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DEPARTMENTS

THE FREELINE: AN OPEN-PIPE TRANSMISSION LINE

By John Cockroft
Contributing Editor

While suspended in a parachute harness from a 25-ton crane, 40′ up in a 60′ deep pit, with a section of a liquid argon transfer line hooked between my legs, I came to the conclusion that if removing one end of a transmission line (TL) enclosure resulted in a sonic improvement, maybe removing the other end would also help. (Good luck to the reader attempting to correlate relevancy between my pondering and my pendulous situation.) With both ends of the enclosure open, the pressure pulses from the cone movement would appear to be minimized.

Enclosure vibrations are excited by pressure variations caused by a functioning enclosed speaker cone. A considerable reduction of enclosure pressure is effected by removing one end of the enclosure, such as happens when a TL enclosure is used or, perhaps to a lesser extent, through the use of an aperiodic, or pressure-relief type of enclosure. These reduced panel vibrations lower competition from extraneous voices and allow the speaker to present a more natural, uncolored sound.

INSPIRATION

For both ends to be open, you need to position the speaker elsewhere from the traditional TL placement at one end of the enclosure. I placed the woofer center one-third the distance from one end of the line. With the speaker asymmetrically located in relation to the line ends, I believed that frequency-induced pressure irregularities might be spread and thus minimized. I don't know whether speaker placement is a contributing factor. It does, however, allow you to place the speakers at a reasonable distance from the floor.

Smoothness of sound is an attribute of this Freeline design, which is a hybrid TL loudspeaker system of unusual configuration, in that both ends of the transmission line are open to the atmosphere. The bass driver is centered at one-third the line length from the upper end of the TL.

PERSPIRATION

A couple of weeks after being lowered from the crane, I constructed the Freeline. The design attempts to: (a) reduce pressure variations within the line; (b) spread out any resonances that might be inherent in the TL, by providing dissimilar exit lengths from the rear of the driver to the atmosphere; and (c) evaluate possible hybrid TL configurations.

My prototype structure is a small system designed for apartments and other dwellings with small to moderate room size. The concept, I believe, could be used for larger systems, in rooms of larger volume, just as with other types of loudspeaker systems.

DIMENSIONS/SPECS

The internal dimensions are 3′ long × 5″ × 5¼″. I used fiberfill at a density of 0.9 lbs/ft^3 and installed a modified Radio Shack 40-1022 4″ woofer that I planned to use in an Octaline (*SB* 3/87, p. 9) and a Peerless K010DT 1″ tweeter. I simply set the Freeline up against the wall in the manner of the Simpline (*SB* 2/93, p. 14) and the Unline (*SB* 4/88, p. 28) and temporarily mounted the tweeter on the front of the enclosure just above the woofer. I biamped the setup with a Marchand fourth-order active filter, with the crossover set at about 2kHz. This breadboarded setup sounded great.

I next tried an Audax TW 74 tweeter, resetting the crossover to about 3.5kHz. This also sounded good. Finally, I tried the passive crossover that I used in my Octaline. Again, it sounded fine. Having satisfied myself that it would work, I set it in a corner and forgot all about it.

A year or so had passed when I received a mysterious package from Larry Sharp of Mahogany Sound. The box contained Acousta-Stuf, which he had recently marketed and asked me to evaluate. (It turned out to be really great.) To test the stuffing material, I chose the Freeline with its two open ends, since it was quite easy to reach inside and exchange the stuffing. Acousta-Stuf and the Freeline were very happy with each other and let their joy be known in the way that speaker systems do best.

Once again, I abandoned the Freeline for a few years. I designed and built a folded version, but I wasn't happy with it (although there is a recent happy sequel to that). For

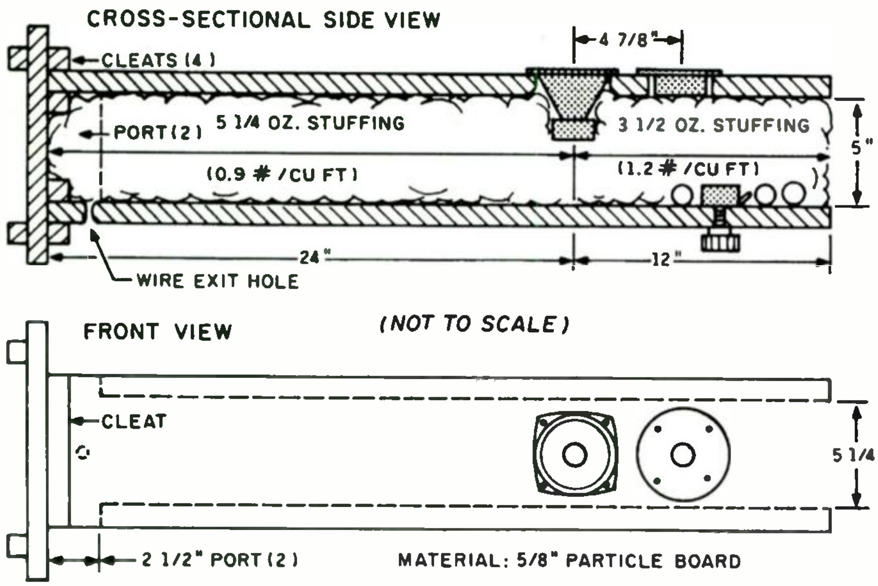

FIGURE 1: The Freeline design.

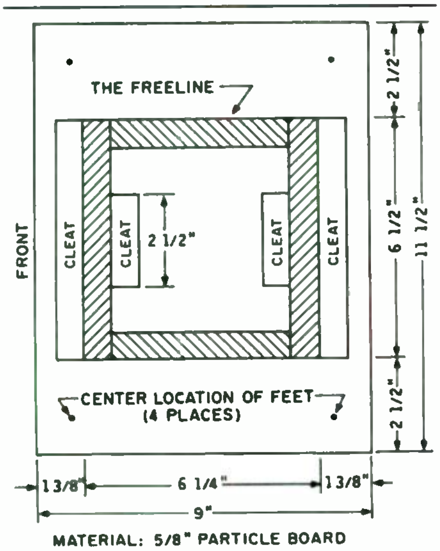

FIGURE 2: Base plate and the location of the Freeline.

the most part, the Freeline sat in corners, closets, and corners of closets, waiting for my inspiration to revitalize.

REINVENTING THE WHEEL

Shortly after my Simpline article appeared, I dragged out the Freeline with the idea of improving its posture. It was taller than the Simpline and, when angled up to the wall, presented a problem for careless feet, although it was handy for experimenting with time alignment of the two speakers. A mere change of the angle and the woofer/tweeter spacing was changed (I reached no conclusions, however). The operation was achieved with ease. I merely pulled out a little stuffing from the bottom of the Freeline (so it wouldn't get tangled in my sabre saw blade), and then made cuts on the bottoms of both side panels. The sides were 2½" shorter than the front and rear panels, causing two 2½" × 5" ports to appear at the sides of the Freeline when it stood upright (*Fig. 1*).

I then glued the bottoms of the front and rear panels to a 9" × 11½" base plate for stability. (You could perform this operation on the Simplines, especially if you're squeamish about leaning towers, and then you could call them the "Not-So-Simple-Simplines." Be sure to place the Simplines as far back on the base plates as stability allows, so the speaker will be as close to the rear wall as possible. This is an important design consideration.)

When placed upright on the carpet, the Freeline was a bit wobbly. It was too light for the surface area of the base plate. I solved this by gluing four cubes of 5/8" particleboard underneath the four corners of the base plate (*Fig. 2*). Perhaps spikes would work better, especially with high-pile rugs (but the cubes worked for me).

I conceived the Freeline idea about a year and a half prior to the idea that inspired the Unline article. As a result, I arbitrarily chose to use 0.9 lbs/ft^3 for the density of the fiberfill stuffing in the line. It was a fortunate choice, because it worked and made me realize that the Freeline is a viable design.

FOLDING THE WHEEL

After reading Darin K. Johns' article ("A 15" Transmission Line Woofer," *SB* 7/94, p. 10), I experimented with the folded Freeline that I briefly mentioned earlier. Mr. Johns referred to a slight midbass boominess in the woofer response that he eventually eliminated with additional stuffing.

The folded Freeline had a similar, but probably more aggravated, problem, which I attributed to configuration. I folded the line so the open ends were adjacent to one another. I decided it was a matter of mutual coupling. Despair and other activities led to the creation of another long-term closet inhabitant.

Mr. Johns' comments got me thinking again. This time I had the benefit of my Unline thoughts. I also decided that maybe the Freeline was functioning as two shorter TLs in parallel. I began to view the Freeline as an Unline in parallel with a Microline (*SB* 5/89, p. 28).

With this in mind, I restuffed the folded Freeline as a Microline and as an Unline. There was an immediate sonic improvement. Thank you, Mr. Johns (and *Audio Amateur* for publishing the article).

Following that happy experience, I restuffed the upright Freeline. I eventually (based upon listening tests) ended up with stuffing densities of about 1.2 lbs/ft^3 for the upper line and about 0.9 lbs/ft^3 for the lower one. I again used Acousta-Stuf, which is an excellent damping material, but fiberfill pillow stuffing also works very well. If you stuff the line as above and place the rear of the line (not the base plate) about 3" from the rear wall and away from corners, and you use the speaker complement which I shall prescribe, a marvelous natural bass extension for a 4" speaker, and a smooth clarity in the higher regions, will result.

NEW, OLD DRIVERS

Originally, I used one of the old Radio Shack 40-1022 speakers in the initial testing of the Freeline, but they are no longer available. Radio Shack has initiated at least two changes of these speakers, the latest one involving a considerable change of design. I did not know whether the current version of this speaker would still be suitable for use in the Freeline. I decided to try it because of its widespread availability to many *SB* readers and because, in the past, there was a great consistency from unit to unit. (I guess I also was curious and needed to determine if I would be able to make things work.)

The current 1022s have a higher resonant frequency and a higher Q_{TS} than the models of the past. The cone is different, with ribs as opposed to the smooth cone of yesteryear. A fiber gasket beyond the outer edge of the surround protects the surround if the speaker is placed face down (as when you solder the leads). I suppose the gasket is for mounting the speaker on the rear of a panel, but I cannot imagine who would choose to do that.

Because of the higher Q_{TS} of 0.45, as compared to 0.35 of the older models, I was more conservative about adding mass to the new model, in spite of the fact that the new one had an f_S that was 30Hz higher than its predecessor. I added 5g of lead in the form of nine B-B-sized, split-shot fishing sinkers. I slit them in half so they fit around the dust cap/cone joint very nicely (*Fig. 3*).

I brushed a coat of full-strength white glue onto the cone and dust cap, leaving a little puddle of the stuff around the joint at the base of the dust cap. Perform this operation after the speaker is installed and the enclosure is speaker-face-up on a kitchen stool or table top. While the glue is still wet, pick up the pieces one at a time with tweezers, or a pair of needle-nose pliers, and place them around the base of the dust cap. The flat side (the split side) faces the dust cap dome. The round part (the half ball) is on the up side, and the little tail rests down on the cone just below the first cone rib.

Space the pieces as best you can for appearances. When the glue is dry, apply another coat of full-strength glue to the cone and dome. Again leave a pool of glue at the dust cap/cone joint—just broad enough to reach the base of the first cone rib. The little tails of the lead will be covered. Only the balls will show above the glue ring. Allow the glue to dry in a warm area (68–70°); otherwise, the glue pool may remain milky and take longer to dry.

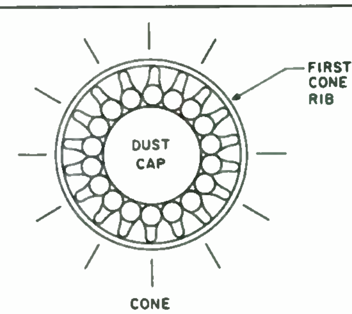

FIGURE 3: Showing the disposition of the 18 halves of split-shot B-B fish sinkers (see text) to the cone/dust cap junction of the woofer, to add mass.

INITIAL TESTING

The speaker didn't sound bad when I first listened to it in the Freeline, but I determined something was lacking in the lower, foundation end of the response. I finally corrected this by adding some series resistance to the positive leg of the woofer circuit to raise the speaker Q_T (making it a better match to the Freeline).

You have probably read that adding more than an ohm to a woofer circuit causes deterioration of the sound. This may be true in some circumstances, but in this case and with the Simpline, let me assure you that no such deterioration occurred, even though I used 10Ω in the Simpline and 8Ω in the Freeline. (In the Simpline a capacitor paralleled the series resistor to also shape the treble response.) I determined the value of both these resistors through many hours of music and voice material listening.

The tweeter is a current Radio Shack 1" dome tweeter (the 40-1276), and an excellent companion for this modified woofer. As with the Octaline, I was able to make use of a very simple crossover with fine results (*Fig. 4*).

I designed the Freeline, like most of my systems, for apartment or small-room use. It will play loud enough to satisfy most listeners, but it is not for those who wish to break leases with a sonic barrage. Its smooth, open, and detailed sound, coupled with its accurate musical timbre, makes the Freeline a very joyful speaker.

CONSTRUCTION

Building the Freeline is about as simple as it gets. If you build the prototype version (the one that leans), you simply assemble four sides and cut two speaker holes. The upright version is almost as simple. Cut the two 5" sides so they are 2½" shorter than the other two sides. Then cut a base plate board (this requires only one cut from one of the scrap pieces, for each Freeline) and the eight 5/8" cubes (or whatever you plan to use for Freeline feet). If you don't have a saw, most lumber companies will make the cuts for a modest fee.

The handiest material, and the easiest to work with, is particleboard shelves (4' long × 11½" wide × 5/8" thick). For a stereo pair of Freelines, you'll need four of them.

Cut 1' from one end of all four boards, leaving them 3' long. Rip off a 5" section down the length of all 3' boards. Cut off 2½" from one end of these four 5" wide boards (note: for the leaning version of the Freeline, don't make the 2½" cuts). At this point you have four side pieces, 5" wide and 33½" long, and four front or back pieces 6½" wide (minus the width of the saw kerf) and 36" long, if you are building the upright version. All the boards for the leaning version will, of course, be 36" long.

The upright version requires a base plate for stability. You can easily make this from a piece of the 11½" × 12" scraps. Simply cut the 12" length to 9" and then repeat this step on another scrap for the other Simpline. Carefully cut the 5/8" cubes. If you are unskilled or inexperienced at using a table saw, I suggest you use something else for the feet. Use ¼"-20 × 1¼" long, hex-head machine screws, placed in ¼" holes drilled about 1" in from the corners. With two nuts for each screw, they can be adjusted for leveling (*Fig. 2*).

On one of the 6½" wide boards, measure 1' down from the end and draw a line across the board at a right angle to its sides (*Fig. 5*). Make another mark 4-7/8" above the first cross line. Draw a line across the board parallel to the first one. Now draw a center line, parallel to the side edges of the board that intersects the two previously drawn cross lines and extends several inches above and below the lines.

HOLELY ORDERS

Set your compass to a radius of 1-13/16" and scribe a 3-5/8" circle with the compass point on the intersection of the center line and the lower cross line. Make the compass lines heavy enough so you can easily follow them when you are cutting with a sabre saw. This first circle is the marking of the woofer hole.

For the tweeter hole, set your compass to a 1-7/16" radius, scribe a 2-7/8" circle with the compass point set on the intersection of the center and upper cross lines. On the upper cross line, mark one point 1½" to the left of the center line and one point 1½" to the right of that line. Center-punch these two marks (or hammer a small nail hole) as a guide for the drill. These are the center points of ½" diameter clearance holes for the tweeter lead connectors.

Use a ½" wood bit. If you need to use a twist drill, make a small pilot hole, say 1/8" in diameter, to guide the larger drill. Drill these holes, which will be useful to guide the sabre saw around the cut, before you cut out the tweeter hole. Use a new wood blade, which is much easier to guide through the binding material in particleboard. Cut the holes with the saw blade just touching the inner side of the pencil mark.

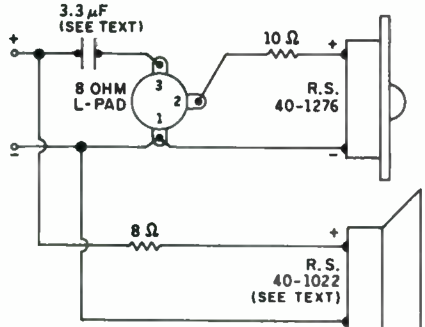

FIGURE 4: The crossover and woofer Q_T adjusting circuit.

On the lower cross line, mark a point ¼" inside the intersection of the 3-5/8" circle (*Fig. 5*), center-punch it, and drill a ½" hole (as per instructions above). This is the starting hole for the saw. Cut out the woofer hole. With sandpaper, a rasp, or a rat-tail file, break the corners of the circle edges (i.e., bevel the corners a little bit). Position the speakers in the holes, or enlarge the holes until they fit nicely, without binding, especially around the tweeter connectors, which are fragile. Make sure there will be enough room for the connecting wires when they are installed.

A nice touch is to make about a ¼" bevel around the back of the woofer hole to produce more space for the sound to enter the line. The system will still work without the bevel, or with a smaller one.

When the holes are correct and lightly sanded, brush a coat of white glue around them on the front face of the board, out to a distance of about 1". This helps produce a good surface to seal the speakers in place. Repeat these steps for the second front panel.

ASSEMBLY

You should preglue all joints by brushing both surfaces with a coat of glue and allowing it to dry before putting the structure together (with more glue, of course). Use white glue, which you will use to modify the woofer and is excellent and inexpensive for this type of work.

Start assembly with the back panel. Attach the two side panels to the back panel. As with most of the enclosures I've built over the past several years, I use only glue in construction, mainly because I live in an apartment and don't wish to annoy my neighbors. Some jigs, fixtures, and weights I found at a flea market make it reasonable for me to do this.

Most of you may prefer to use nails along with the glue. If so, I suggest 1½" finishing nails about every 6" or so, starting about 1" in from the ends. The glue supplies the strength and seals the enclosure. The nails mostly help with the alignment and eliminate movement while the glue dries. If you are a novice builder, refer to my Simpline article (*SB* 2/93), which details the construction of a similar enclosure.

After fastening both sides to the back panel, allow the glue to dry. Then add anoth-

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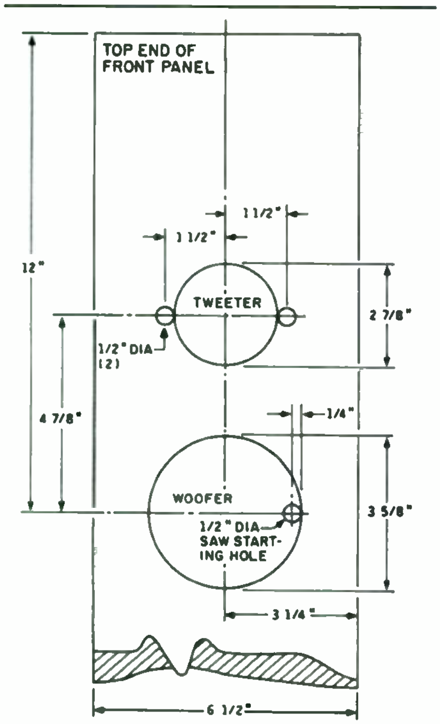

FIGURE 5: Speaker hole layout on front panel.

er coat to both joints for added strength and a good seal.

CROSSOVER

The original Freeline was an experimental enclosure, with the speaker wires merely coming out the top of the line and trailing down the back. The crossover was external for easy access. You may prefer a more sophisticated design with some sort of termination to hook your amplifier wires. Several types, such as five-way binding posts and push-button, are available, but you may have your own preferences. You could simply run the wires out one of the bottom ports, or drill a small hole on the back panel a couple of inches above the base plate and snake the wires out that way. You could solder a couple of banana jacks to them and use banana plugs on the amplifier wires.

It is a good idea to mount the crossover elements on the upper end of the back panel for easier access (simply pull out the stuffing). You could run the input wires to the crossover down to the terminals at the bottom (or to the little hole drilled at the bottom), then from the crossover to the speakers.

The crossover consists of only three elements: a 3.3µF capacitor, a 10Ω resistor, and an 8Ω L-pad. Another 8Ω resistor is included, but it isn't part of the crossover system, although it makes sense to group these parts together (*Fig. 4*). I wrote about it earlier when I mentioned matching the woofer Q_T to the Freeline. (Actually, this system isn't really a crossover; it's simply a high-pass filter, which is only half a crossover, the half that allows the tweeter to relax and kick back when it's feeling low. In the case of the Freeline the woofer takes care of itself.)

A Mylar® (polyester) film capacitor worked well for the 3.3µF high-pass. You could use a polypropylene type, but I don't recommend a nonpolarized electrolytic capacitor (such as the one that came with my tweeter). If you can't find a 3.3µF, try three 1µF capacitors in parallel. After I listened to this configuration for several days, it sounded just as good as the 3.3µF capacitor.

You can glue the resistors and capacitor(s) to the inside of the rear panel with RTV silicon rubber, an adhesive caulk such as DAP Qwik Seal (my favorite), or hot glue. If you use 5/8" particleboard, the Radio Shack 25W, 8Ω L-pad's shaft will be just long enough to reach through, if you leave out the escutcheon plate, the extra nut, and the washers.

WIRING

After the crossover elements are in place and the adhesive has dried, proceed with the wiring. Use only rosin-core electronic solder. I used 16-gauge zip cord.

Make the speaker leads long enough so about 8" will extend beyond the speaker mounting holes. Next, run the speaker input lines down to the terminations, or to the little drilled hole I mentioned previously. Leave enough slack in these lines for about an inch of the stuffing between the lines and the back panel to prevent the lines from vibrating against the panel during bass passages.

STUFF IT

With the wiring completed, except for the actual hookup of the speakers, start placing the stuffing material. Lay the enclosure on its back and, with two blocks or books underneath it to allow room for the L-pad knob, draw a line across the inside back panel, 1′ down from the top of the back panel. Place 3½ oz of polyester fiberfill pillow stuffing or Acousta-Stuf above this line and 5¼ oz below it.

A beam balance scale, or perhaps a postal scale, makes measuring the stuffing fairly easy. You will probably need to weigh several portions and add them together. Absolute accuracy isn't required (these hybrid TLs appear to be quite forgiving), but try to come close.

Place the lower amount (5¼ oz) of stuffing in the enclosure after pulling apart any tight knots and clumps. Lay it in as evenly as you can. You should lift up the wires to the terminals on the back panel and place an inch or so of the stuffing under them to isolate them from the enclosure walls. For now, you can bring the loose speaker wires up into the upper part of the enclosure.

Fill the upper part of the line *after* you fasten the front panel in place. It is easy to reach in and dress the speaker wires after mounting the speakers, and the upper part of the line is short enough to easily reach in and place the stuffing.

MAKING UP THE FACE

Before attaching the front panel, be sure to preglue all the surfaces. But first, press down the stuffing along the edges where you will apply the glue, so the fibers won't interfere with the joint. Also, place newspapers under the enclosure to catch the drippings.

Then place a rather massive glue bead (about ¼" wide) along the upper surfaces of the two sides. (Before doing this, you should remove the enclosure, still on its back and blocked up, to a place sturdy enough to allow the pounding of nails, which you should have driven partway through the front panel. Drive the four corner nails deep enough so the points just show through the back of the panel, flush with the surface.)

Carefully place the front plate (make sure the speaker holes are on the upper end) by lining up one edge and the top with one of the sides and holding it in place while you drive the corner nail in about halfway. Check the alignment of the front and side panel at the other end of the side panel (the port end) and drive in that nail halfway. Repeat this process for the other side of the front panel.

When everything is lined up, drive in all the nails on the front panel and wipe off the excess glue with a damp paper towel. Wipe the inside of the line to where it is free of stuffing. The lower part must fend for itself. (After the glue has dried, brush some glue over the outside joints of the enclosure and the inside joints on the upper 1′.)

SECURING THE SPEAKERS

Return the enclosure to its face-up position. Set the speakers in place. Mark the location of the mounting holes and punch the hole centers to provide a starting indentation for a 1/16" drill and a pilot hole for the #6 × ¾" panhead sheet metal screws to secure the speakers.

Place a bead of sealing material, such as Mortite, around the back of the speaker mounting flange and press it down so it won't fall off when you hold the speaker face up. Make sure you carefully knead together the ends of the bead to prevent leaks at the joint. Also be sure the bead is on the inside of the mounting holes.

Place the woofer face down on the front panel at the edge of the woofer hole, with the

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Specifications

Impedance	8 Ohm
Resonance	65 Hz
Upper Frequency	10,000 Hz
SPL 1/W/1/M	84 dB
Power Handling	50 W
Qts	0.41
Vas	4.54 L
Voice Coil	25.5 mm
Magnet	10 Oz.

$14.62 EA

DOB130R-PP/8 5-1/4"

- Polypropylene Cone
- Rubber Surround
- 2-Way Reflex Systems
- Excellent for Car-Stereo

Specifications

Impedance	8 Ohm
Resonance	45 Hz
Upper Frequency	9,000 Hz
SPL 1/W/1/M	87 dB
Power Handling	50 W
Qts	0.49
Vas	13.88 L
Voice Coil	25.5 mm
Magnet	10 Oz.

$17.58 EA

FQB165R-PP/8 6-1/2"

- Polypropylene Cone
- Rubber Surround
- Closed Box or Small Bass
- Excellent Power Handling

Specifications

Impedance	8 Ohm
Resonance	36 Hz
Upper Frequency	8,000 Hz
SPL 1/W/1/M	86 dB
Power Handling	60 W
Qts	0.36
Vas	29.26 L
Voice Coil	35.5 mm
Magnet	18 Oz.

$25.91 EA

FQB210R-PP/8 8"

- Polypropylene Cone
- Rubber Surround
- Smooth Response
- Medium Size Base Reflex Enclosure

Specifications

Impedance	8 Ohm
Resonance	33 Hz
Upper Frequency	6,500 Hz
SPL 1/W/1/M	87 dB
Power Handling	60 W
Qts	0.44
Vas	75.73 L
Voice Coil	35.5 mm
Magnet	18 Oz.

$31.05 EA

SHIELDED SPEAKERS

DOB100R-PP/8SC 4"

- Double Magnet with Cup
- Home Theater Center Channel
- Computer Multimedia
- Sealed Box Enclosure

Specifications

Impedance	8 Ohm
Resonance	65 Hz
Upper Frequency	10,000 Hz
SPL 1W/1M	85 dB
Power Handling	60 W
Qts	0.35
Vas	6.3 L
Voice Coil	25.5 mm
Magnet	5 & 10 Oz.

$19.17 EA

DOB165R-PP/8SC 6-1/2"

- Double Magnet with Cup
- Home Theater Center Channel
- Computer Multimedia
- Vented Enclosure

Specifications

Impedance	8 Ohm
Resonance	36 Hz
Upper Frequency	6,000 Hz
SPL 1/W/1/M	86 dB
Power Handling	80 W
Qts	0.30
Vas	35.28 L
Voice Coil	25.5 mm
Magnet	5 & 10 Oz.

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wire-connecting terminals facing the hole. Plug in your soldering iron. (I recommend a pencil soldering iron of about 30–35W, with a 1/8" chisel point. Be sure you tin the point correctly, according to the manufacturer's instructions.) Strip about 3/8" of insulation from the speaker leads and twist the leads tightly between your fingers.

Tin these ends and the connecting lugs on the speaker, leaving excess solder on the parts. Bring the lead that comes from the 8Ω resistor to the positive (marked +, or red spot) connector on the woofer. Grab the lead with a pair of needle-nosed pliers, about an inch behind the tinned lead. Twist your wrist so the end of the lead is pointing downward (the speaker is upside down).

Bring the tinned lead into contact with the tinned connector and hold it there. Now with the soldering iron in your other hand, place the tip so it contacts both the lead and the connector. If you provided enough excess solder on the parts, a smooth fillet of solder will form at the joint. Remove the soldering iron and hold your hand very still until the joint solidifies. Now do the other wire.

With the wires soldered in place, recheck the bead of Mortite for damage and the sealing surface of the front panel for dust. Lift the speaker and, pivoting around the edge closest to the woofer hole, turn it face up. With the wires going neatly into the woofer hole, carefully center the speaker so the pilot holes are directly under the speaker mounting holes and gently lower the speaker into place.

When driving the mounting screws, protect the speaker cone with your hands or lay a magazine across the cone. Screw the heads down to the face of the speaker flange and, a quarter of a turn at a time, turn one screw and then the one opposite it. Finish the two other screws in the same manner.

Continue this process until the speaker is firmly in place, but not so tightly it distorts the flange. The Mortite provides a good seal without being completely compressed. If any of the Mortite should extrude beyond the flange, remove it with the screwdriver tip.

Repeat this step with the tweeter, allowing enough space so the fragile tweeter connectors are not compressed against the front panel. If you need to refile the hole, be sure there is no dust on the sealing surfaces when setting the tweeter in place. Place the final 3½ oz of stuffing into the top end of the line, as evenly as possible, around the wires in such a way they don't vibrate against the panel or each other.

SPEAKER SETUP

Now you can listen to your Freeline. Place it about 1' from the rear wall and away from side walls and then tip it backward until the rear corner edge docks up against the wall. Hook up your speaker wires, turn the L-pad knob fully counterclockwise (the off position), and then back, clockwise, so the tip of the pointer travels about 3/8". This will set the knob close to where it should be. With this ultra-simple crossover, I didn't allow myself (or you) the luxury of a nicely centered, correctly positioned L-pad marker.

TABLE 1

FREELINE PARTS LIST (for a pair)

QTY	DESCRIPTION
4	4' × 11½" × 5/8" particleboard shelves (or equivalent)
1	24 oz bag of polyester fiberfill pillow stuffing (17½ oz actually required) or Acousta-Stuf
2	Radio Shack 40-1022 4" woofers (the current model, with the ribbed cone)
2	Radio Shack 40-1276 1" dome tweeter
2	25W, 8Ω mono L-pad Radio Shack 40-980 (or equivalent)
2	3.3µF polyester (Mylar) film capacitors, or six 1µF (see text), Radio Shack 272-1055 1µF caps OK. Don't use electrolytic.
2	5W or 10W, 10Ω wire-wound resistor, Radio Shack 271-132 OK.
2	20W, 8Ω wire-wound resistor, Radio Shack 271-120 OK.

MISCELLANEOUS

White glue, 1½" finishing nails, 16-gauge zip cord, #6 × ¾" sheet metal screws (16), Mortite (for sealing in the speakers), DAP Qwik-Seal caulk (or equivalent), a package of B-B-sized split-lead-shot fish sinkers (18).

If you plan to use machine screws for the baseboard feet, you will need: ¼"-20 × 1¼" long, hexhead (or sockethead) machine screws (8), ¼" nuts (8), with washers, if you're fussy.

One of the constraints I placed on myself (perhaps foolishly, considering the added effort) was to design a system you could build from Radio Shack parts without compromising its sound. Despite their claims as a supply store, Radio Shack's selection (value-wise, not quality-wise) of resistors and capacitors was dismal. But I still managed to accomplish my goal.

When you are satisfied you have listened enough, place and center the Freeline on the base plate (*Fig. 2*). Scribe a pencil line around it and glue the front cleat in place. You can use cleats of quarter-round stock for a better appearance.

When the glue has dried, turn the base plate over and glue on the little cube feet, or you can drill the ¼" holes for the ¼"-20 × 1¼" hexhead (or sockethead) machine screws for the alternative feet. Next, preglue all the surfaces involved in the mounting of the Freeline on its base, including the rear cleat, the two shorter inner cleats, and their mating surfaces. Push the stuffing up, or remove some of it to make working in that area easier. (Don't forget to set things back in order later.)

Set the baseboard on a firm level surface, on a piece of newspaper. Apply the required glue on the baseboard and the rear surface of the previously glued front cleat. Set the Freeline—facing the correct way (I goofed once, on another system!)—on the baseboard, lining it up with the pencil lines and pressing it forward, up against the front cleat.

Apply glue to two adjoining surfaces of the rear cleat. Press it into place behind the back of the Freeline and the base plate and line it up. Repeat this procedure with the two shorter cleats, which are placed inside, through the port openings. Leave the Freeline undisturbed for several hours (or overnight), while the glue dries. Then replace or adjust the stuffing.

The functional part of the Freeline(s) is now completed. I leave the matter of appearance in your hands.

Set the Freelines with the back edge of their base plates ½" from the rear wall. Allow 3' or so of space between the speakers and the side walls and 7–8' between the speakers. In limited space they could be spaced only 5' apart, but the soundstage will not be as wide or open.

TUNE-UP

As previously explained, set the L-pad knob pointer about 3/8" above the off position. Listen to your best material at that position, even if it doesn't sound quite right. When you decide to move the knob, do so only about 1/16" and listen again for quite a while. Patiently continue with subtle movements. It will pay off in rich, natural, very detailed, balanced sound. If you go astray, return to the 3/8" position and start again.

Your test listening material should include passages in which the full swell of the orchestra is crowned with brasses. If you set the treble too high, you will lose the full orchestral sense. If the treble is too low, you lose clarity, and the bass sounds a bit mushy or vaguely hollow. When you reach the right balance point, you'll feel power and a sort of monolithic solidity (assuming the recording is capable of such performance).

In real life we hear voices from a little distance, so it helps to compare reproduced voices under similar conditions. When you reach the correct balance adjustment for naturally reproduced voices, listening to close-miked recordings can be a great experience.

Organ music, of the quality of the Wilson Audiophile recordings, is excellent test material. WCD-177/278, *Concert and Recital*, with James Welch, organist, is probably my most-used reference. The first seven cuts were recorded using an organ

located seven or eight miles from where I live and which I have heard several times live. The third cut contains a beautiful interaction between the full-range organ and an English horn (played by Robert Hubbard). The most subtle nuances of timbre float into the room with a correctly adjusted prototype Freeline. (This cut played through a pair of Simplines still leaves me in awe and with a greater appreciation of the laws of physics, even after dozens of times. Space them about 7′ apart.)

White noise (or FM interstation hiss) is very helpful in obtaining a good treble balance. Turn the knob of the L-pad back and forth very slightly and very slowly, in the area where the treble response just begins to come in and go out of the general overall white noise sound. Try to set it where it just balances, or perhaps achieves a very slight "bloom" over the overall sound. I know I'm getting close to the correct adjustment when the midrange really starts to sound right. It's amazing how much even a slight amount of excess treble takes the edge off the naturalness of the midrange.

I hope you enjoy building and listening to your new Freelines, and I predict years of musical enjoyment to come.

COMPUTER-AIDED BASS HORN DESIGN

By A.L. Senson

Designing and building a horn system is not as straightforward as doing a vented or closed-box design. The complexity of the computation involved can discourage anybody with a limited math background. Calculating the old-fashioned way can take several hours, and changing a dimension or trying a different driver, flare rate, or value for M can result in a long day at the drawing board.

This program, patterned after Brian Smith's tractrix horn-design program,[1] calculates dimensions for a catenoid, exponential, or hyperbolic bass horn and prints out in less than a minute. Although it is written in QBASIC, the program flow is similar. I also used the same abbreviations and mnemonics, where applicable, to preserve continuity and some form of standardization. It includes some additional features, however, that will make you more productive at the drawing table. Not only does it provide a horn's general data, such as mouth and throat areas and back-chamber volume, but it will also give you the dimensions to design your enclosure. The formulas were taken from Dr. Edgar's "Show Horn" article.[2]

Type in QBASIC at the DOS prompt to start the program. From the File menu, choose New and type in the horn program. Pay special attention to the punctuation marks, which are vital to the program's proper operation. Save the file as "horns.bas," turn on your printer, then choose Start from the Run menu.

USER INPUT

Just like the tractrix horn-design program, this one starts by asking you to enter the driver's Thiele-Small parameters. The woofer you select must have a reasonably high free-air resonant frequency (f_S) and a low Q_{ES}. The high-mass roll-off frequency (F_{HM}) is immediately computed, and, if it is below 280Hz, a warning message appears stating that the selected driver may not be suitable for horn applications. You then have a choice of continuing or trying another driver. Answering no terminates the program.

The formulas for throat area and back-chamber volume give the output in square inches. V_{AS}, on the other hand, is expressed

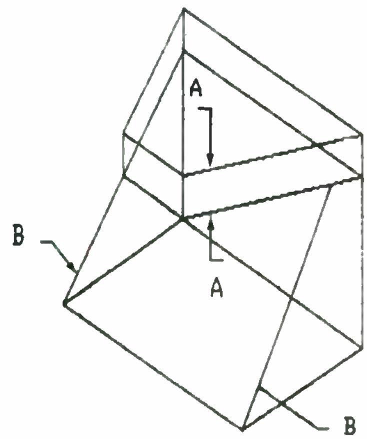

FIGURE 1: A and B show the parallel sidewalls of a horn.

differently by various speaker manufacturers and distributors. Peavey and Zalytron catalogs list this parameter in liters; Parts Express and MCM use cubic feet; and Electro-Voice uses both cubic inches and cubic meters. The program, however, allows you to enter V_{AS} values no matter which unit is used. Just follow the directions on the screen.

The program then asks if your horn type is to be free-standing, floor type, or corner horn. According to your response, the program determines the appropriate mouth size and the experimental back-chamber volume. Later on, the program stops the printer at a point just after the mouth size, when it does the calculations for the air column. Select the appropriate letter and then enter your desired flare frequency.

GRAFTED TRAPEZOIDS

Since no one is likely to build a circular horn, I did not include calculations for radii information. For ease of construction, most horns are built using grafted trapezoids, or, more correctly, trapezoidal prisms (*Fig. 1*). Starting at the throat, the driver exhausts into the initial horn section that has two parallel and two nonparallel sidewalls, following as much as possible the horn's expansion rate in the form of a trapezoidal prism. This initial section joins another trapezoidal prism at a 90° bend, and continues the expansion rate to the mouth.

The program asks you for the distance between the parallel sidewalls (A). Then it asks for dimension B, which can be an alternative choice for A or another parallel sidewall dimension of the next section.

After you enter a value for M, the screen clears and the printer prints out the horn's data, including the appropriate mouth area and back-chamber volume for the desired horn type. The printout displays in columns the distance from the throat, the horn area, one-half the horn area, and the corresponding dimensions for the nonparallel sidewalls A and B every 2″. This takes between 45–90 seconds, depending on your horn choice and computer hardware.

The results are printed accurate to six decimal places. However, you must decide on the final values for throat areas and back-chamber volume. You calculate the final back-chamber volume by using the referenced reactance-annulling techniques.[3] The only dimensions you should adhere to as closely as possible are area values at chosen points in the air column to maintain the proper expansion rate.

APPLICATION

The sample printout (*Fig. 2*) is for a Peavey 12″ Black Widow driver intended for use in a folding configuration, as in *Fig. 1*, with the following T/S parameters: f_S = 50.4Hz; Q_{ES} = 0.246; and V_{AS} = 96 liters. The program converted V_{AS} to 5,857.92 in^3 (3.39 ft^3). Dimension A (the initial section after the throat) is 5″, and dimension B (the width of the rest of the horn) is 24″.

ABOUT THE AUTHOR

A.L. Senson is an electronics-engineering technology student at Tidewater Community College (VA). His interests include playing the piano and bass guitar, MIDI, and building horn speakers.

For a copy of this listing, send a stamped, self-addressed, business-size envelope to *Speaker Builder*, Editorial Dept., PO Box 494, Peterborough, NH 03458. These programs are available on 3½″ DS/HD disk as SOF-HORIB3 for $19.95 plus $3 shipping USA, $5 other surface, or $10 other air. Contact Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467.

At about 35″ from the throat, the A column shows the distance between the nonparallel sides to be equal to dimension B (24″). The B column shows this point to be equal to dimension A. This is then the grafting point for horn sections A and B. The rest of the horn will use column B, starting at the 35″ point.

You can use the half-area values when designing a horn with dual symmetrical flow paths, such as the Electro-Voice Sentry IVB, LaScala, or Klipschorn. These values can be used directly to design a half-section. You can run the program again with other data if your horn has more than two sections with different sidewall dimensions, or if you wish to try out different values for M, flare frequency, or even a different driver.

You cannot use the program directly for rear-loaded horns, which can tolerate higher Q_{ES} and lower F_{HM}.[4] The back chamber, coupled to a throat port, acts as a low-pass filter and you must treat it differently, as shown by the formula:

$$V_{BC} = 2 \times Q_{ES} \times f_S \times V_{AS}/F_{BC}$$

where V_{BC} is the back volume, and F_{BC} is the upper roll-off corner frequency due to the cavity compliance of the back chamber. You can use the program's calculations for the sidewall dimensions, however, for this application.

PROGRAM MODS

- Line 10 (LET B = 280) controls the cutoff point for marginal horn drivers, in this case an F_{HM} of 280Hz. If you use the horn as a subwoofer, change this value accordingly.
- Line X = 2 (before the DO statement) and line X = X = 2 (before the LOOP UNTIL statement) set the area expansion to be calculated every 2″. You can change this value also.
- The calculations for the nonparallel sidewall dimensions are based on full-area values. To calculate for half-area values, change lines S3! = S!/HT1 and S4! = S!/HT2 to S3! = S21/HT1 and S4! = S21/HT2.

Although I will not cover it here, by adding a few lines to the program, you can modify the sidewall calculations so all four sidewalls expand outward.

REFERENCES

1. B. Smith, "Tractrix Horn Design Program" (*SB* 4/86): 29-31.
2. B. Edgar, "The Show Horn" (*SB* 2/90): 10-23.
3. D. Plach and P. Williams, "Reactance-Annulling for Horn Loudspeakers," *Radio-Electronic Engineering*, Vol. 24 (Feb. 1955): 15-17.
4. T. Giffard, "Horn Loudspeaker," *Elektor Electronics* (May 1990): 20-22.

To continue further experimentation on tractrix horns, you can incorporate the V_{AS} conversion and sidewall calculations into Brian Smith's program. You must delete the radius calculations and lines that no longer apply. This, however, can become messy if you are new to programming. Although the flow is similar, the formulas for flare calculations are different. If you need help on modifying the tractrix program, write to me, c/o *Speaker Builder*.

Lastly, I did not write this program to produce a computer-designed horn speaker, but rather to give you the tools to serve as design aids on the drawing table. As mentioned

to page 73

```
###############################################
#                                             #
#           DATA FOR A HORN DESIGN            #
#                                             #
#       WITH DUAL SYMMETRICAL FLOW PATHS      #
#                                             #
###############################################

Driver's parameters:Name      Peavey 1202
                    Fs        50.4 hertz
                    Qes       .246
                    Fhm       409.7561 hertz
                    Vas       5857.92 cu. inches

Horn data:Flare rate          50 hertz
                    M         .6
                    Type      Corner Horn
          Rear chamber        2047.76 cu. inches
            Mouth area        725.1496 sq. inches
           Throat area        33.803 sq. inches
```

Length fm throat	Horn area at X	One-half horn area	5 inches A sides	32 inches B sides
2	35.79323	17.89662	3.579323	.5592692
4	38.00002	19.00001	3.800002	.5937504
6	40.4425	20.22125	4.04425	.6319141
8	43.14184	21.57092	4.314184	.6740913
10	46.12146	23.06073	4.612145	.7206478
12	49.40718	24.70359	4.940718	.7719871
14	53.02748	26.51374	5.302749	.8285545
16	57.01378	28.50689	5.701378	.8908403
18	61.40062	30.70031	6.140061	.9593846
20	66.22604	33.11302	6.622603	1.034782
22	71.53188	35.76594	7.153186	1.117686
24	77.36414	38.68207	7.736414	1.208815
26	83.77339	41.8867	8.377339	1.308959
28	90.81521	45.4076	9.081521	1.418988
30	98.55064	49.27532	9.855063	1.539854
32	107.0468	53.52338	10.70468	1.672606
34	116.3772	58.18861	11.63772	1.818394
36	126.6229	63.31147	12.66229	1.978484
38	137.8728	68.93638	13.78728	2.154262
40	150.2242	75.11209	15.02242	2.347253
42	163.7843	81.89217	16.37843	2.55913
44	178.6708	89.3354	17.86708	2.791731
46	195.0126	97.50629	19.50126	3.047071
48	212.9514	106.4757	21.29515	3.327366
50	232.6429	116.3214	23.26429	3.635045
52	254.2576	127.1288	25.42576	3.972776
54	277.9831	138.9915	27.79831	4.343486
56	304.0251	152.0125	30.40251	4.750391
58	332.6092	166.3046	33.26092	5.197019
60	363.9835	181.9917	36.39835	5.687242
62	398.4198	199.2099	39.84198	6.225309
64	436.2168	218.1084	43.62168	6.815887
66	477.7023	238.8511	47.77023	7.464098
68	523.2357	261.6178	52.32357	8.175557
70	573.2121	286.606	57.32121	8.956439
72	628.0646	314.0323	62.80646	9.813509
74	688.269	344.1345	68.8269	10.7542
76	754.347	377.1735	75.43471	11.78667

FIGURE 2: Sample printout of the Bass Horn Design program.

A MIKE/PROBE PREAMP FOR SOUND-CARD MEASUREMENTS

By Bill Waslo

The recent availability of high-performance PC-based multimedia sound cards has opened up some interesting possibilities for loudspeaker and audio electronics designers needing sophisticated test equipment. The mass-produced and therefore economically priced plug-in card includes two 16-bit analog-to-digital converters (ADC), two or more 16-bit digital-to-analog converters (DAC), and antialiasing and interpolation filters.

The more advanced units feature a programmable digital signal processor (DSP), a wide selection of sample rates, and a good chunk of local memory. This considerable collection of stimulus and data-acquisition gear can cost under $180, and, along with some appropriate software (such as my Liberty Audiosuite), is all you need for high-quality, lab-based network, loudspeaker, audio spectrum, or distortion analysis.

Or, I should say, *almost* all you need. Good lab equipment should be more than just functional in principle; it must also be practical. And here, sound cards on their own can run into some difficulties. Liberty Audiosuite began as an all-software, IMP-like product for use with commercial sound cards, avoiding the need for Liberty Instruments to deal with parts ordering and stocking, inventories, kitting, and so forth. Practical considerations and the desire to make the system complete, however, leads me to strongly suggest the addition of minimal extra hardware.

This article describes an external battery-powered Mike/Probe Preamp for use with sound-card-based measurement systems (*Photo 1*). And while you can get this device in various levels of kit or assembled form from Old Colony Sound Lab, Liberty Instruments, or perhaps other sources, you can also make your own with minimal cost and difficulty.

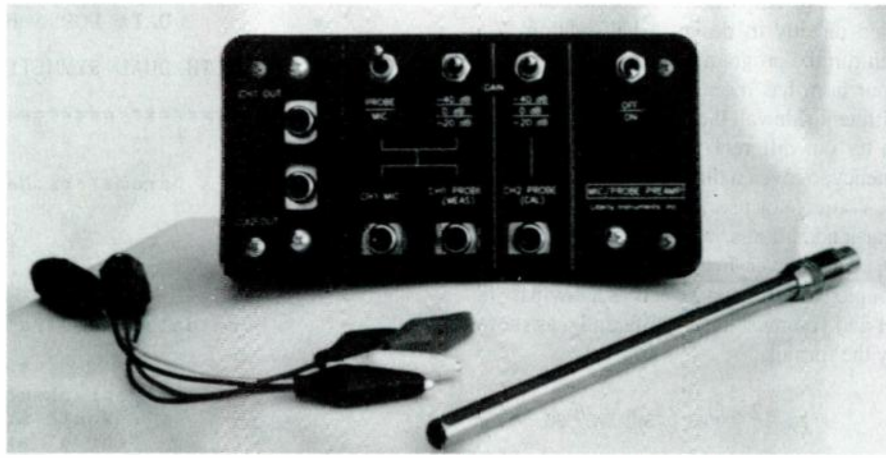

PHOTO 1: Mike/Probe Preamp with test probes and mike wand (photo by Darek Ball).

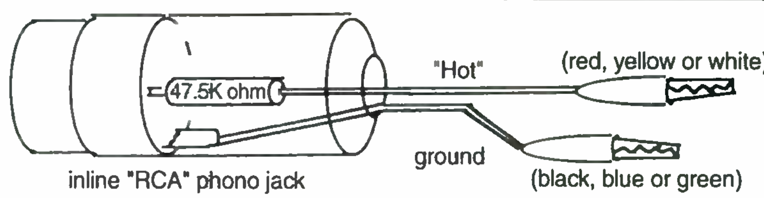

FIGURE 1: IMP-type probe, for connection to long patch cable.

FLIES IN THE OINTMENT

One problem of which you should be aware if you plan to use a sound card to examine speaker-level voltages is input voltage limitations. These cards aren't typically designed to deal with the 30 or more peak volts which routinely appear across your speaker terminals. The maximum peak input voltage handling of a sound card is often 2 or 3V, as the aux inputs are intended for low line-level sources. The sound card might let you know that this has been exceeded by ceasing operation permanently, a rather high price to pay for one misadjusted level control.

Another problem is the type and quality of the microphone preamplifiers on typical sound cards. These are usually designed for low-output, inexpensive dynamic microphones, which need no battery or phantom power but don't even approach measurement quality. The preamps are typically rolled off in the high end (the intended mikes have little output there anyway, so the rolloff is an improvement) and are very susceptible to digital noise pickup in the environment of a computer's insides.

Even if you can stand that, if you drive the mike input with an IMP mike or capsule (powered by some external means) or a Mitey-Mike, for example, you will likely drive the internal mike preamp into distortion, invalidating any kind of measurement. The gain of the internal mike preamp just isn't set up for condenser mike levels. And the condenser mikes lack sufficient output to comfortably drive the line inputs.

DESIGN OPTIONS

An obvious way to increase the voltage handling of the sound-card line input is to simply attenuate the voltage being measured. This is easy and requires little more than a resistor or two. But with the attenuator in line, you end up with a different problem: now you can't get normal-level signals up near full-scale. A 16-bit digitizer is only 16 bit if you can drive it to near full-level. While you don't need any extra total gain before the sound-card input, you would like to be able to run the input wide open without risk of damage.

You can solve this with a set of attenuators or a variable attenuator. But the lower attenuation settings result in either higher operating impedances on the connecting cables (which results in rolloffs with longer runs) or low probe impedances (which can

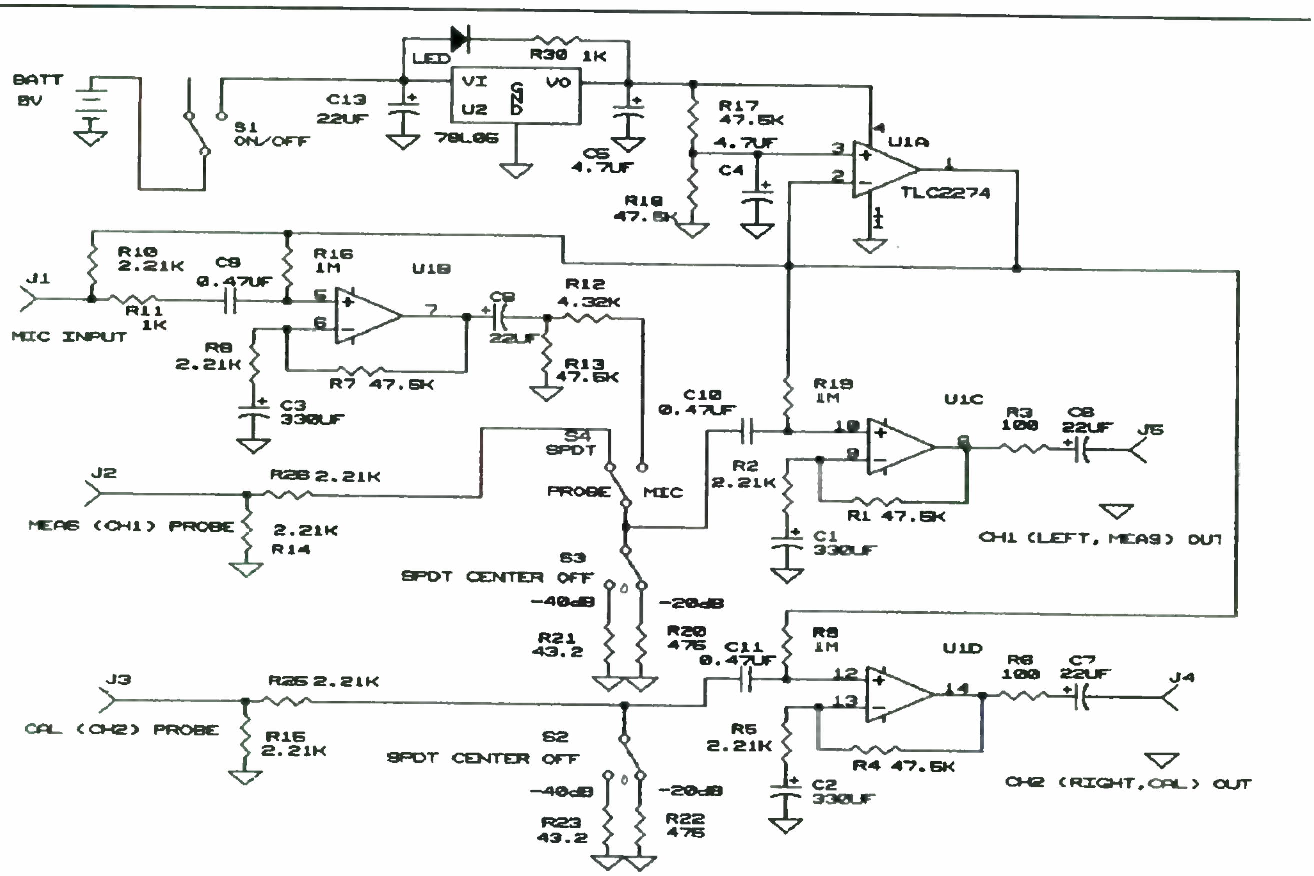

FIGURE 2: Mike/Probe Preamp schematic.

upset the circuit being measured). And you again run the risk of frying your sound card. You could add protection or clamp diodes, along with limiting resistors, on the lines, if you don't mind the added distortion.

CIRCUIT DESCRIPTION

But, since we need to provide a decent mike preamp anyway, a few extra blocks of active stages cost little more and can maintain a high probing impedance along with low operating impedances for the cable runs, input protection, and low distortion. They also allow more versatility by providing several more precision attenuation range steps.

The probe type intended for use with the Mike/Probe Preamp is the 47.5k resistor probe as used with IMP (*Fig. 1*). This is simply a clip probe with a 47.5kΩ resistor in series with the hot lead, which feeds a cable going to the preamp. The 47.5kΩ resistor provides this high impedance to the circuit under test, while the 2.21kΩ load at the preamp keeps cables up to 25′ from introducing a significant rolloff.

The attenuation introduced by these helps the input voltage handling, and the active stages in the preamp restore the gain. You can easily make a pair of these probes with some wired alligator clips and in-line phono jacks available at Radio Shack or other parts stores. Tie a knot in the alligator clip wires after passing them through the phono jack cover and install the resistor inside the jack.

The mike stage is essentially the same as that in the IMP unit, only battery operated. It provides phantom power for the Panasonic WM-60 or 63 electret capsules as used in IMP, or can work with Mitey-Mike. It uses section B of U1 (*Fig. 2*), and a TLC2274 or TLC274 CMOS quad op amp, which provides good bandwidth and low noise with low battery consumption. Section A of

TABLE 1

MIKE/PROBE PREAMP PARTS LIST

REF	DESCRIPTION
CAPACITORS	
C1–3	330µF, 6.3V miniature axial electrolytic
C4, 5	4.7µF, 10V tantalum
C6–8, 13	22µF, 25V miniature axial electrolytic
C9–11	0.47µF 5% film
RESISTORS	
R1, 4, 7, 13, 17, 18	47.5k ¼W 1% metal film
R2, 5, 8, 10, 14, 15, 25, 26	2.21k ¼W 1% metal film
R3, 6	100Ω ¼W 5%
R9, 16, 19	1M ¼W 5%
R11, 30	1k ¼W 5%
R12	4.32k ¼W 1% metal film
R20, 22	475Ω ¼W 1% metal film
R21, 23	43.2Ω ¼W 1% metal film
MISCELLANEOUS	
U1	Texas Instruments TLC2274CN (TLC274CN alternative)
U2	78L05 low-power regulator
LED	Super-bright red LED (Radio Shack 276-087 or Mouser 592-SLH56VC3
J1–5	I/O connectors. For PC mount, use Mouser 161-0164 (see text)
S1, 4	SPDT toggle switch. For PC mount, Jameco 26331 or equivalent.
S2, 3	SPDT center-off toggle switch. For PC mount, Jameco 26315 or equivalent (also, normal SPDT if –40dB position not required)
Enclosure	For all PC-mount construction, Radio Shack 270-233 or equivalent. Cover plate requires drilling or punching.
Battery holder	For PC mount, use Mouser 534-1294; otherwise, any standard 9V battery clip and holder.
	0.25″ long, #4 tapped round spacers (3)
	5/16″ #4 machine screws (to mount board to cover plate and battery holder to board) (8)
	#4 lockwashers (5)
	#4 nuts (2)
	PC board (see text) (1)
	Not Used: C12, R27, R28

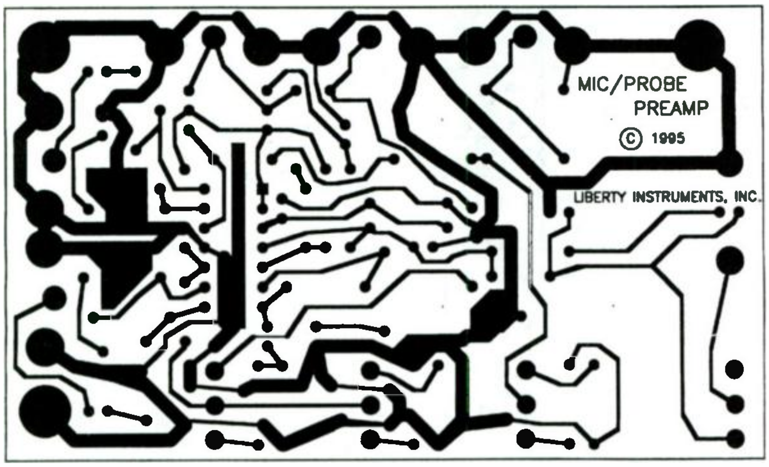

FIGURE 3: Board pattern (solder side).

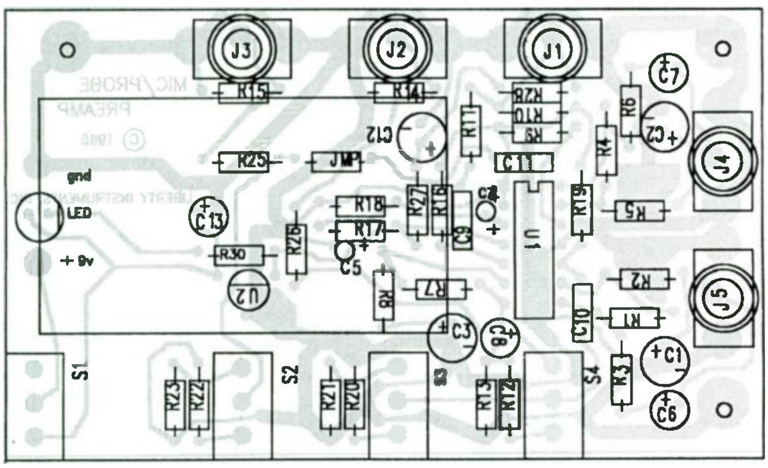

FIGURE 4: Stuffing guide.

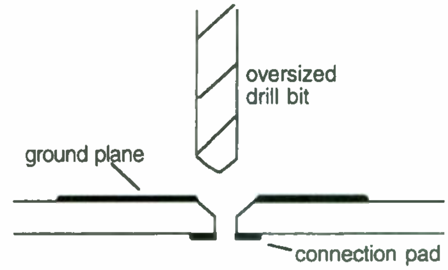

FIGURE 5: Clearing the area around a hole in ground plane layer.

U1 provides the internal virtual ground and mike bias voltage using the output of the main power supply 5V voltage regulator U2 as the reference source. Two other sections of the quad op amp package serve as gain restoration stages for two channels of probe input.

The main channel 1 (or left) measure output can be switched between mike or probe #1. The channel 2 (right, or Cal in Liberty Audiosuite) output is always from probe #2. Each input gain can be optionally scaled down by either 20dB (1/10) or 40dB (1/100) for use with peak signals up to about 100V, by setting the on-off-on toggle switches. If you don't need the –40dB setting, you can skip the center-off toggle switch (and R21 and R23) and use the more commonly available SPST or SPDT types.

A fourth toggle provides power switching, and a super-bright type LED will indicate when the voltage regulator has sufficient headroom to do its job. If the LED becomes dim, it's time to replace the battery.

A printed circuit pattern is provided in *Fig. 3*, with a stuffing guide in *Fig. 4*, for those wishing to etch a circuit board. The board can be a double-sided, plated-through type, but need not be. You really need only process the solder side (*Fig. 3*), since the primary function of the pattern for the component side is to allow PC-mounted phono connectors to be strengthened by soldering their ground pins on both sides of the board. If you use the board-mounted connectors, you can duplicate this two-sided ground easily by making the component side of the board a solid ground plane and removing the copper from around each component hole (to prevent shorting) by partially drilling with a larger drill bit (*Fig. 5*).

The board is set up so all of the components, from switches to jacks, can be mounted directly to it. Even the battery holder mounts to the back (solder side) of the board and solders into the holes provided. You can nicely mount the entire Mike/Probe Preamp in a plastic project box, with a suitable punched or drilled cover for the switches and access to the phono connectors.

Be careful if you are drilling the holes in the cover; the metal edges can be sharp and the drill bit can catch the cover and set it spinning. I recommend sheet-

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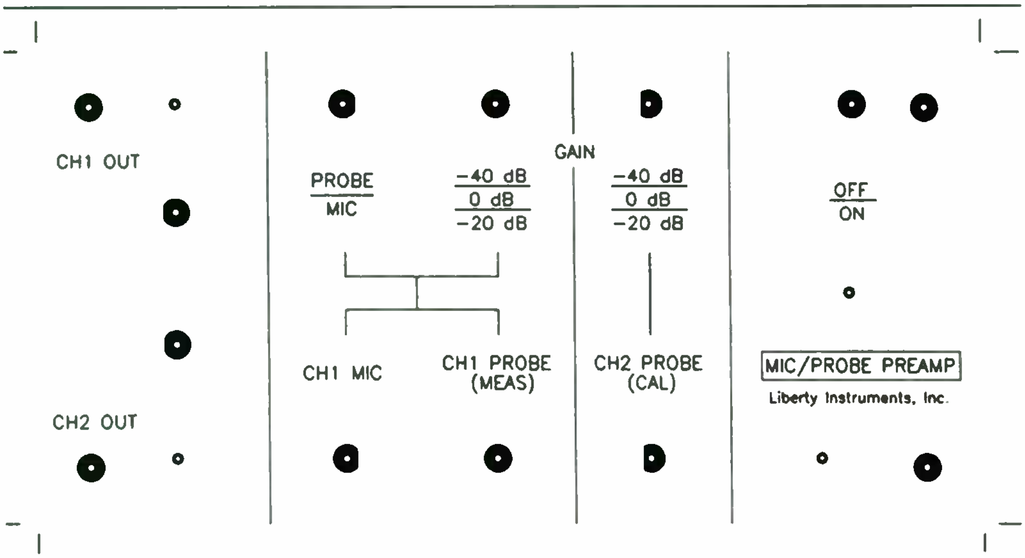


FIGURE 6: Front panel silk-screen pattern (white silk screen on flat black paint).

vifa

Audiphile Price List April 1995

Madisound Speaker Components
(8608 University Green)
P.O. Box 44283
Madison, WI 53744-4283 U.S.A
Voice: 608-831-3433
Fax: 608-831-3771

Model	Size Description	Imp. Ω	Fs Hz	Qts	Vas Liters	X-Max mm Peak	Power Watts	Rating db @Freq	Sens. 2.83V/1M db	Cost Per Unit
D19TD-03 No Magnetic Field	¾" Dome[***,3]	8	1400				50	12@5K	90.5	$13.50
D19TD-05	¾" Dome[1,3]	8	1700				80	12@5K	89	$11.50
D20TD-05	¾" Dome[1,4]	6	1700				100	12@5K	88	$12.50
D19AD-05	¾" Dome[1,5]	6	1700				100	12@5K	87.5	$18.20
D25AG-05	1" Dome[1,5]	6	1500				100	12@4K	90	$20.20
D25ASG-05 Shielded Alum.	1" Dome[1,5,***]	6	1500				60	12@4K	90	$24.10
D25AG-35	1" Dome[1,2,5]	6	850				100	12@3K	89	$24.80
D25TG-05	1" Dome[1,3]	6	1500				90	12@4K	90	$14.70
D25TG-85	1" Dome[1,2,3]	6	750				100	12@3K	90	$19.90
D26SG-05 Shielded textile	1" Dome[***,1,4]	6	1450				60	12@4K	92	$20.40
D26TG-05	1" Dome[1,4]	6	1500				100	12@4K	91	$15.90
D26TG-35	1" Dome[1,2,4]	6	940				100	12@3K	90	$20.30
D27TG-05 Flat Face	1" Dome[1,SILK]	6	1000				100	12@4K	91	$16.30
D27TG-35 Flat Face	1" Dome[1,2,SILK]	6	650				100	12@3K	90	$20.70
D27TG-15 Bulged Face	1" Dome[1,SILK]	6	1000				100	12@4K	91	$16.30
D27TG-45 Bulged Face	1" Dome[1,2,SILK]	6	650				100	12@3K	90	$20.70
H25TG-05 Horn Loaded	1" Horn[1,3]	6	1500				100	12@4K	95	$16.60
H25TG-35 Horn Loaded	1" Horn[1,2,3]	6	750				100	12@4K	94	$21.30
D75MX31	3" Dome Mid[4]	8	300	Both these mids require chambers.		.5	80	12@500	92	$33.80
M10MD-39	4" Cone Mid[6,8,10]	4,8	110			.65	80	12@700	88	$22.10
M11WH-09	4.5 " Cone Mid[6,8,10]	8	68	.31	4.3	3	30		87	$33.20
P11WH-00	4 ½" Woofer[6,7,10]	8	52	.21	10.3	3	30		88	$33.40
P13MH-00	5" Cone Mid[6,7,10]	4,8	60	.33	10	1	40	6@200	89.5	$30.80
M10-IC, M13-IC	Isolation chamber for M10MD-39—$1.50				Isolation chamber for P13MH—$4.00					
M13SG-09 Shielded Magnet	5" Woofer[***,6,8,10]	8 / 16	54	.35	12	2	35		88	$30.00
P13WH-00	5" Woofer[6,7,10]	4,8	60	.33	10	4	40		88	$31.50
C17WG-69	6½" Woofer[8,9]	8	44	.55	35.8	3	50		89	$18.20
M17SG-09 Shielded Magnet	6½"Woofer[***,6,8,10]	8	34	.34	53	3	50		89	$32.00
P17SJ-00 Shielded Magnet	6½"Woofer[***,6,7,10]	8	41	.35	33	4	70		87	$42.00
P17WJ-00	6½" Woofer[6,7,10]	4,8	37	.35	35	4	70		88	$32.30
M18WO-09	6½" Woofer[6,8,10]	4/8	35	.37	28.5	4	70		87.5	$44.40
P21WO-12	8" Woofer[6,7,10]	8	33	.34	85	3	70		91	$43.00
P21WO-20	8" Woofer[6,7,10]	8	28	.33	113	4	80		91	$45.30
M22WR-09	8" Woofer[6,8,10]	6	30	.33	55	6.5	150		88	$65.20
P25WO-00	10" Woofer[6,7,10]	8	24	.28	178	4	80		90	$49.00
M26WR-09	10" Woofer[6,8,10]	8	26	.32	130	6.5	130		88.5	$70.40
Automotive Speakers and Home Applications Requiring 4 Ω Speakers. (All drivers are suitable for automotive use!)										
D26NC-05 Neodymium	1" Dome[1,4]	6	1800				30	12@3K	88.5	$25.70
D26NC-05 Accessaries:	Flush mount kit Each $3.80; Surface mount wedge $3.80; Replacement voice coil $8.00									
A13WH-01	5" Coax[6,7,10]	4	110	.50	3.5	3	40		89	$48.50
A17WG-01	6½" Coax[6,7,10]	4	58	.60	15	3	60		84	$49.00
M18WN-19	6.5" Woofer[6,8,10]	4	63	.66	10	4	110		88	$42.50
M22WR-19	8" Woofer[6,8,10]	4	48	.70	20	5.5	150		88	$64.20
M22WR-29 Dual Voice Coil	8" DVC Woofer[6,8,10]	4/4	47	.66	20	5.5	75/75		87	$66.30
M26WR-19	10" Woofer[6,8,10]	4	44	.69	53	5.5	150		90	$73.10
M26WR-29 Dual Voice Coil	10" DVC Woofer[6,8,10]	4/4	40	.67	53	5.5	75/75		89	$76.50
Grills	D25/D26/H25/M10MD ----$7/Pair				P13—$10/pair		P17—$12/pair		P21—$15/pair	

1) Ferrofluid
2) Chamber and Tinsel Leads
3) Polyamid dome
4) Textile dome
5) Aluminum dome
6) Magnesium Cast Frame
7) Mineral Filled Polycone
8) Very stiff conical Paper cone.
9) Foam surround
10) Rubber surround
*** Double Magnet/ Shielded

FOUR-POSTER SPEAKER STANDS

By Duncan MacArthur

The practice of calling a small speaker a "bookshelf" speaker is almost always false advertising. You should place most, if not all, of these speakers anywhere except in a bookshelf. The best location is often on a speaker stand. Although many good commercial units are available, customizing your own offers significant advantages.

DESIGN DECISIONS

I designed the four-poster stands to accompany Vance Dickason's Audax A652 design (see "Kit Report," *SB* 2/95, p. 42). These don't exactly match since the Madisound-supplied box has slightly different dimensions from (although the same internal volume as) the original design. But they're close enough to work well. You should build your speakers before the stands.

A good speaker stand should do four things well. First, it should support a "bookshelf" or other small speaker (some bookshelf speakers are not very small) at the correct height for listening. Second, it should couple the speaker solidly to the floor. This is especially important with carpeted floors, where the coupling should be to the subfloor, but is also important with other surfaces.

Third, the stand should be solid and well-damped so it doesn't interfere with the sound of the speaker, and strong enough to support the speaker. You can use an elegant "single support" stand for a small speaker, but a larger speaker (such as the A652) may require the larger support system of a four-poster stand. Fourth, you might consider the aesthetic appeal of the stand-speaker combination. If your spouse considers any stand and any speaker ugly by definition, you might convince him/her otherwise with a more "finished" product. While you should not sacrifice acoustic properties for aesthetic reasons, pleasing everyone involved has its own reward.

PARTS DESCRIPTION

The basic structural element of my four-poster stand (*Photo 1*) is a section of PVC pipe, which attaches to both the top and bottom of the stand with circles of wood that fit

PHOTO 1: A completed four-poster speaker stand. I designed this stand to complement the Audax A652 speaker system. It is 22" high and has a 9" × 10¾" upper surface. Speaker spikes connect the bottom of the stand to the floor, and integral support cones fasten the speaker to the stand.

snugly inside the pipe. These wheels screw onto the top and bottom pieces (*Fig. 1*).

Rather than attaching the wheels directly to the pipe, I assembled the unit with a threaded rod running down the center and held against each board with a nut and washer. This "sandwich" provides structural rigidity only when you tighten the nuts.

The threaded rods include points ground onto the upper ends that extend above the top of the stands and connect to the speakers. The pipes are filled with dry sand, both to increase the mass and to help damp spurious vibrations. The integral speaker supports are linked directly into this sand for added damping. Insert carpet-piercing spikes (not shown in *Fig. 1*) into the bottom of the stands to increase stability and improve the coupling of the stand to the floor.

Assembly of the four-poster stands involves many steps, but is straightforward. No special tools are required. The parts for a pair of stands cost about $100. You can build them in a couple of weeks of part-time work. I believe the results are worth the cost and effort, and the finished units work as well as, and look better than, commercial models costing several times as much.

I build stands for specific speakers, screwing them to the stand for optimum coupling. You can permanently attach the four-posters in this manner, but these stands provide good support even without screws.

STEP-BY-STEP INSTRUCTIONS

Although assembly is not particularly dangerous, this project requires the use of power tools. Always take appropriate safety precautions.

You can construct a pair of speaker stands using the parts listed in *Table 1*. Each numbered instruction step is followed by a short description and occasionally more detailed instructions. Those techniques useful in other projects are identified by the word "TIP." The figures supplement the assembly sequence.

1. Cut four pieces of ¾" particleboard to about 10" × 12" and four more pieces to about 13" × 15". The exact dimensions are not important, but make the cuts as straight as possible. Glue these boards together to form two pieces 10" × 12" × 1½" (tops) and two pieces 13" × 15" × 1½" (bottoms).

TIP: When gluing two boards together to form a thicker one, cut them about 1" oversize in each direction. Then cut the thicker composite down to the desired size. The two pieces will line up exactly with no visible joint.

2. Cut the four 1½" thick blanks down to 9" × 10¾" × 1½" (tops) and 12" × 14" × 1½" (bottoms).

DRILLING HOLES

3. Mark for the holes, located at the corners of a 5" square, in both tops (*Fig. 2*). Also mark the screw holes, if you plan to secure the speakers to the stands.

TIP: If the important dimensions are the hole-to-hole measurements (5" in this case), make these values exact and let the measurement from the hole to the edge of the board vary slightly. This way the squareness and spacing of the columns does not depend on the accuracy of the board cutout.

4. Mark for the holes in both bottoms (*Fig. 3*). The main holes are again located at the corners of a 5″ square. Four speaker spikes are in the bottom surface of the bottom pieces. Mark these holes 1″ in from each side in each corner of the bottom pieces.

5. Drill 1/8″ holes at each marked location in the tops and bottoms, making sure you drill through the boards for the main supports and through the top boards for the speaker-mounting screws. Drill the spike-mounting holes 1″ deep from the bottom of the bottom pieces. Use a drill press or drill guide to make these holes perpendicular to the boards and ensure the squareness of the completed stands.

TIP: If you require precise hole location, drill first with a small bit (1/8″ in this case) and then enlarge the hole to the required dimension.

6. Using a speed bit, drill four 1-1/8″ diameter holes 5/8″ deep in the top of each top piece and in the bottom of the bottom pieces (*Fig. 1*).

TIP: If you need a large hole and a medium hole in the same location, first drill the larger hole part way through and then the medium hole through the rest of the board. Most drill bits rely on the center of the bit to guide the drill. Removing the wood with the medium drill leaves no support to guide the larger bit.

7. Drill 3/8″ holes the rest of the way through the tops and bottoms in the main locations.

8. Drill holes (3/8″ diameter, 1″ deep) in the bottom of the bottom pieces for the spike inserts. I used Fowler Tiptoes from Madisound.

TIP: If you need a mounting hole for an insert, drill a bit deeper than the length of the insert, and don't screw it in all the way.

9. Optional: Drill 11/64″ holes (for #8 screws) through the tops to screw the speakers to the stands. Countersink these holes on the underside of the tops.

10. Round the corners (1½″ dimension) of all four pieces of particleboard with a ¼″ rounding-over router bit.

TIP: When rounding the thickness dimension of several pieces, clamp them together rather than individually (*Photo 2*). This gives the router a larger surface on which to balance.

11. Use the same router bit to round the other edges of the tops and bottoms. Lightly sand all surfaces to remove any rough spots. Set these aside.

12. Cut eight pieces of 2½″ ID PVC pipe 18″ long. You must make square cuts. To make sure you're using squared pieces, you may purchase precut PVC pipe, which works perfectly for these stands, from speaker suppliers such as Madisound.

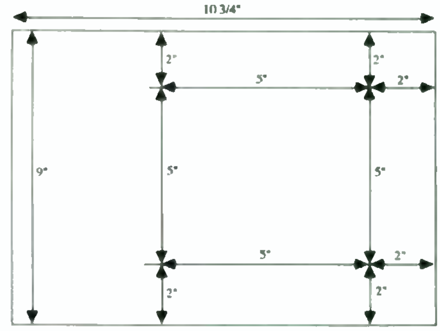


FIGURE 2: Marking pattern for the main holes in the tops of the speaker stands (not to scale). Additional holes are required for screwing the speakers to the stands.

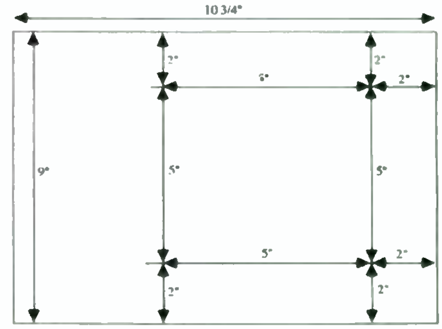


FIGURE 3: Marking pattern for main holes in the bottoms of the speaker stands (not to scale). Additional holes are required for the speaker spikes.

WHEEL ALIGNMENT

13. Sixteen 2½″ wooden wheels are required to attach the top and bottom pieces to the support pipes. If the wheels fit smoothly into the ends of the pipes, then skip to step 17; if not, reduce the wheel diameter as described in steps 14 through 16.

14. Insert the 2½″ hex-head 3/8″ bolt through a wheel. Fasten the bolt by adding a flat washer, lock washer, and nut. Screw

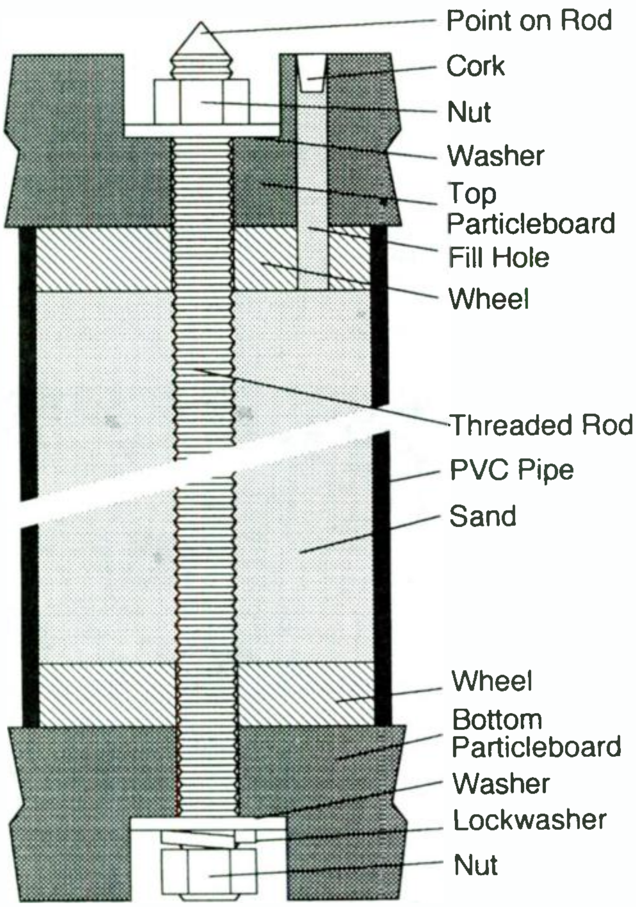


FIGURE 1: Cross section of the rod and pipe assembly of a single support column (not to scale). Four of these assemblies are used in each of the stands.

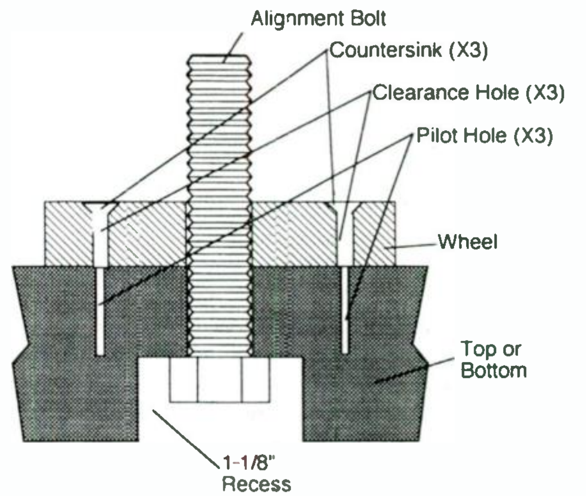


FIGURE 4: Detail of wheel screwed onto top or bottom board (showing two of the three mounting screws).

down the nut to compress the lock washer and hold the assembly tightly together (*Photo 3*).

TIP: A hex-head bolt has a smooth shaft, while the carriage bolt has a threaded shaft, slightly smaller in diameter. If a hex-head won't quite fit, try the carriage bolt.

15. Chuck this assembly into a 3/8" hand drill. Hold the spinning wheel against a piece of sandpaper (50 to 80 grit) to evenly reduce the wheel (the sandpaper should be resting on a hard, flat surface). Test the wheel frequently in the pipe to prevent oversanding.

TIP: If you must place a threaded rod or screw in a drill, use a metal file to smooth off the tops of the threads on the bolt. This gives the chuck a surface to grip without bending the remainder of the threads.

16. Repeat steps 14 and 15 for all 16 wheels. This set of hardware will now be scarred and unsuitable for reuse.

17. Attach four wheels to the inner surface (top of bottom pieces and bottom of top pieces) of each particleboard piece (*Fig. 4*). Use 1½" #8 wood screws (three per wheel) with the following steps.

17a. Push the 2½" carriage bolt through one of the main holes so the head is in the 1-1/8" recess. Place a wheel over the exposed end of the bolt (on the top of the bottom piece or the bottom of the top piece).

17b. Drill three 1½" deep, 5/64" diameter, pilot holes through the wheel into the board while the wheel is still positioned on the carriage bolt. These holes should be about ¼" from the wheel's outer edge.

TIP: Drill the holes in a nonsymmetric pattern. The holes index the proper orientation of the wheel and no reference mark is needed. Drill an extra "divot" in the upper surface of the wheel to identify that side if you drop the wheel.

17c. Remove the wheel (leaving the bolt in place), enlarge the pilot holes on the wheel to 11/64" through holes, and countersink the end of these holes on the exposed (divoted) side.

TIP: Using an "all-in-one" bit that drills all three holes in the end-grain of hardwood, such as the wheels, will produce an unwanted burning effect.

17d. Replace the wheel on the alignment bolt, line up the three holes, and screw down the wheel.

TIP: My favorite "screwdriver" is a drywall screw bit chucked into an old-fashioned bit-brace. This combination operates almost as fast as an electric screwdriver and gives you a much better "feel" for the screw and hole. Both Phillips and straight bits are available. One word of caution: this device can exert a very large amount of torque; it is easy to strip screw holes and heads if you're not careful.

17e. Repeat steps 17a–d for all 16 wheels on the four tops and bottoms.

17f. Mark each wheel on the top plates 11/32" in from the edge of the wheel, avoiding the screws already in place. Drill a ½" through hole at each of these locations. These holes are only required in the tops and are used to fill each pipe with sand. Since they are so close to the edges of the wheels and large holes, a small amount of chipping or breakthrough on either side is acceptable.

SPARE THE ROD

18. Check that the upper and lower surfaces of your top and bottom pieces resemble those in *Photos 4* and *5*, respectively.

19. Using the drill press or drill guide, redrill the 3/8" holes in all four tops and bottoms (16 places) to ensure the threaded rod passes easily through the parts.

20. Select the eight 3/8", 24" long, threaded rods. Using an electric grinder, grind a 3/16" to ¼" long cone on the end of each rod (*Fig. 1*). It isn't critical that these be identical, only similar to each other. Use the nut that was scarred in reducing the wheels to check the threads. If you have difficulty threading the nut onto the pointed end of the rod by hand, use a small metal file to clean up the damaged area on the rod until you achieve a smooth fit. (Put two nuts on each rod before clamping in a vise and cutting. Threading the nut near the cutoff cleans burrs.)

21. Measure 21-1/8" from the pointed end and mark the threaded rods. Cut the rods to length with a hacksaw. Again, the lengths need not be identical. Smooth off any burrs or sharp edges with a metal file, using the same nut and technique as above to check the threads.

TIP: When marking and cutting threaded rods, don't attempt to work on top of a thread. Both the marker and the saw will fall into the groove on either side. Instead, rotate the rod until the desired length falls into a groove. This will provide an easy starting mark for the saw blade.

DRY RUN

22. Practice assembling one stand to make sure everything fits.

22a. Place a flat washer, a lock washer, and a nut on the flat end of four rods. Screw the nut down so it is flush with the end of the rod.

22b. Thread all four rods through a bottom piece so the washers and nuts fit into the recess of the board. Stand this assembly on the floor with all four points upward.

22c. Place four sections of PVC pipe over the wheels on the bottom piece. The rods should stick up a couple of inches from the center of the pipes.

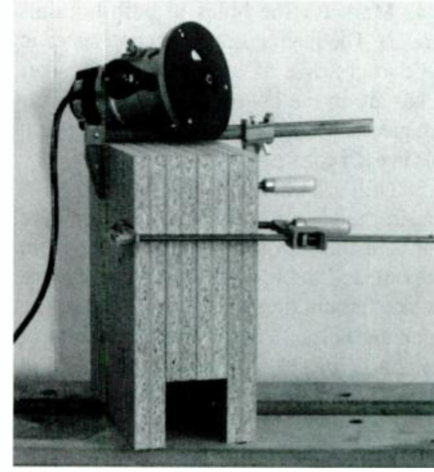


PHOTO 2: Two tops and two bottoms clamped together to round the corners.

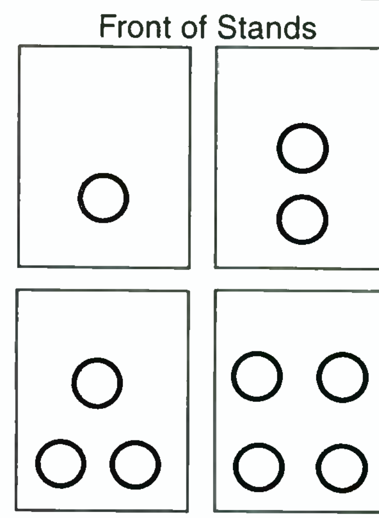


FIGURE 5: Design options for using one, two, three, or four support columns per stand. You should not use the built-in support cones with fewer than four columns. Larger-diameter pipe may be required for single- and double-column stands.

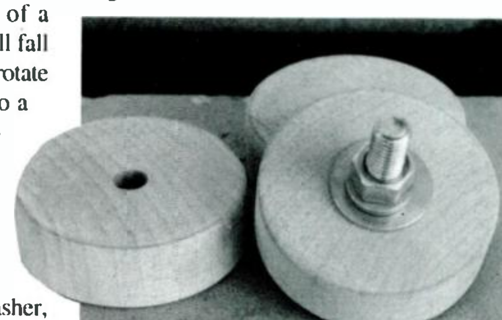


PHOTO 3: Hex-head bolt, wooden wheel, flat washer, lock washer, and nut. This assembly is chucked into a 3/8" drill to spin the wheel.

22d. Place a top board on the rod points. The wheels should be on the underside of the top and the recesses pointing upward. Match all four rods with the holes in the top. Push down evenly on the top to allow the rods to

come through the holes. Carefully adjust the pipes until all four fit around the wheels mounted on the top. Press the top down so it is supported by all four pipes.

22e. Place a flat washer and a nut over the pointed end of each rod. If these require more than finger strength to tighten, fix the threads now. At this stage you have plenty of time, but during final assembly caulk will be drying while you work.

23. Repeat steps 22a–e for the other stand. After everything fits, disassemble both stands and proceed with final finishing.

24. Use drywall repair compound or spackle to fill the cut edges of the particleboard tops and bottoms. You don't need to create a uniform layer of spackle, just enough to fill the voids in the particleboard. After the spackle has dried, lightly sand any excess.

25. Clean and rinse all eight sections of PVC pipe with detergent and water. Stand each pipe on end to air-dry.

TIP: Don't sand PVC pipe; you'll raise threads of PVC and roughen the surface.

26. Stand all eight pipes on end on newspaper. Spray-paint with flat black paint. Use several light coats and turn the pipes end-for-end several times to help the paint adhere. Don't worry if they aren't perfect, you can fix the paint later. One can of paint will cover eight pipes. Let the paint dry overnight.

TIP: Spray-paint in a well-ventilated area, such as outdoors on a calm day.

27. Paint all surfaces on the tops and bottoms with flat black spray paint. Pay particular attention to the areas which will be most visible later (i.e., the top of the bottom pieces and all edges of all pieces). One can of paint will cover two top pieces and two bottom pieces. Let the paint dry overnight.

28. Install the spike inserts in the bottom holes of the bottom pieces. Fowler Tiptoes require a 6mm hex key.

FINAL ASSEMBLY

The caulk will set about 15 minutes after application, so have everything ready before you start. Finish one stand at a time.

29. Place a flat washer, a lock washer, and a nut on the flat end of four rods. Screw down the nut flush with the end of the rod. Also prepare a 1/8" thick strip of clean wood, which you'll need later.

30. Apply a ring of black latex caulk around each wheel on a bottom piece.

TIP: Silicone caulk cannot be painted. If further painting is required, use a latex brand.

31. Thread all four rods through the bottom piece and fit the washers and nuts into the recesses in the bottom. Stand this assembly on the floor with all four points (and the caulked wheels) pointing upward.

32. Place four sections of PVC pipe over the wheels on the bottom piece. The rods should extend a couple of inches from the center of the pipes. Rotate the pipes to spread the caulk, leaving the excess for now.

33. Place a top board on the points of the rods. The wheels should be on the underside of the top and the recesses pointing upward. Match all four rods with the holes in the top. Push down evenly so the rods come through the holes. Carefully adjust the pipes until all four fit around the wheels mounted on the top. Do *not* caulk the top wheels. Press the top down so it is supported by all four pipes.

34. Place a flat washer and a nut over each pointed end and tighten with a wrench. Wipe off any excess caulk from the bottom of all four pipes.

35. Set the stand upright on two books so the bottom ends of the rods extend a bit out of the bottom plate. Loosen one of the top nuts. Using the 1/8" board as a guide, adjust the top nut to allow the point of the rod to protrude 1/8" beyond the top surface.

36. Tip the stand on its side, grasp the point of the rod with a pair of pliers (to keep it from turning), and tighten the associated bottom nut until the lock washer is

TABLE 1

FOUR-POSTER PARTS LIST

QTY	DESCRIPTION
HARDWARE	
17	3/8" nuts
17	3/8" flat washers
9	3/8" split ring lock washers
8	3/8" diameter × 2' long threaded rod
48	1½" long #10 flat-head wood screw
1	3/8" diameter × 2½" long hex-head bolt (14)
1	3/8" diameter × 2½" long carriage bolt (15)
8	Speaker spikes with inserts
WOOD	
4	9" × 10¾" × ¾" particleboard (tops) (2)
4	12" × 14" × ¾" particleboard (bottoms) (2)
16	2½" diameter wooden wheels
MISCELLANEOUS	
8	2½" ID × 18" long PVC pipes
3 cans	Flat black spray paint
	Black latex caulk (30)
50 lb	Play sand
8	½" corks

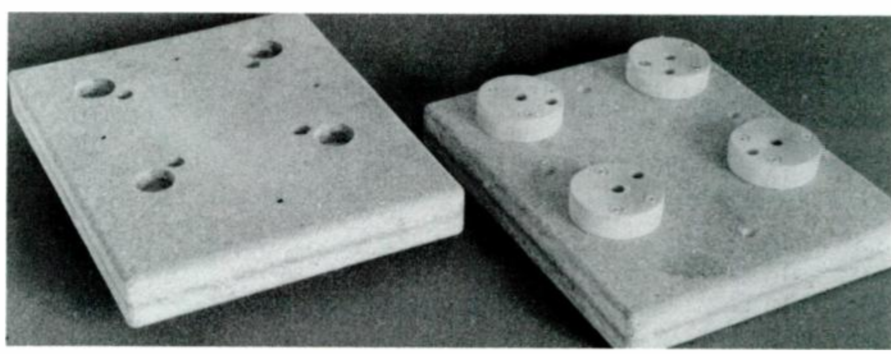


PHOTO 4: Top (left) and bottom (right) view of the top of the stand. The 1-1/8" recess should be on top of these pieces and the wheels should be screwed to the bottoms.

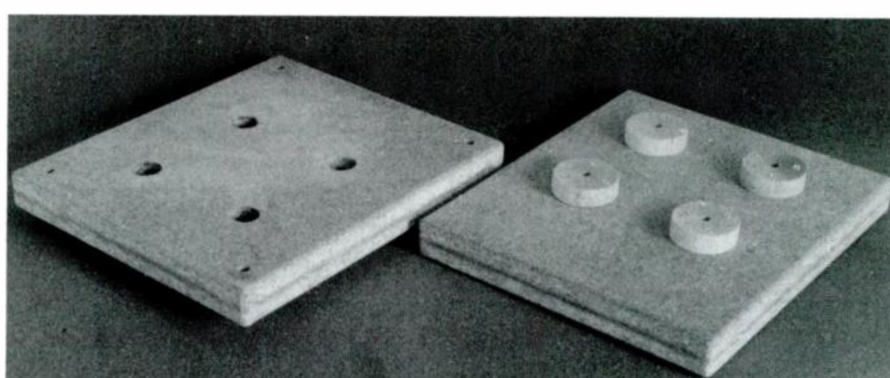


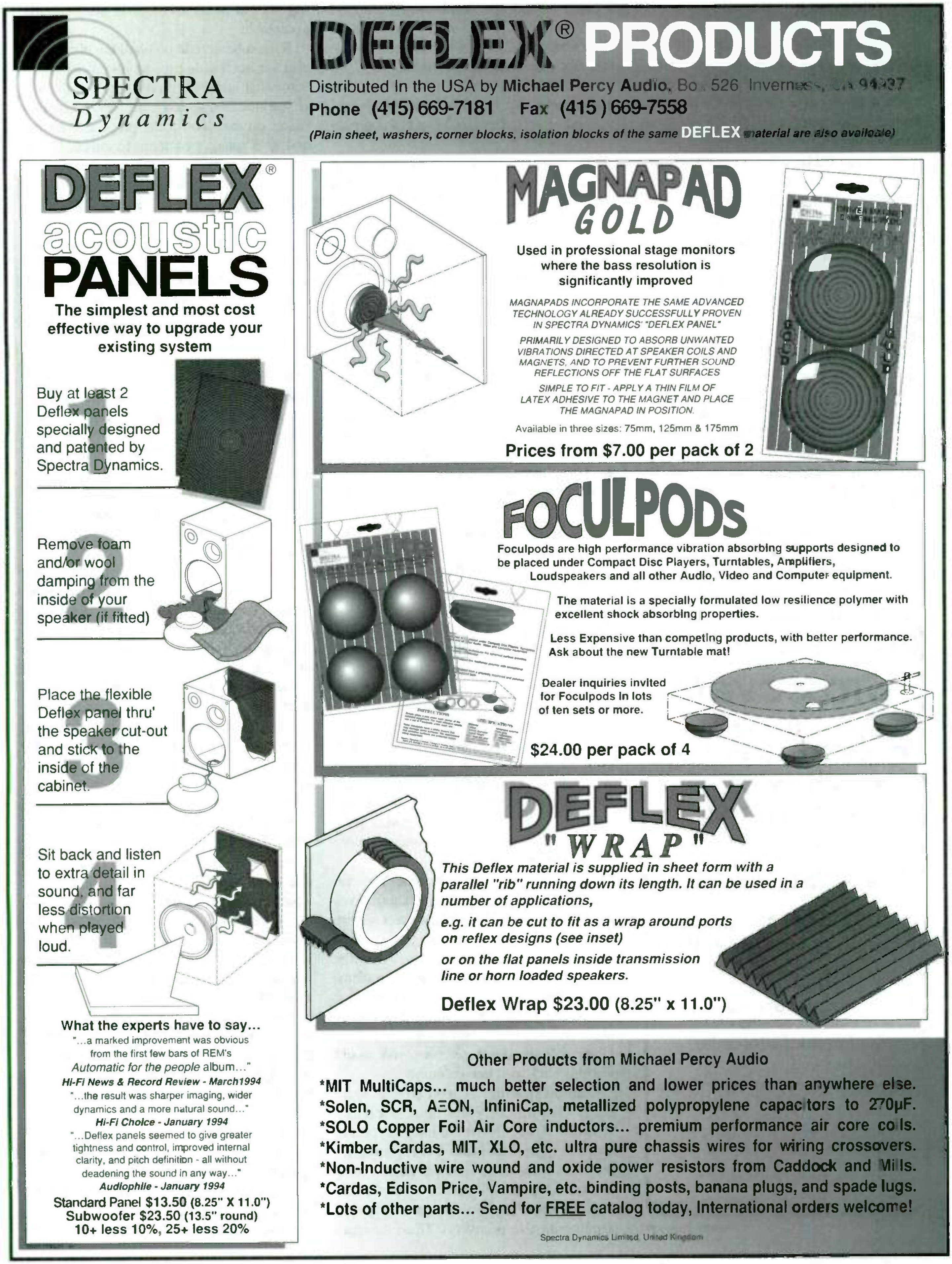
PHOTO 5: Top (right) and bottom (left) view of the bottom of the stand. The 1-1/8" recess should be on the bottom of these pieces along with the holes for the inserts, and the wheels should be screwed to the top.

SOURCES

Madisound Speaker Components
PO Box 44283, Madison, WI 53744
(608) 831-3433
Fowler Tiptoes, cut pipes

Cherry Tree Toys
PO Box 369, Belmont, OH 43718
(800) 848-4363
Wheels

Reader Service #21 →



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Hi-Fi News & Record Review - March1994
"...the result was sharper imaging, wider dynamics and a more natural sound..."
Hi-Fi Choice - January 1994
"...Deflex panels seemed to give greater tightness and control, improved internal clarity, and pitch definition - all without deadening the sound in any way..."
Audiophile - January 1994
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or on the flat panels inside transmission line or horn loaded speakers.
Deflex Wrap $23.00 (8.25" x 11.0")
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fully compressed.

37. Repeat steps 35 and 36 for each of the four rods, keeping three rods tight at all times to hold the stand together. All four points should extend uniformly above the top surface. It doesn't matter if a bit of rod extends below the bottom. The caulk in this stand should dry overnight.

38. Repeat steps 29–37 for the other stand.

39. Place both stands upright on newspaper. Touch up the paint job with a third can of spray paint, concentrating on the exposed areas. Let the paint dry overnight.

40. Carefully lay the stands on their sides. Screw in all four spikes and set them to the same height (~1"). Set the stands upright again.

TIP: A clean carpeted floor is ideal for final assembly.

41. Using a small plastic or paper funnel (e.g., a 2 oz plastic funnel), fill all eight pipes with dry sand through the ½" fill holes. Pound on the stand tops and tap the sides of the support columns to settle the sand. The two stands require about 40 lb of sand.

TIP: Pouring from a 50 lb bag of sand is difficult. It's better to transfer some of the sand into a bucket, and pour from it.

42. Top off the sand in each column from a paper cup. If the stands may be tipped over, push a ½" cork into each fill hole. Otherwise, play music with the speakers sitting on the points to settle the sand.

43. After a few days top off the sand in each column. Apply a small amount of wood glue on each of the eight ½" corks and plug each fill hole.

44. Align the speakers on the four points on top of the stands. After you're satisfied with the position and appearance of both the speakers and the stands, secure the speakers to the stands with 2½" #8 screws.

DESIGN OPTIONS

Stand height: You can design the stands for a specific speaker and match its height requirements exactly rather than compromising for salability. My design uses double particleboard (3"); the spikes extend about 1" from the bottom and the cones 1/8" from the top; and the length of the pipes is 18", for a total height of over 22". The final figure depends on whether you measure from the top of the carpet with the spikes sticking into it, but I don't think a 1/8" difference is significant here.

Top and bottom sizes: Another option for the amateur but not for the commercial designer is matching the size of the top boards with the speakers. This meets our previously stated criterion of increasing the aesthetic appeal without affecting the acoustical properties. The size of the bottom boards is more arbitrary. The larger the board, the further apart the spikes can be and the more stable the resulting stand.

However, the bottom board defines the footprint for the stand. Thus, the bigger it is, the more floor space the speaker will occupy, and the more unwieldy the stand will be. After surveying commercial stands, I determined 12" × 14" is the average size for a large stand. Modify these dimensions for your requirements.

Number of support columns: You can vary this number with several alternatives (*Fig. 5*). In general, heavier speakers require more support, which is another advantage of designing for a specific speaker. Also, the fewer support columns, the larger their diameters should be. The built-in support cones will not be very stable with three columns and will not work at all with one or two columns. If you use these cones, you should probably stick with four columns.

Column spacing: Support column spacing is also a purely aesthetic decision. However, if you choose the integral speaker support cones, column center spacing depends on the cone spacing. In this case, you should space both the cones and the column far apart (for speaker stability and stand rigidity, respectively). Thus, the spacing should be as large as is aesthetically acceptable.

Remember to add one-half the pipe diameter on each side and to place the pipes somewhat inward from the edges of the top. If you question speaker stability, place four points on the floor at the proposed spacing and sit a speaker on them to check. If you attach the speaker to the stand, this stability issue becomes secondary.

Diameter of pipes: This also affects the stand's appearance. Larger-diameter pipes will, in general, produce a more stable stand. However, you'll face two important constraints: (1) wheels and precut pipes are only available in smaller diameters, and (2) it is more difficult to make a square cut in a large pipe than a small one.

TOOLS AND TECHNIQUE

Rounding corners: I used a router and ¼" rounding-over bit to shape the edges and corners of the top and bottom pieces to produce a much more "finished" appearance. A stationary shaper does a faster job, but don't buy a shaper just for this project. An electric sander will work in a pinch, but it takes a long time and produces uneven results.

Drilling: All this project's holes (except the screw holes) should be perpendicular to the tops or bottoms. Even a slight misalignment of one of the support rods will result in a crooked stand. A drill guide and portable drill (which I used) will work; however, a drill press will make this easier.

Grinding and cutting rods: A hacksaw works fine for cutting 3/8" rods, and a metal file cleans up the cut edges. However, a power tool can really help you shape the points on the rods. Or, you could use a file, but count on 15 minutes and one skinned knuckle per rod. Use a grinder instead.

Wheels: You can cut your own wheels with a hole saw and an electric drill. This is inexpensive and allows for larger wheels (the largest size Cherry Tree stocks is 2½", which I used in this design). On the other hand, cutting wheels is tedious and the results are less uniform than purchased wheels. A hole saw in a portable drill works well, but results are much more uniform with a drill press.

Paint: The flat black paint works quite well on particleboard, but not very well on PVC pipes. I would be pleased to hear readers' suggestions for better methods of finishing PVC.

CONCLUSION

The PVC support assembly with integral speaker supports is versatile and attractive. The units meet all four of my stated criteria and enhance the appearance and sound of its matching speaker.

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- ☐ Acoustics: An Introduction to Its Physical Principles and Applications/Pierce. ASA. BKAC2. $52.95.
- ☐ Experiments in Hearing/von Békésy. ASA. BKAC3. $34.95.
- ☐ Acoustical Designing in Architecture/Knudsen, Harris. ASA. BKAC4. $34.95.
- ☐ Acoustics/Beranek. ASA. BKAC5. $44.95.
- ☐ Hearing: Its Psychology and Physiology/ Stevens, Davis. ASA. BKAC6. $34.95.
- ☐ Electroacoustics: The Analysis of Transduction, and Its Historical Background/Hunt. ASA. BKAC7. $34.95.
- ☐ Vibration and Sound/Morse. ASA. BKAC8. $44.95.
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- ☐ Digital Audio Signal Processing/Strawn, ed. BKAR4. $21.95.
- ☐ Feynman Lectures on Physics (3 vols.). BKAW2. $67.95.
- ☐ Loudspeaker and Headphone Handbook/ Borwick, ed. BKB1. $59.95.
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- ☐ Audio IC Circuits Manual/Marston. BKB24. $34.95.
- ☐ Newnes Audio & Hi-Fi Handbook/Sinclair, ed. BKB25. $79.95.
- ☐ Acoustics in the Built Environment: Advice for the Design Team/Templeton, ed. BKB26. $97.95.
- ☐ Advanced Audio Production Techniques/Ford. BKB27. $29.95.
- ☐ Audio Engineer's Reference Book/Talbot-Smith, ed. BKB28. $159.95.
- ☐ Op Amp Circuits Manual/Marston. BKB34. $34.95.
- ☐ Valve Amplifiers/Jones. BKB40. $39.95.
- ☐ Audio Electronics/Hood. BKB42. $35.95.
- ☐ Analog Electronics with Op-Amps/Peyton. BKC4. $37.95.
- ☐ Sound Reinforcement Handbook/Davis, Jones. BKHL1. $34.95.
- ☐ Modern Acoustical Imaging/Lee, Wade, eds. BKIE2. $79.95.
- ☐ Applications of Signal Processing to Audio and Acoustics (Wkshp.Proc.). IEEE. BKIE3. $78.95.
- ☐ Acoustics, Speech, and Signal Processing (1992 Conf. Proc.). IEEE. BKIE4. $292.00.
- ☐ Acoustics, Speech, and Signal Processing (1993 Conf. Proc.). IEEE. BKIE5. $292.00.
- ☐ Acoustics, Speech, and Signal Processing (1994 Conf. Proc.). IEEE. BKIE6. $304.00.
- ☐ The Wood Effect/Johnsen. BKMA1. $12.95.
- ☐ Sound, Structures, and Their Interaction/ Junger, Feit. ASA. BKMT3. $34.95.
- ☐ Acoustical Engineering/Olson. BKPA1. $49.95.
- ☐ SPICE for Circuits and Electronics Using PSPICE/Rashid. BKPH6. $33.95.
- ☐ Digital Signal Processing Experiments/Kamas. BKPH7. $34.95.
- ☐ Handbook for Sound Engineers/Ballou, ed. BKS28. $99.95.
- ☐ Reference Data for Engineers/Van Valkenburg, ed. 8E. BKS50. $99.95.
- ☐ The Theory and Design of Loudspeaker Enclosures/Benson. BKSA1. $24.95.
- ☐ Using Time Code in the Reel World II/ Tanenbaum, Klemme. BKSR1. $19.95.
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MINING FOR GOLD ON THE MADISOUND BBS

By William R. Hoffman

Madisound Speaker Components, a major US loudspeaker distributor and supplier, sponsors a computer bulletin board for speaker-building enthusiasts. Audiophiles with a modem and computer can upload or download program files related to many audio subjects, including loudspeaker system and crossover design (both passive and active), and much more. These files are available just for the phone time required to log on to the system and download the desired files. At the time of this writing, the program files listing contained nearly 270 entries, with the majority covering speaker system design.

One of the problems with this particular bulletin board (a BBS in computer lingo) is its lack of organization. Most programs include only a brief half-line description of their content. For shorter files (fewer than 50K bytes), the cost for phone-line time is low; therefore, to guess what a program does and download it involves little financial risk. On the other hand, to download the files which extend to over 300K bytes (even with a fast 9,600 baud modem) could involve substantial telephone charges.

ACCESSING THE BBS

For a glimpse of what's available, I have conducted a survey of these programs and sorted them by subject area:

1. Enclosure design software (51 files),
2. Passive crossover design programs (12 files),
3. Active crossover design programs (7 files),
4. Room acoustics programs (5 files),
5. Driver data files (4 files),
6. Demonstration programs (7 files),
7. Miscellaneous programs (7 files).

From this file listing, I chose and researched some of the best and most popular programs to help both those experienced *SB* readers looking for particular software and the burgeoning audience of speaker building "rookies."

For the benefit of the novice speaker builder with a personal computer and modem but little or no experience with a bulletin board, I've included a quick course in accessing the Madisound BBS (see "Using the Madisound BBS" sidebar).

Also, note that a good, fast modem (at least 2,400 baud) is an asset, because it will reduce phone-line charges. In addition, the BBS supports the Z-modem transfer protocol, which is a useful feature if you are thinking of buying a modem (or FAX/modem) program.

OVERVIEW

I begin with three general comments about these programs. First, many of them are similar in function. This should not discourage you from writing and submitting new programs to the BBS, however; but know that there are only so many ways to implement a simple speaker-enclosure design program. The dozen or so most commonly used equations for calculating box volume, Q_{TC}, for example, are very easily coded into a program, and I, among others, have done so and published the results.[1,2] Therefore, use the descriptions provided in the program categories to help you to choose.

Second, the individual programmers' basic math skills and knowledge of the subject matter determines the programs' accuracy. As the end user, you have no control over this, so when you use one of these programs for the first time, it is a good idea to check your results against other programs or similar projects published in *SB* or other sources.

Third, there is an old saying in the computer business: "GIGO" (garbage in, garbage out), or use caution when entering information into the program. Be sure to enter the correct values for the driver, inductor, or capacitor you use. Too often the product is not what its manufacturer claims. I have measured some wild, sample-to-sample parameter variations from some very reputable loudspeaker driver manufacturers: for instance, variations in V_{AS} of ±40% from the manufacturer's specs, and F_O values that range over ±20%. Some enclosure designs will not tolerate these discrepancies, so beware.

Finally, all of these programs have some limitations, most of which are not spelled out by their authors in the program's information. If you use them to design a project that is unusual or nonstandard, they may produce erroneous results. Therefore, if you are unfamiliar with a particular program's design or its limitations, be careful how you use it.

PROGRAMS

Enclosure Design Software. This is the largest program group on the BBS, with 51 program file choices. The novice speaker builder seeking a simple, all-purpose program for a box design can select AUDIOLAB.ZIP, ENCVOL.ZIP, LISTBOX.BAS, or VBOX.BAS. In addition, programs such as 2WAYSYS.ZIP, AVI.EXE, and FILTBOX.ZIP offer both enclosure and simple crossover designs.

The more experienced user can explore AV10.ZIP, which provides the dimensions of cylindrical- or spherical-shaped enclosures, or BANDBOX.ZIP, BPASS.WK1, SPKRWKSP.ZIP, and SYMSPDR.ZIP, which offer bandpass and other forms of compound system design. For highly specialized loudspeaker system design work, try HORN.ZIP as an aid to horn system design and TLINE.ZIP for transmission lines.

Files for more advanced design work include BOXLOSS.BAS to calculate vent-

ABOUT THE AUTHOR

William R. Hoffman was born in 1945 and grew up in northern California. He has two general arts college degrees (1969 and 1989) and is currently pursuing a BSEE degree. He was a teaching assistant in electronics at the University of Nevada from 1981 to 1989. During these years, he was active in the high-fidelity industry, first as corporate vice president of engineering for Advanced Acoustics/Design, Ltd. (1975-77), then as chief engineer of AudioGraph Engineering Co. (1976-78), both of Reno, NV. Most recently he has been a project engineer and manager of customer services for Threshold Corp., Sacramento, CA.

Users with a computer and modem can download a list of the audio-related files (including filename, size, and description) offered on the Madisound BBS by dialing (608) 836-9473.

ed box losses, HELM201.ZIP (and HELMHZ. ZIP) to compute duct sizes and lengths for tuning, NCLOSURE and PORTNOTE for notes and information on vented system designs, and VARIO.BAS (and VARI.BAS) for sample designs using the Dynaudio Variovents as part of an enclosure system.

Finally, the multiple file "Master" programs contain within their 100K–300K bytes of information many similar program files. Some of these are very elaborate compilations of design programs from a single source. For instance, JBL.ZIP contains many duplicate files found elsewhere on the Madisound BBS (see the file CONTENTS for a listing of those programs), but if you have the time and a speedy modem, try this one. JBL.ZIP is a genuine gold mine of information on JBL-manufactured products and contains another program file, NEWBOX.BAS (also listed separately), which I describe later.

PERF450.ZIP deserves special mention. It is a good sealed/vented system design program, with graphic output for a printer. It also includes a useful program called ROOM to help you position your loudspeakers in a room for best sound (not available as a separate BBS file). I will review this program later as well.

Passive Crossover Design Programs. These 12 deal with either passive crossover design or with related subjects. If you are a beginning designer, buy Vance Dickason's book, *The Loudspeaker Design Cookbook*, 4th ed. (available from Old Colony Sound Lab), and read the section on crossovers and crossover design (Chapter 7). Many more advanced *SB* readers might also find it very informative.

These programs could not begin to cover all the factors of a complete crossover design, particularly a complex one. Passive crossover networks by their very nature are imperfect. The program can be considered complete only when its design is made part of a mathematical model, which includes information on the driver characteristics.

To begin a survey of passive crossover design files, let's first look at some simple programs. Beginners should start with 2WAYXOVR.ZIP, 3WAYXOVR.ZIP, or XOVRTOLS.ZIP, which is a short collection of simple but useful programs. Advanced users might try the more elaborate, detailed TWOX.ZIP or THREEX.ZIP.

For those extra functions in a crossover design, try L-PAD.ZIP and PADS, which help design a fixed passive driver attenuation network, and NOTCH1.ZIP for creating a notch filter to eliminate peaks in a driver's response.

Active Crossover Design Programs. Some of these files can aid you in designing elaborate filters for active crossover design. If you are experimenting with simple op-amp-based active filters, the program ACTCROSS.BAS is a good place to start.

As you progress, try FILTER.ZIP for a complete circuit diagram of a filter. Finally, program files such as FILTERCAD.ZIP, FILTER.EXE, FILTER.LZH, and FILTR170.ZIP are very elaborate and presume a background in filter design, so be forewarned.

To the best of my knowledge, none of these program files integrate driver data directly into their calculations. If you build a filter and apply it to a loudspeaker system, it will not necessarily give you any better results than using an equivalent function passive-type crossover.

Room Acoustics Programs. These five program files are each very interesting. The first three, BOUNDARY.ZIP, BOUNDRY.ZIP, and BSTPLACE.ZIP, were written by Roy Allison (formerly of Acoustic Research and Allison Acoustics and now with RDL Acoustics in Bellingham, MA).[3] All three are also available from RDL, (800) 227-0390, along with reprints of some of Mr. Allison's research papers. The cost for the entire package is $5. These programs help you find the best place in your listening room to position loudspeakers to minimize the negative effects of room boundary reflections on bass and midbass response.

The program ROOM in the PERF450.ZIP file is graphics-based and shows the low-frequency room standing wave modes for many loudspeaker/listener positions. To use it, move the speaker site and listener seating positions around the graphics display until they match those in your listening room. The program calculates and displays the distribution of all the room modes. I highly recommend ROOM and its companion program PERFECT BOX.

Driver Data Files. In these four, several driver manufacturers' products (Allison, JBL, Audax/Polydax, and Pyle) and their pertinent specifications (usually the Thiele/Small values) are listed. Also, many list suggestions for driver use (box volume, tuning, crossover frequency, and so forth). However, many of these files offer only older-product specifications, which are useful if you are designing a speaker system around noncurrent drivers.

Demonstration Programs. These seven files are demonstrations (or product advertisements) for loudspeaker design instrumentation or software programs. Some other demo programs scattered among the other software files I surveyed are usable and therefore can be considered a working product. All are quite interesting and informative to help you plan your purchases.

Miscellaneous Programs. These offer many useful functions that are peripherally related to speakers, amplifiers, and music. AUDIO.ZIP, AUDIOF10.ZIP, SONGBS20.ZIP, and SOUNDDZ22.ZIP are database programs for sorting and storing information on record/CD/tape collections. In addition, TAPEMAKR.ZIP catalogs cassette collections and creates labels for them on a printer. Lastly, TF.ZIP allows you to use a PC as a programmable frequency generator.

THERE'S MORE

After you try some of these simple design programs, you might require something even more powerful, but where do you go? The full-featured, commercial software packages are the next stop. They usually contain printed guidebooks, manuals, and a complete set of integrated software tools, and are able to handle complete loudspeaker system design.

They can be expensive, however, and most require some technical expertise. You can't "design something with the push of a button." (See the sidebar "Commercial Software Programs" for a listing of some of the more popular packages and their publishers. Old Colony Sound Lab is probably the

USING THE MADISOUND BBS

Accessing this audio-oriented bulletin board system is simple and free if you have a computer with a modem. Currently about 270 software files are available for only the telephone line cost. Just follow the steps below:

1. Dial the Madisound BBS, (608) 836-9473.
2. A welcome screen will appear.
3. Answer the questions about who you are and select and enter a private code that only you will know. Remember this number, as you need to repeat your name and this code number each time you log on to the system.
4. Follow the system prompts until you reach the first menu. Enter the letter "F" (to see files under the viewpoint heading).
5. At the next prompt, enter the letter "D" (to download from the exchange heading).
6. At the prompt for filename(s), enter each file you wish to download. After entering the last filename, hit the Enter key a second time. This will end the entry procedure and start the downloading process.—**WH**

COMMERCIAL SOFTWARE PROGRAMS

In addition to the above bargain programs, there are a number of excellent, full-feature, commercial packages that include printed documentation and GUIs (graphical user interfaces) to simplify complex speaker design tasks. The following is a partial list of these programs and shopping resources.

PROGRAM: BASS BOX/X-OVER 2.0
PUBLISHER: Harris Technologies
PO Box 622, Edwardsburg, MI 49112
(616) 641-5924

PROGRAM: BOX CALC/NETCALC
PUBLISHER: Ariba
Baroniehoeve 32
5244 Hz Rosmalen
The Netherlands

PROGRAM: LEAP
PUBLISHER: Linear-X Systems, Inc.
7556 SW Bridgeport Rd., Portland, OR 97224
(503) 620-3044

PROGRAM: SPEAKER SYSTEM DESIGNER
PUBLISHER: ME Technologies
PO Box 50
Dyens Crossing, NSW
2429 Australia
(Distributed in the US by Old Colony)

PROGRAM: TOP BOX
DISTRIBUTORS: Zalytron Industries Corp.
469 Jericho Turnpike, Mineola, NY 11501
(516) 747-3515

A&S Speakers
3170 23rd St., San Francisco, CA 94110
(415) 641-4573

The following is an inexpensive program devoted to transmission line speaker design:

PROGRAM: QUICK & EASY TRANSMISSION LINE DESIGN (Q&ETLD)
PUBLISHER: Mahogany Sound
2610 Schillingers Rd. #448, Mobile, AL 36695
(205) 633-2054

The following are sources for a number of specialty design programs. You can call them to order a catalog:

Old Colony Sound Lab
PO Box 243, Peterborough, NH 03458
(603) 924-6371

OCSL current offerings include:
MODES FOR YOUR ABODES
BOXRESPONSE
BANDPASS BOXMODEL
CALSOD (standard and professional)
PC-ECAP
FITDUCT
LDCAD
LMP (Loudspeaker Modeling Program)
BOXMODEL
NON-OPTIMUM VENTED BOX SPREADSHEET
PASSIVE CROSSOVER CAD
ROOM DESIGN POWERSHEET
LOUDSPEAKER DESIGN POWERSHEET
PXO PASSIVE CROSSOVER CAD
QUICK BOX
Q&ETLD
SPEAKER DESIGNER
SPEAKER SYSTEM DESIGNER
TRANSMISSION LINE BOXMODEL
THE LISTENING ROOM
TERM LOUDSPEAKER DEVELOPMENT
TWO-WAY ACTIVE CROSSOVER PLUS
VENTWRK VENTED BOX CONSTRUCTION (PLUS)

Sitting Duck Software
PO Box 130, Venetta, OR 97487
(503) 935-3982

largest distributor of these programs and has an excellent catalog.)

AND THE WINNER IS...

I have offered general comments on the content and usefulness of the various programs in our subject listings, but now will make recommendations to the amateur speaker builder. These are based on my own experience as a professional system design engineer and a home experimenter.

1. When I perform an initial search, say, for a bass driver/enclosure combination for a particular project, I begin by using the program PERFECT BOX. I load the Thiele/Small parameters of a preselected group of drivers into its database file. I then use the program's quick, easy menu system and good graphing capability to determine the basic enclosure type and parameters for each of these drivers. From there, it is a simple task to select a driver that will meet the needs of the project. Especially useful is the electrical power and SPL limiting graph the program provides.

More amateur speaker builders should see this particular graph. It is often a revelation and a shock to see what happens to the bass response of your favorite speaker system (which may have very flat, extended bass at low power levels) when it is heavily driven. Many vented systems have poor design and no headroom (the driver quickly loses radiated power capability over some part of the bass range, causing severe volume compression).

2. If I find one (or more) promising driver/enclosure combinations (with a vented design), I use D.B. Keele's program NEWBOX, which provides a more accurate look at the system characteristics than PERFECT BOX. It also offers a more complete set of working tools to optimize the system design.

3. Finally, for a project that requires sophisticated, integrated system design, I use a professional, PC-based program package called CALSOD (also available from Old Colony). From driver-enclosure matching to optimized crossover values, this program can do it all. Its ability to vary many parameters in a particular design and to predict the results almost immediately are its greatest assets. On the down side, you need a technical background to use this program.

THE END

It was impossible for me to download the hundreds of design program files available

to page 73

SOURCES

Madisound Speaker Components
PO Box 44283
Madison, WI 53744
(608) 831-3433, FAX (608) 831-3771

REFERENCES

1. W.R. Hoffman, "Design Your Own Loudspeakers," *Popular Electronics* (Feb. 1994). Also reprinted in *Electronics Hobbyists* handbook, fall 1994.

2. W.R. Hoffman, "Design Your Own Loudspeakers, Part 2," *Popular Electronics* (scheduled for July 1995 issue).

3. R. Allison, "The Best Place For Your Speakers? Your Computer Knows!," *Audio* (Aug. 1994).

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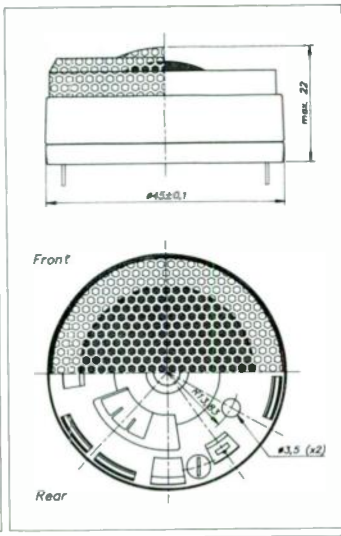
NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268-5)	30	W
FREQUENCY RANGE	2,5–20	kHz
SENSITIVITY (1W, 1m)	88,5	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	1,3	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	1800	Hz
MOVING MASS (incl. air)	0,27	g
FORCE FACTOR, B x l	2,5	Txm
MAGNET WEIGHT (0,4oz)	11	g



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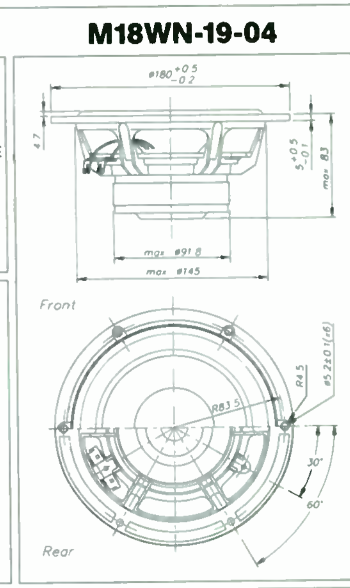
- STIFF MOISTURE RESISTANT PAPER CONE
- LOW DAMPING RUBBER SURROUND
- KAPTON® V.C. FORMER
- OPTIMIZED AS CAR-WOOFER

NOMINAL IMPEDANCE	4	Ω
VOICE COIL RESISTANCE	3.0	Ω
NOMINAL POWER (IEC 268-5)	110	W [1]
SHORT TERM MAX. POWER (IEC 268-5)	300	W [2]
LONG TERM MAX. POWER (IEC 268-5)	150	W [3]
OPERATING POWER	12.6	W
SENSITIVITY (1W, 1m) / (2.83V, 1m)	85/88	dB
FREQUENCY RANGE	63-3000	Hz
FREE AIR RESONANCE	63	Hz
VOICE COIL DIAMETER	40	mm
VOICE COIL HEIGHT	14	mm
AIR GAP HEIGHT	6	mm
VOICE COIL INDUCTANCE	0.6	mH
EFFECTIVE CONE AREA	132	cm²
MOVING MASS (incl. air)	15.5	g
MAGNET WEIGHT (14.6 oz)	415	g
FORCE FACTOR, Bxl	4.8	Txm
Vas	10	ltr
Qms	3.88	
Qes	0.80	
Qts	0.66	



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0, 30 and 60 deg.

Input: 2.83 V
Mic.: 1 m

320 ltr. box mounted in infinite baffle.

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Half space free-field above 100 Hz
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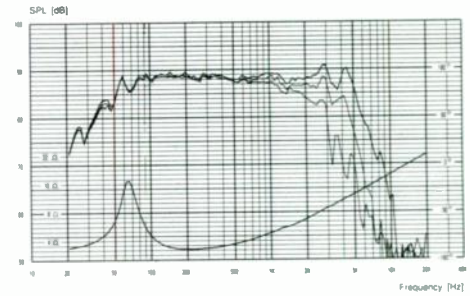

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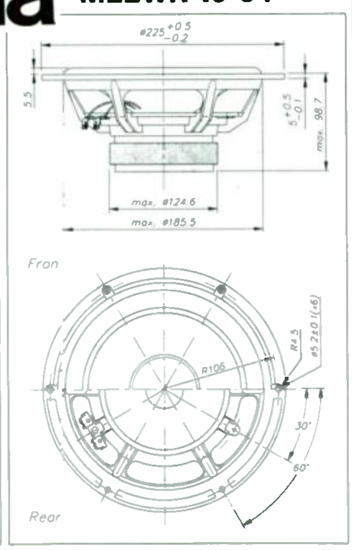
- VERY STIFF PAPER CONE
- LOW DAMPING RUBBER SURROUND
- MAGNET SYSTEM WITH 2" V.C. AND LARGE VENT
- OPTIMIZED AS CAR SUBWOOFER

NOMINAL IMPEDANCE	4	Ω
VOICE COIL RESISTANCE	3.2	Ω
NOMINAL POWER (IEC 268-5)	150	W [1]
SHORT TERM MAX. POWER (IEC 268-5)	500	W [2]
LONG TERM MAX. POWER (IEC 268-5)	350	W [3]
MUSIC POWER (DIN 45500)	200	W
OPERATING POWER	12.6	W
SENSITIVITY (1W, 1m) / (2.83V, 1m)	85/88	dB
FREQUENCY RANGE	45-500	Hz
FREE AIR RESONANCE	48	Hz
VOICE COIL DIAMETER	50	mm
VOICE COIL HEIGHT	19	mm
AIR GAP HEIGHT	8	mm
VOICE COIL INDUCTANCE	0.9	mH
EFFECTIVE CONE AREA	224	cm²
MOVING MASS (incl. air)	35	g
MAGNET WEIGHT (37.4 oz)	1062	g
FORCE FACTOR, Bxl	6.4	Txm
Vas	20	ltr
Qms	3.44	
Qes	0.87	
Qts	0.70	



M22WR-19-04

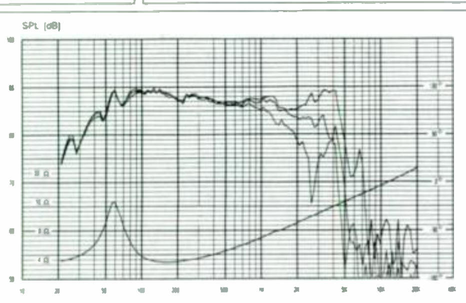
DISPERSION PLOT
0, 30 and 60 deg.

Input: 2.83 V
Mic.: 1 m

320 ltr. box mounted in infinite baffle.

18.04.94 KK

Anechoic room 8x7x8 meters
Half space free-field above 100 Hz
Brüel & Kjær 2012 Audio Analyzer



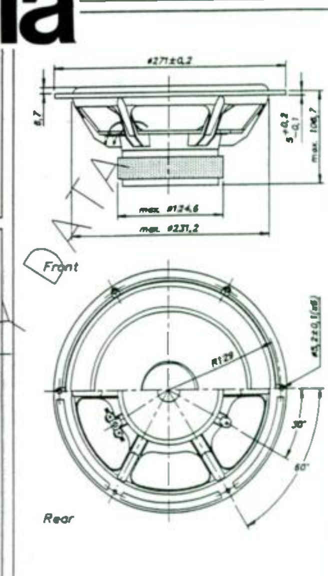
10" WOOFER vifa M26WR-19-04

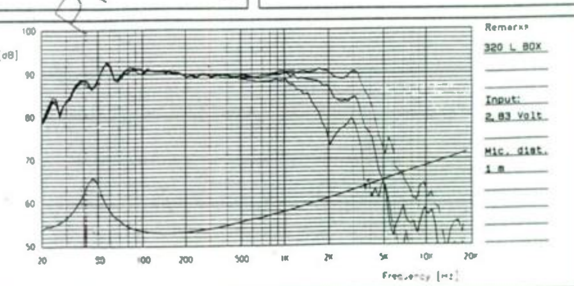
Special Features:

- Long Stroke, Low Reflexion, Magnesium Basket
- Very Stiff Paper Cone
- Low Damping Rubber Surround
- Magnet system with 2" VC and Large Vent
- Optimized as Car Subwoofer

Nominal Impedance [ohm]	4
Nominal Power (IEC 268-5) [W]	150
Music Power (DIN 45500) [W]	200
Frequency Range [Hz]	40-1000
Sensitivity (1W,1m)/(2,83V,1m) [dB]	87/90
Effective Cone Area [sq.cm]	337
Voice Coil Resistance [ohm]	3,2
Voice Coil Inductance [mH]	0,95
Operating Power [W]	8,1
Voice Coil Diameter/Height [mm]	50/19
Air Gap Height [mm]	8
Moving Mass (incl. Air) [g]	41
Free Air Resonance [Hz]	44
Force Factor, B*l [T*m]	6,4
Magnet Weight [g]/[oz]	1062/37,4
Qms	3,65
Qes	0,84
Qts	0,69
Vas [ltr]	53

PRELIMINARY DATA





CLIO CAN DO:
- FFT of input signal with THD analyzer
- THD with burst signals
- Electrical and acoustical phase
- Anechoic transfer function
- MLS analysis of linear systems
- System impulse response
- Energy-time curve
- Cumulative spectral decay
- Sinusoidal steady state transfer function
- Near-field and inside-box response
- Impedance function
- Thiel-Small parameters measurement
- Real time 1/3 octave analysis
- Generation of pre-defined waveforms
- Time signal (oscilloscope)
- SPL, dBV, dBm, V amplitude
- Inductor and capacitor meter

GENERATOR
- Two channels 16 bit sigma-delta D/A conv.
- Frequency range: 1 Hz - 22 KHz (+0, -1 dB)
- Frequency accuracy: better than 0.01 %
- Frequency resolution: 0.01 Hz
- Output impedance: 100 Ω
- Maxi output (sine): 12 dBm (3.1 V RMS)
- Attenuation: 64 x 1.5 dB step + mute
- THD + Noise (sine): 0.015 %

ANALYZER
- Two channels 16 bit sigma-delta A/D conv.
- Input range: + 30, - 42 dBV
- Input impedance: 64 KΩ (2.7 KΩ mic. input)
- Phantom: 8.2 V

MISCELLANEOUS
- Sampling frequency: 51.2 KHz, 6.4 KHz
- Card: 14 cm, 8 bit PC slot card

MICROPHONE MC-01:
- Condenser electret
- +/- 1 dB, 20 Hz to 10 KHZ.
- +/- 2 dB, 10 KHz to 20 KHz (direct field)
- Maximum level: 120 dB SPL
- Dimensions: 8 mm diameter, 250 mm long
- Accessories: 3 m (10 ft) cable, stand adaptor

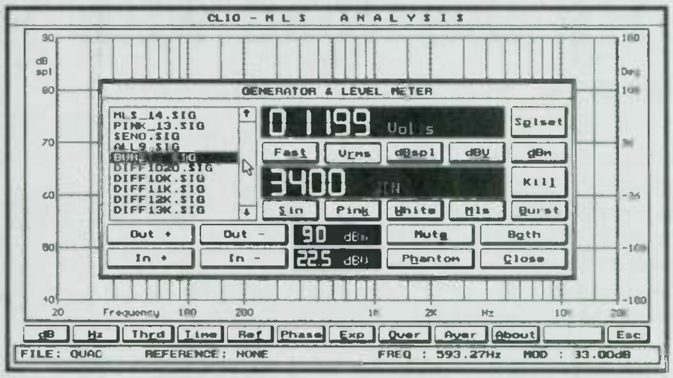
PC HOST MINIMUM REQUIREMENTS:
- 80286 or better
- One 8 bit half size free slot
- EGA video board or better
- 640 KBytes RAM
- Hard disk
- No co-processor needed

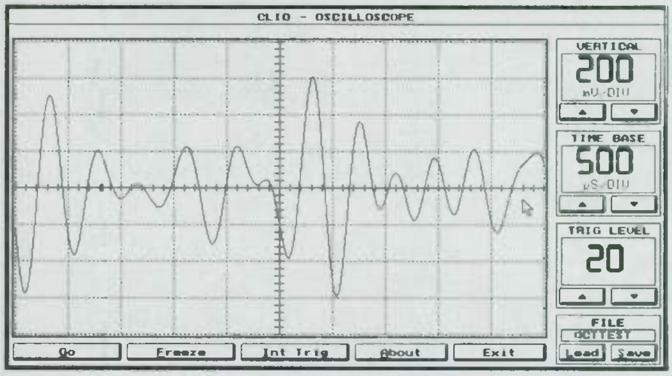
FREE DEMO DISK FROM ORCA UPON REQUEST.

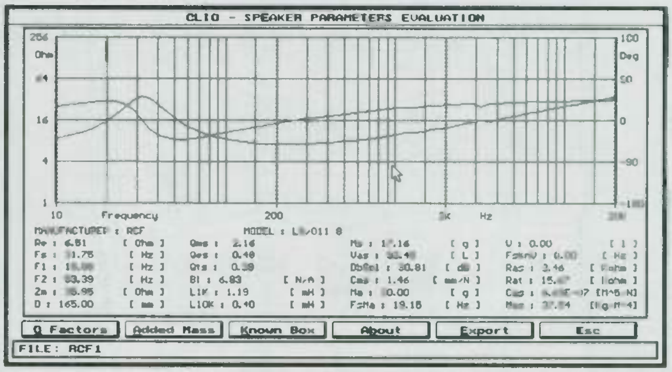
Most commonly encountered US price for the complete hardware/software CLIO system: US$ 1,595.00. This price includes, in the USA, one hour of direct technical support for registered users.

OEMs, importers and large organizations, please contact ORCA directly for special industry accomodation.

COMING SOON: automatic programmable PASS/FAIL software module for production QC.







CLIO, another ORCA offering

ORCA is happy to bring you the new CLIO system for its exceptional value and ease of use. In our opinion it is the first fully assembled professional quality MLS based system affordable to DIY amateurs.

ORCA Design & Manufacturing Corp.
1531 Lookout Drive. Agoura, CA 91301 - USA
tel: (818) 707 1629 fax: (818) 991 3072

STOCKING DISTRIBUTORS

BIG COVE Huntsville, AL. FAX (205) 430 3961 TEL (205) 379 3982
MADISOUND Middleton, WI. FAX (608) 831 3771 TEL (608) 831 3433
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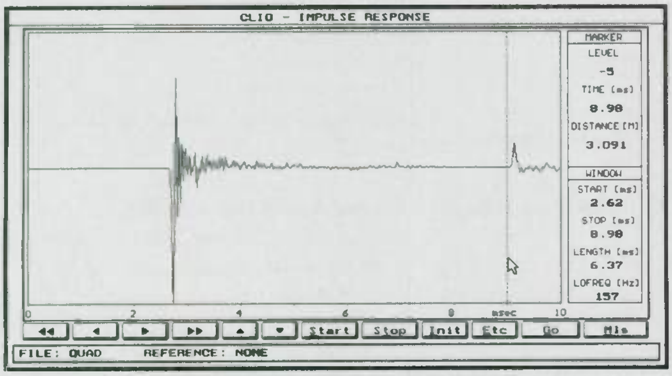
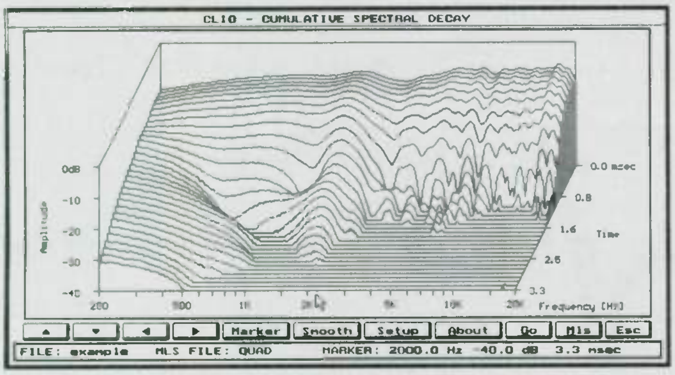
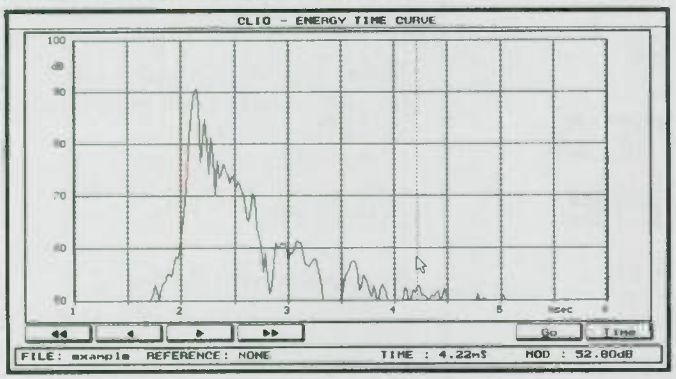
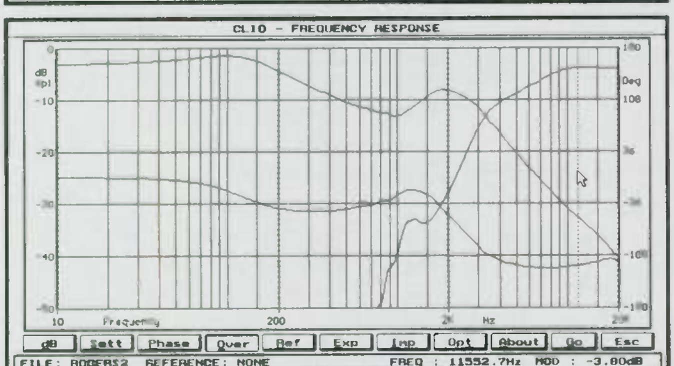
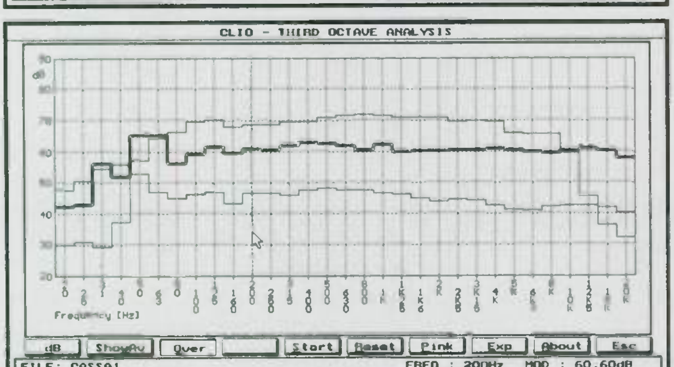
ZALYTRON Mineola, NY. FAX (516) 294 1943 TEL (516) 7473515



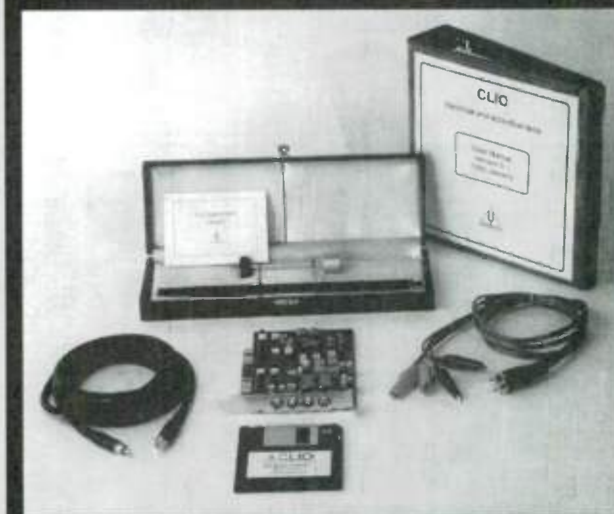








CLIO



CLIO is a complete 16 bit dual I/O channels electro-acoustic measurement system, with a variable sampling rate, built on a half size PC board ! It comes complete with a precision calibrated condenser microphone (with stand adaptor), cables and connectors, and a remarkably user-friendly manual including thorough explanations and clear illustrations.

CLIO works as a precision A/D & D/A audio front-end. It will generate any pre-defined stimulus such as sine, square, burst, noise and MLS (medium length sequence), as well as any others saved in its libraries. CLIO will also analyze input signals both in frequency and time domain. Install CLIO in your laptop and you have a professional portable measurement system to take with you in the field (ideal for in-car measurements). As an option, CLIO offers a battery-operated pre-amplifier with A-weighted filtering for when the microphone is more than 10 ft from your PC host (anechoic chamber).

CLIO software release 3.0 gives you a "Windows-like" mouse oriented user interface. The various control panels show on your screen as the real instrument. CLIO gives you powerful editing capabilities as well as import and export facilities (LEAP, CALSOD ...).

Reader Service #34

READING ROOM

Speaker Builder regularly receives audio-related literature—catalogs, newsletters, books, and other publications—covering products and topics of interest to *SB* readers. The following is a compilation of recent references and titles:

The Audio Adventure
A monthly newsletter that reviews affordable high-end audio products.
TOMART Publishing, Inc.
PO Box 15256, Chevy Chase, MD 20825
(800) 566-6617

The Binaural Source No. 3
List of CDs and cassettes especially designed for headphone listening.
The Binaural Source
Box 1727, Ross, CA 94957
(415) 457-9052

The Electrostatic Loudspeaker Design Cookbook
Reference book for building electrostatics by audio veteran Roger Sanders.
Old Colony Sound Lab
PO Box 243, Dept. B95,
Peterborough, NH 03458
(603) 924-6371/6526, FAX (603) 924-9467

Fair Radio Sales Catalog
Lists company's offerings of electronic surplus equipment.
Fair Radio Sales
1016 E. Eureka St., Lima, OH 45802
(419) 223-2196, FAX (419) 227-1313

Legacy Audio 1995 Catalog
48-page booklet profiling speakers, surround sound processors, and amplifiers.
Legacy Audio
3021 Sangamon, Springfield, IL 62702
(800) 283-4644, FAX (217) 744-7269

Mouser Catalog
276-page index to electronic components.
Mouser Electronics
2401 Hwy. 287 North, Mansfield, TX 76063
(817) 483-4422, (800) 992-9943

NetWell Noise Control Catalog
Twenty pages listing the company's line of acoustic-control products.
NetWell Noise Control
6125 Blue Circle Drive,
Minnetonka, MN 55343
(612) 939-9845, FAX (612) 939-9836

North Creek Music Systems—1995
Catalog of loudspeaker components and accessories for the hobbyist.
North Creek Music Systems
PO Box 1120, Old Forge, NY 13420
(315) 369-2500

Parts Express 1995 Catalog
Free 212-page catalog listing speakers and audio accessories, and semiconductors.
Parts Express
340 E. First St., Dayton, OH 45402
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RAG Electronics Catalog
A resource of used equipment and factory authorized new equipment.
RAG Electronics, Inc.
2450 Turquoise Circle,
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Audio accessories, including chassis, adapters, transformers, and amp designs.
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2100 Ward Dr.
Henderson, NV 89015
(702) 565-3400, FAX (702) 565-4828

We invite manufacturers and publishers to send releases for inclusion in this column to:
Speaker Builder
Reading Room
305 Union St., Peterborough, NH 03458
FAX (603) 924-9467

Wayland's Wood World

MOUNTING IRREGULARLY SHAPED DRIVERS

By Bob Wayland

Cutting a hole for a woofer or circular tweeter is not complicated when you have made the right jigs (*SB* 4/95, p. 40). Many tweeters, special-purpose speakers, and the ubiquitous 6×9 require irregularly shaped mountings. Although not difficult to make, these holes require their own special jigs, one for each size and shape.

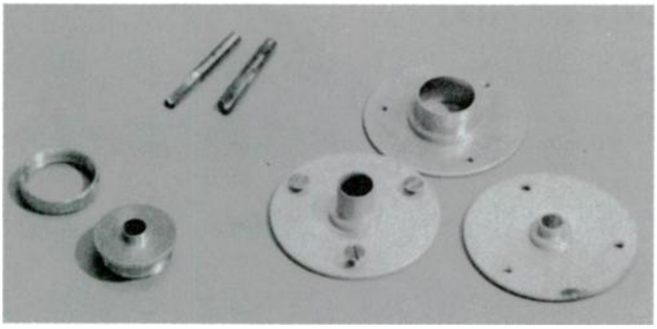


PHOTO 1: Guide bushings and router bits for template routing of irregularly shaped holes.

PREPARATION

A router is the main tool you'll be using. The best router bits for this operation are spiral upcut and straight bits (*Photo 1*). You will also require a round guide bushing, such as the one for Porter-Cable routers shown in *Photo 1*. If you experience difficulty finding one that fits your router, you can buy a universal base plate with bushings that you can adapt to your router.

The guide bushing rubs against a template guide, allowing you to rout out even the most complex holes. In this example we'll cut an insert around a hole for a midrange/tweeter driver so the face of the driver is flush with the mounting board. By routing through the board, you can cut a hole for an irregularly shaped speaker, e.g., a 6×9 oval. The usual material for making templates is 1/8″ Masonite™, but plywood, plastic, or even metal can be used with equally good results. I use whatever scraps are in my shop.

Don't commit the common mistake of making the template too small. The edges of the Masonite guide should be 3 or 4″ beyond the cutout in the template. This gives you a firm surface to rest your router while you are routing.

Attach the guide bushing to the base of your router. After you have locked the router bit in place, carefully center the guide bushing so the distance from the cutting edge of the router bit to the outside edge of the guide is exactly the same all the way around (*Photo 2*). Record this distance, which you will need when laying out the template.

Finally, you will need some double-sided tape, the kind used to hold down carpet and loose rugs and available at most hardware stores. This carpet tape is about the right width and offers great holding power.

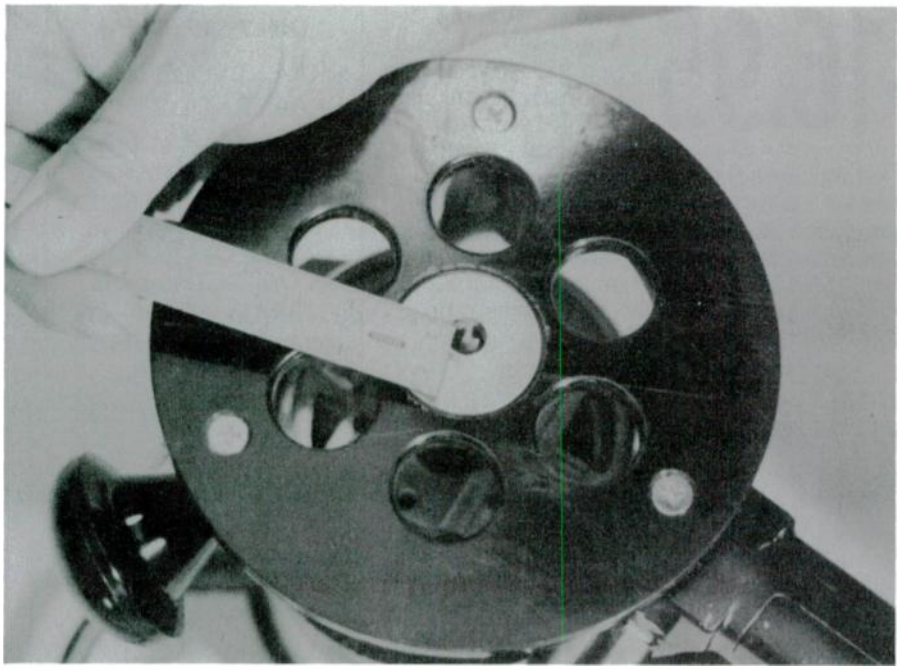


PHOTO 2: Measuring the distance from the cutting edge of the router bit to the outside edge of the guide.

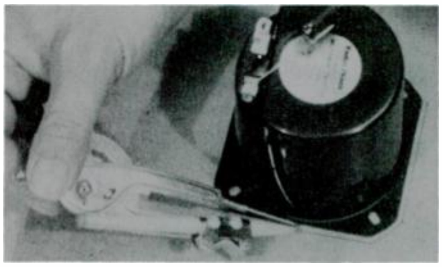


PHOTO 3: Tracing the correct pattern by carefully following the edge of the speaker with the pivot point and letting the pencil mark the right-sized pattern for the opening.

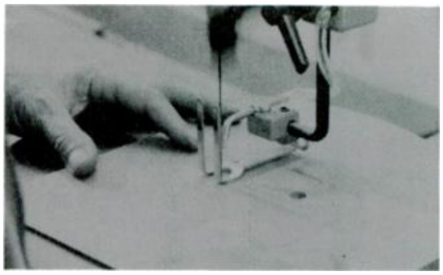


PHOTO 4: Using a scroll saw to remove the cutout for a template.

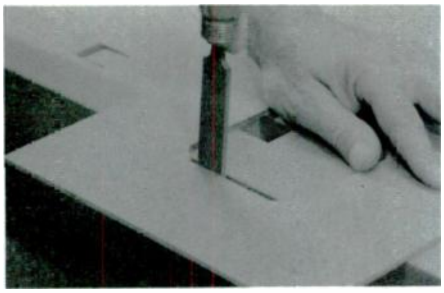


PHOTO 5: A mill file removes the irregularities on the edges of the cutout.

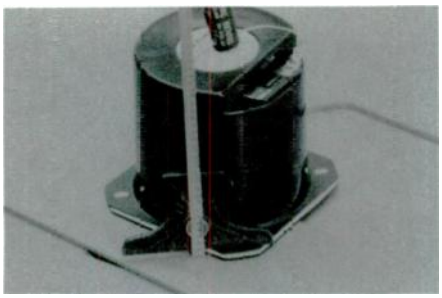


PHOTO 6: Measuring the amount of offset to the top of the flange of a speaker set on its gasket on a flat surface.

plate to trace the correct pattern. Another way is to trace the pattern on the back side of the template. The problem with Masonite, however, is that its back side is usually quite rough, making it difficult to trace accurately.

Remove the marked-out pattern, either with a scroll saw or a coping saw. I usually cut to the inside of the line, so when the edge is straightened, the pattern is the right size. Be careful to keep the blade perpendicular to the template, or, to your work table (*Photo 4*), if you are using a scroll saw. Either way, you'll need to fix the irregularities of the pattern edges. You can use a mill file (*Photo 5*)—a half-round file for rounded corners and a round file for tight-radius situations. While you are at it, remove all burrs. Apply the carpet tape, covering as much of the back side of the template as you easily can. Don't remove the backing from the tape until just before you're ready to use it.

SETTING AND CUTTING

If you are cutting a trough for the speaker flange to fit into, you must determine how deep to make it. If you are making a through-cut for a speaker, you can skip this step. Place the speaker on its gasket on a flat surface and measure the amount of offset to the top of the flange (*Photo 6*). If you're using a silicone seal instead of a gasket, add about 1/16" to the thickness of the flange, or make a spacer the thickness of the silicone and measure. With a depth gauge, set your router depth to this premeasured amount (*Photo 7*).

Now position the template over the area to be routed. Remove the backing from the double-sided tape on the back of the template (*Photo 8*). Clean the surface of the mounting board before positioning the template. Holding the template just above where you'll mount the speaker, carefully center the opening and then firmly press the template onto the mounting board.

With this preparation work completed, you're now ready to cut. If you have a normal router (not a plunge router), place the bit inside the hole in the template. Tip the router up so the bit isn't touching the surface of the board, turn the router on, and then slowly

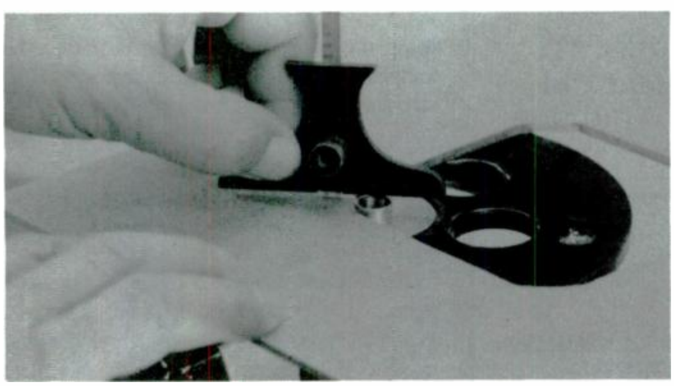


PHOTO 7: Using a depth gauge to set the router depth to the amount measured in *Photo 6*.

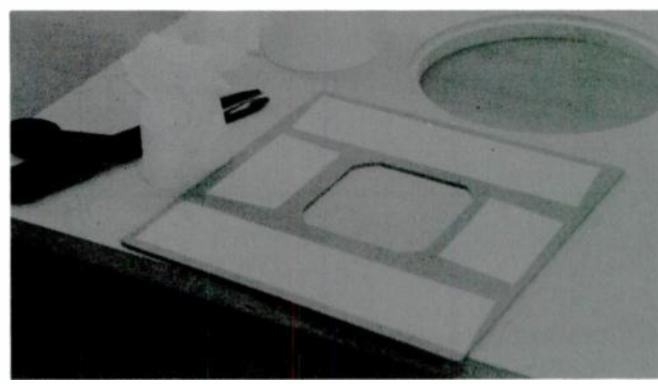


PHOTO 8: Removing the backing from the double-sided tape on the back of the template.

MAKING THE TEMPLATE

Next, we'll determine and trace the opening to be cut onto the template. The tracing must allow for the presence of the guide bushing. One way is to set the compass the same distance from the center pivot point to the pencil at the measurement you made when centering the guide bushing. Carefully trace the correct pattern by following the edge of the speaker with the pivot point and letting the pencil mark the right-sized pattern for the opening (*Photo 3*).

Usually, with symmetry in the pattern of the flange or the opening for the speaker, you can simply place the speaker face down on the template and trace around the edge. If the irregular shape prevents you from doing this, first make a pattern on scrap cardboard by placing the speaker face down, tracing it, then cutting out the pattern and turning it over. You can then use this new pattern as a tem-

PHOTO 9: Moving the router around until the entire surface of the hole has been covered.

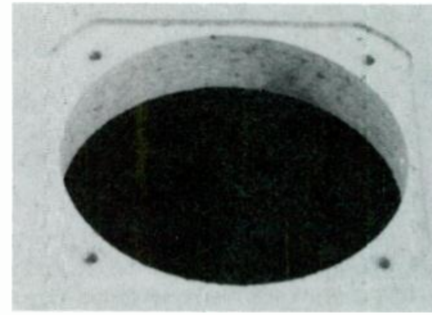


PHOTO 10: Finished hole.

PHOTO 11: An accurate flush-mounted speaker.

lower the router until its base is flat on the template. If you have a plunge router, this step simply involves lowering the router bit, once you have set it in place on the template.

Move the router around until the entire surface of the hole has been covered (*Photo 9*). Be sure the bushing guide rides firmly against the edge of the template.

If you are making a through hole, you need to cut only the edges about 1/8" in each pass. When the through hole is very large, you may need to put a scrap piece of the template material in the center to give the router base good support. If you are routing a trough for a flange, you must cut a through hole for the body of the speaker. I discussed a number of methods in the last column.

The completed hole is ready (*Photo 10*) and should make an accurate flush-mount for the speaker (*Photo 11*).

PREVIEW
Audio Amateur

Issue 2, 1995

- Instrumentation Preamp
- 6V Supply for RS CD3400
- Regulators for High-Performance Audio, Part 2
- Toward an All-Digital Power Amp
- Assemblage DAC-1 Kit Report

Software Report

CLIO TEST SYSTEM

Reviewed by Jean-Claude Gaertner

CLIO, Audiomatica s.r.l., Via Faentina 244/G, 50133 Florence, Italy, +39-55-5000402, FAX +39-55-575221. ORCA (1531 Lookout Dr., Agoura, CA 91301, (818) 707-1629, FAX (818) 991-3072) is the sole distributor of CLIO for the US, Canada, Mexico, Australia, and New Zealand. System requirements: IBM PC with a 286 processor or better; one 8- or 16-bit, half-size ISA slot; 640K RAM hard disk; EGA or better video card; DOS 3.2 or later. A math coprocessor is not required.

Depending upon the options you order, the CLIO system consists of the following components:

- HR2000 PC board;
- MIC-01 or MIC-02 microphone;
- 3381/A microphone preamplifier (optional).

A 3.5″ disk with the CLIO software, a user's manual, and cables are included in the package.

THE BOARD

The HR2000 is a half-size PC card, which fits the 8-bit ISA slot in PC-compatible computers to perform all the A/D and D/A interfacing. The rear bracket offers four RCA plugs: two for output and two for input. *Table 1* lists the board's technical specifications.

The cleverly engineered HR2000 needs only nine ICs to perform all its functions. Those located near the ISA connector provide the necessary interface with the PC bus. The two-channel, 16-bit A/D and D/A converter is from Crystal Company. A couple of TL-072 op amps and relays are included.

The two jumpers are labeled JP1 and JP2. Only the latter should be set by the user; the former should remain factory configured. As with almost every PC board, you must select an I/O address. With JP2, you can choose either a 300 or 310Hex address. Be sure to take already installed boards into account, so as to avoid any conflicts. (A wise practice is to write down the I/O address, IRQ, and DMA settings used by each card in a system.) Even though these addresses were originally left by IBM for prototype cards, I would have appreciated more freedom of choice, especially since there are many "holes" in the I/O ports, ranging from 200 to 3FFHex (see sidebar).

The board requires two Direct Memory Access channels, one for output generation and the other for acquisition. DMA saves circuitry and expensive SRAM, but, unfortunately, the two channels are potential sources of conflict with many scanners and sound boards.

The CLIO software does not yet utilize the following features: the HR2000 can simultaneously sample two channels at up to 51.2kHz; by using two D/A converters for signal generation, you can output whatever signals you wish through software. The extremely flexible board design thus provides room for future software expansion.

MICROPHONE OPTIONS

The MIC-01 and -02 microphones differ only in the lengths of their bodies: 25cm and 12cm, respectively. They come packaged with stand adaptors and calibration charts. The charts reflect only the claim frequency accuracy, but you get the sensitivity in mV/Pa.

Although the overall look of the microphones is very appealing, I have two minor gripes. I would like to see a reference part number imprinted on the body; I also find the asymmetrical output a little annoying, as this design limits the cable length from the mike to the input plug. The 3m mike cable (included with the system) represents the practical maximum length which can be used without too much trouble.

With my MLSSA system, I checked the MIC-01 against a $1,500 ACCO D7012 microphone. Both the frequency range and sensitivity of the MIC-01 were within the claimed specifications (*Table 2*).

3381/A MIKE PREAMP

This optional unit is designed to work with either MIC microphone. It provides the power and calibrates the gain, which is variable in 10dB increments from 70 to 120dB SPL, full scale. For the technical specifications, see *Table 3*.

The voltage at the output is calibrated to provide 1V RMS when a sinusoidal pressure applied at the microphone is equal to the

FIGURE 1: The main CLIO screen.

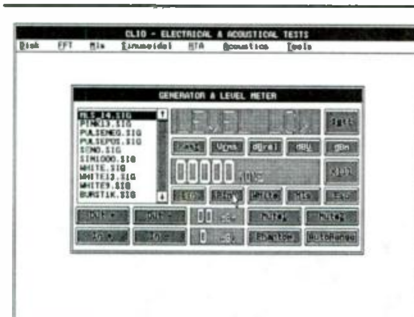


FIGURE 2: The Generator and Level Meter control panel.

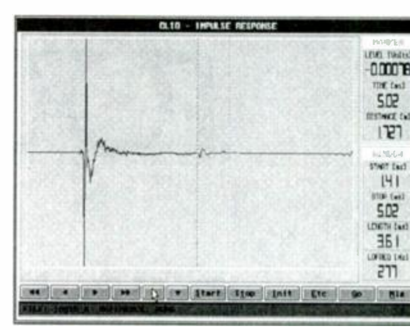


FIGURE 3: Two-way satellite impulse response.

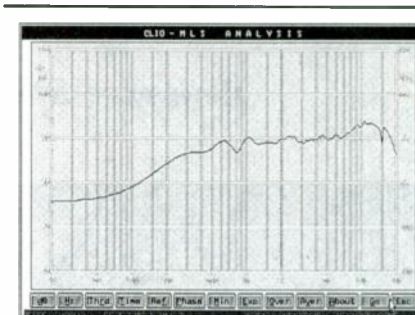


FIGURE 4: MLS frequency screen.

value of the selected scale. For instance, provided you have selected 100dB gain on the preamp, you will have 1V RMS at the output, if a 100dB pressure is in front of the mike.

The preamplifier is battery-powered, and you can check the charge state by pressing a test button. Of course, the AC charger will also power the preamp.

If you need more than 3m between the mike and the board, the preamp is required for accurate measurement. Although the "phantom" power supply for the mike is a bit low at about 3V, I think the 3381 is well-designed. I was rather disappointed with the cheap-looking cabinet and disliked the poor-quality Taiwanese AC main adaptor, which was glued and blocked the output voltage switch selector.

CLIO SOFTWARE

The software for computer measurement instruments is extremely important, since it is both the front-end of the dedicated measurement and the heart of the system. Audiomatica developed DOS-based CLIO with a Windows™-like, mouse-oriented user interface. During the past year, I have seen several revisions—each one an improvement. This indicates that Audiomatica is concerned with the quality of its product. Release 3.10 is the subject of this review.

Installation is fairly easy, thanks to a simple batch file which creates a main CLIOxx directory (xx being the current software release) with two subdirectories (\DATA and \SIGNAL). The program directory name is easily modified. Contrary to current software trends, CLIO's main program is rather compact, and installation requires less than 800K on the hard disk.

Audiomatica recommends a minimalist PC configuration, with Config.sys and Autoexec.bat files to minimize problems and ease troubleshooting. CLIO is not manufacturer guaranteed to run from Windows. Despite the warnings, I have successfully used it under some complex configurations, including DOS resident software, EMM386, QEMM 7.04, NETROOM 3.03, MS-DOS versions 5.0, 6.0, and 6.2, 4DOS 5.0 and 5.5 Rev B, and in a DOS session run from Windows for Workgroup 3.11. The hardware I used for this review included a cheap

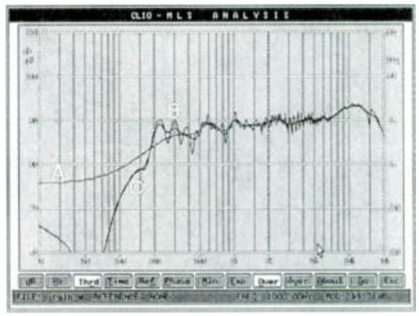


FIGURE 5: Curve presentation.

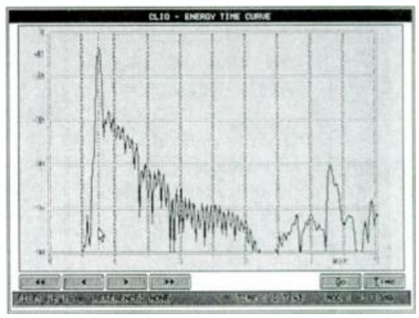


FIGURE 6: Energy time curve.

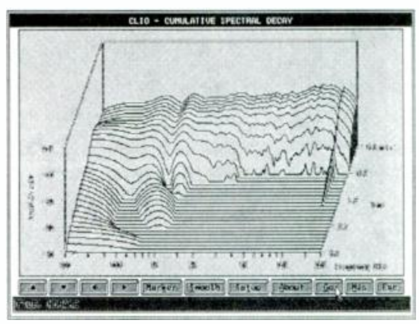


FIGURE 7: Cumulative Spectral Decay of the impulse response in *Fig. 4*.

Taiwanese VLB board with Eteq chips, an AMI Baby Voyager, and the Lexar LXM510 ISA-VLB-PCI.

You can usually start every measurement with either a one-key touch or a mouse click.

TABLE 1

HR2000 SPECIFICATIONS

GENERATOR	
Type	Two-channel, 16-bit sigma delta D/A converter
Frequency range	1Hz–22kHz (+0/–1dB)
Frequency accuracy	Better than 0.01%
Frequency resolution	0.01Hz
Output impedance	100Ω
Max. output level (sine)	3.1V RMS
Attenuation	64 × 1.5dB step + mute
THD + Noise (sine)	0.015%
ANALYZER	
Type	Two-channel, 16-bit sigma delta A/D converter
Input range	+30 to –42dBV
Input impedance	64kΩ (2.7kΩ when the mike input is selected)
Phantom	8.2V
MISCELLANEOUS	
Sampling frequency	51.2kHz, 12.8kHz, 3.2kHz
Card type	14cm, 8-bit PC slot card
Card connections	4 RCA plugs & adaptor cables to speaker terminal

The Import and Export facilities provided for ASCII text files are also understood by several other software packages, including CALSOD from Audiosoft and LinearX's LEAP. *Figure 1* is the main CLIO screen. If you get the "hardware not found" warning, check whether you have selected the proper I/O address in the Disk, Setup menu.

Several hot keys are available in nearly every window. The most frequently used is likely to be "F4." This brings up the Generator and Level Meter control panel (*Fig. 2*), which controls the signal generator, the level meter, the input gain, the output attenuation, and global and local settings for each measurement window. Some of the generator files appear on the left of the screen. The file MLS_14.SIG, for example, is used by the D/A converter to create the 14-order Maximum Length Sequence for the MLS window.

With PC instruments, it's fairly easy to implement self-calibration. Four separate calibration files (MLS, Sinusoidal, Real Time Analyzer, and Tools) are required if CLIO is to run properly. Run the calibration periodically to ensure the most accurate measurements. It takes only a few minutes and can save you a lot of time.

The calibration process in version 3.1 is far more robust than with previous CLIO releases. Nevertheless, it can be wrong at times. For example, if you don't follow the manual's advice and forget to connect output 1 to input 1, then start a calibration in the MLS menu, all further measurements will be incorrect without warning. Note that it is very easy to check a calibration's success: start a measure (with the loop still set) and look for a straight line in the frequency domain on both the amplitude and phase.

According to the user's manual, CLIO is capable of performing the following impressive range of analyses and measurements:

- FFT of input signal with THD analyzer
- THD with burst signals
- electrical and acoustical phase
- anechoic transfer function
- MLS analysis of linear systems
- system impulse response
- energy time curve
- cumulative spectral decay (i.e., waterfalls)
- sinusoidal steady-state transfer function
- near-field and inside-box sound pressure response
- impedance
- loudspeaker Thiele/Small parameters
- real-time 1/3-octave analysis

I/O ADDRESS MAP

200-207Hex	: Game port
278-27F	: Parallel printer port 2 or 3
2E0-2E7	: Serial port 4
2E8-2EF	: Serial port 3
2F8-2FF	: Serial port 2
300-31F	: Prototype card
378-37F	: Parallel printer port 3 or 2
380-38F	: SDLC, bisynchronous 2
3A0-3AF	: Bisynchronous 1
3B0-3B7	: Monochrome display
3BC-3BF	: Parallel printer port 1
3C0-3CF	: EGA/VGA display
3D0-3DF	: CGA display
3F0-3F7	: Floppy disk controller
3F8-3FF	: Serial port 1

System Hardware Interrupts

IRQ0	: Timer output
IRQ1	: Keyboard (output buffer full)
IRQ2	: Redirected to IRQ 9 (a second seven IRQ handler was introduced for the IBM-PC AT)
IRQ3	: Serial port 2 (this IRQ is usually shared with Serial port 4)
IRQ4	: Serial port 1 (this IRQ is usually shared with Serial port 3)
IRQ5	: Parallel printer port 2
IRQ6	: Floppy disk controller
IRQ7	: Parallel printer port 1
IRQ8	: Real-time clock interrupt
IRQ9	: Software redirected to IRQ2
IRQ10,11,12	: Reserved (usually free)
IRQ13	: Coprocessor
IRQ14	: Fixed disk controller
IRQ15	: Reserved (note that IRQ15 is now being used by the new EIDE controller allowing up to four IDE devices instead of two)

Direct Memory Access (DMA)

DMA 0–4 supports 8-bit data transfer, DMA 5–6 supports 16-bit data transfer.

DMA0	: Spare
DMA1	: Spare
DMA2	: Floppy disk controller
DMA3	: Spare
DMA4	: Used to cascade channels 0 to 3 to the processor
DMA5	: Spare
DMA6	: Spare
DMA7	: Spare

- generation of predefined waveforms
- oscilloscope
- SPL, dBV, dBm, voltage amplitude scale
- inductance and capacitance meter

Unfortunately, it is not possible to review all of CLIO's capabilities; therefore, I'll just try to provide a good overview.

MLS MENU

Primarily devoted to loudspeakers, this menu is used for all "pseudo-anechoic" measurements. (For a discussion of MLS theory, read D. Rife and J. Vanderkooy's "Transfer Function Measurement with Maximum-Length Sequences" in *JAES*, Vol. 37, 1989.)

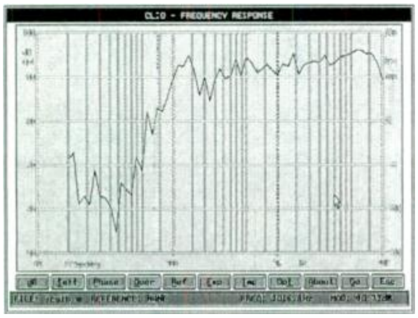


FIGURE 8: Satellite speaker's frequency response.

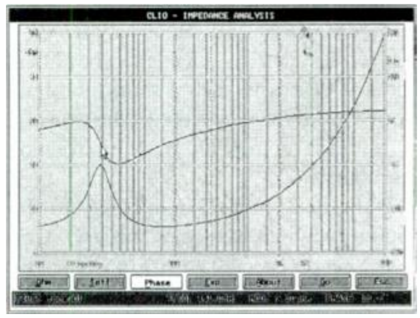


FIGURE 9: Impedance curve of a Focal 21 woofer.

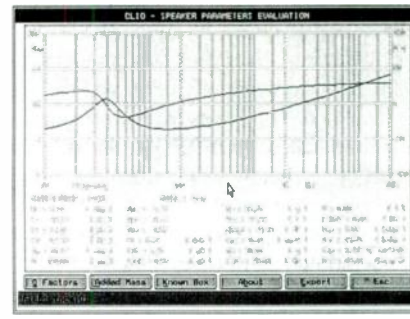


FIGURE 10: Focal 21 woofer parameters using the added mass method.

The spectral content of MLS sequences is similar to white noise. With the appropriate correlation, you will get the impulse response of the "linear" system under test. Audiomatica uses a 14-order sequence. A sample rate of 51.2kHz has an equivalent time window of 320ms—more than enough for the usual measurement. To make it work, simply connect the mike to the HR2000 board, start the software, select MLS, Time, and press the "G" key.

Figure 3 shows the impulse response (which I took in my listening room with the mike at about 50cm on tweeter axis) of my active powered two-way satellite with the subwoofer shut down. (The room is not very large: the ceiling is only 2.5m high.) An understanding of this screen is necessary to making accurate measurements.

During the acquisition process, 16,384 samples are memorized. If the sample frequency is 51,200 samples/sec (as shown here), it will correspond to a time span of 320ms. When you press the "M" key, the software computes FFT at the fixed length of 8,192 points, and brings you to the frequency domain. This FFT length allows you to analyze a maximum of 8,192 samples (160 of the 320ms acquired). The "anechoic" part of the measurement requires using the marker and

TABLE 2

MICROPHONE SPECIFICATIONS

Type	Condenser electret
Accuracy	±1dB, 20Hz–10kHz ±2dB, 10Hz–20kHz (direct field)
Max. level	120dB SPL
Dimensions	8mm diameter, 25 or 12cm long
Accessories	3m cable, stand adaptor

the start/stop time buttons to select the first (1.4ms) and second (5ms) arrivals of acoustic energy at the mike, the latter being the first reflection coming from the floor.

Two very interesting windows appear on the right of the screen. The marker section always gives you the level, time, and distance with respect to cursor location. The "equivalent" distance is based on a sound speed of 344m/s. In the window section you can read the start and stop times you have selected, and the so-called "LOFREQ" value (the lowest frequency below which the measurement may lose significance).

The LOFREQ value is the inverse of the time window length (here about 275Hz). In my opinion, it is also the "acoustical" resolution of the measurement. Note that this is not strictly a limitation of CLIO, but applies to all FFT systems. A way to increase the resolution is to admit some reflections by widening the time window.

Figure 4 is the MLS frequency screen. As you can see, nothing reliable can be obtained under 400Hz with such a short time frame. Above this frequency, you get the equivalent of a dead room with more conventional instruments.

In *Fig. 5* I used CLIO's ability to show you up to ten curves simultaneously. Curve A is pseudo-anechoic, with a 3.5ms time frame (as seen in *Fig. 4*); Curve B has a larger time frame, including several reflections from floor, ceiling, and rear wall. We now have a more realistic view of the low-frequency range, as well as a lot of holes, bumps, and "grass" due to reflections. Clicking on the Thrd button gives you Curve C, an often-useful 1/3-octave smoothing of Curve B. MLS sequences also give you the phase response.

You can see both amplitude and phase by clicking the over and phase buttons. In order to obtain a correct representation of the phase curve, you will need to toggle occasionally to the frequency windows to set the proper propagation delay set by the time start marker. The min button provides the minimum phase behavior according to the frequency curve through the Hilbert transform. (Refer to Dick Campbell's comments about FFT analyzer and phase in his review of "Loudspeaker Recipes, Book One," *SB* 5/94, p. 53).

The energy time curve calculated on the entire acquisition time is shown in *Fig. 6.*

FIGURE 11: Satellite speaker's real-time analysis.

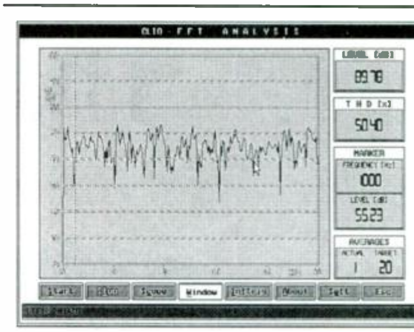


FIGURE 12: The spectral of CLIO's MLS sequence.

TABLE 3

MICROPHONE PREAMPLIFIER SPECIFICATIONS

Type	Rechargeable battery powered individually calibrated for MIC-01 or 02 A-weighting filter
Frequency response	20Hz–20kHz, ±0.1dB 10Hz–100kHz, ±1dB
Max output level	+10dBV
Level range	70–120dB SPL
Attenuator step	10dB
Attenuator accuracy	0.1dB
Accessories	220V AC main adaptor

Depending on the time span chosen, two different convolution algorithms are used, 5ms being the boundary.

Under the MLS menu, you can display Cumulative Spectral Decays (CSD), which are the most useful waterfalls for loudspeaker designers. *Figure 7* is the CSD of the impulse response in *Fig. 4*. Frequency and Time domains are conveniently close.

SINUSOIDAL MENU

As you can guess, this domain uses spot sinusoidal frequencies. You can choose 1–24 discrete frequencies per octave, from 10Hz to 20kHz. *Figure 8* shows the frequency response of my satellite speaker under the same conditions as with MLS. Of course, all reflections are included, and there is no way to remove them. Four main uses for this menu include:

1. Checking electronic circuits, such as crossovers, or preamplifier correction curves, by displaying amplitude and phase curves.
2. Manipulating data with the opt button. You can add, multiply, subtract, or divide by even complex values on-screen data or already saved data files. MLS curves can be imported (provided they have been previously exported in ASCII format) and "split." You can splice a near-field MLS measurement of the 20–200Hz section and a far-field measurement of the 200–20kHz section to produce one full-range curve. (See D.B. Keele's paper on this topic entitled "Low Frequency Assessment by Nearfield Sound Pressure Measurement," *JAES*, Vol. 22, No. 3, April 1974.)
3. Testing impedance is as easy as connecting the two included cables to the driver and pressing the "Go" key. *Figure 9* shows the impedance curve of a 21cm Focal woofer. You can choose 50, 100, or 150Ω for the vertical scale, but I would have preferred an "auto" vertical scale. I have successfully tested impedance accuracy with precision 0.5% resistors.
4. Quickly achieving T/S parameters by choosing between the added mass and box compliance methods in the speaker parameter module. (Read the "Advice and Cautionary Notes" paragraph in the manual.) They use the –3dB frequencies to get the F_S and Q values, but I prefer the root-mean-square method because it is less prone to error. A glance at the displayed impedance curve is a must here. *Figure 10* shows the parameters for the Focal 21 woofer, using the added mass method (20g load).

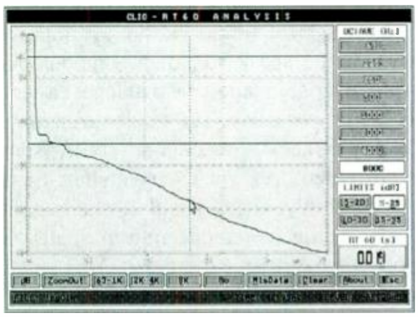


FIGURE 13: The Schroeder plot of the impulse response in *Fig. 3*.

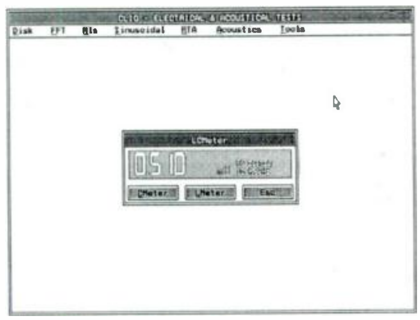


FIGURE 14: Inductance and capacitance values obtained using the LC meter.

À LA CARTE

Do you remember how expensive a 1/3-octave real-time analyzer was a few years ago? For a fraction of the price, you've got one with the HR2000 board. *Figure 11* shows the same satellite speaker. Several possibilities are available, including average, reference curves, and cursor. If you want to know the response of your listening room, or adjust your parametric equalizer, the real-time analyzer menu is a must.

Use the FFT-analysis feature to determine the frequency content of an input signal. Remember, the maximum sampling rate is 51.2kHz, and be careful not to exceed the Shannon frequency. Several time-weighting windows (Hanning, Hamming, Blakman Harris, and Bartlet) are provided. *Figure 12* is the spectral of the MLS sequence. With a pure sine (such as those generated by one of the internal generators), the THD is given as a bonus.

The Acoustic menu allows you to calculate the RT60 based on an MLS sequence via a post-processing technique developed by Mr. Schroeder. This is a very important measurement for any room designer. *Figure 13* is the Schroeder plot of the impulse response in *Fig. 3*. You can choose up to four bands for analysis. As I'm not very familiar with these measurements, I won't evaluate this feature.

The LC meter is a very clever and useful instrument for measuring real inductance and capacitor values (*Fig. 14*). CLIO will automatically choose the proper frequency to generate the most accurate value (within its frequency range).

The CLIO manual is well-written, and will help you to discover the power of the system. Maurizio Jacchia from Audiomatica is writing an appendix on how to generate signals, with examples in C, Pascal, and Basic.

This has been only a quick glance at the CLIO system. I would like to see improvements in the Speaker Parameter module and in phase representation, by making it easier to set the proper delay. The implementation of Quality Check procedures could stand improvement, too. Yet, even with half its current capability, CLIO is still a real bargain. Clever hardware engineering has allowed room for future enhancements. This system will not be "obsolete" in the near future.

I strongly encourage anyone who's looking for such a measurement system to obtain the demo disk. A word of caution: about a year ago, I was sent a CLIO demo disk by an Australian friend, who wanted to know my opinion of it. One month later, CLIO was installed in one of my three computers!

ABOUT THE FIGURES

Two shareware jewels were invaluable in creating the figures for this review. All screen shots were taken using Professional Capture System version 1.02 (DOS and Windows), and all color and file manipulations were performed with Paint Shop Pro version 2.01. Both sharewares are available through JASC, Inc., 10901 Red Circle Dr., Suite 340, Dept. SB, Minnetonka, MN 55343, (612) 930-9171. Please note that original screens were in color and taken in EGA mode.

MANUFACTURER'S RESPONSE

First of all, we'd like to thank Jean-Claude

PREVIEW
Glass Audio

Issue 4, 1995

- Brute Force in a Line Stage
- Small-Signal Pentodes for Audio
- Diff Amp Driver With BJT Source
- Balanced Input Phono Preamp
- Old-New Bias on Bias

Gaertner for his patient and thoughtful review of our measurement system. He is an experienced user of CLIO, as well as other measurement systems. It is also clear that Mr. Gaertner is a skilled PC user, so what he pointed out regarding software and hardware compatibility is a valuable and clear contribution for the user.

As CLIO's manufacturer, we have to deal with Mr. Gaertner's complaints—not so many, luckily. Actually, all the microphones we deliver have a serial number engraved on the body; the same number is present also on the case and in the calibration chart. The complaint about the unbalanced microphone signal makes sense, but the main topic here is the length of the cable. Its capacitance acts on the frequency response and a balanced version would not help in this regard.

Some topics regarding MLS measurement pointed out by Mr. Gaertner should be thoroughly examined. This would require space and we believe it is not appropriate here. Nevertheless, as this argument is of general interest, we would like to add, in a very concise way, our opinion. With Dick Campbell's FFT comments in mind, "TEFie-talk" is in the air here, even if in a subtle way. Low-frequency limit is not an FFT analysis limit but is inherent to any semi-anechoic approach in semi-reverberant rooms, whichever instrument is used.

It is not completely clear to us what Mr. Gaertner means by "acoustical resolution of the measurement"; we prefer to think in terms of amplitude accuracy and frequency resolution. The frequency resolution of an FFT analysis is always (sampling frequency)/(size of FFT), 6.25Hz in the case considered by Mr. Gaertner, regardless of the time window selected. The same cannot be stated regarding amplitude accuracy. If the impulse response of the device under test is shorter (i.e., decayed to a negligible value) than the selected time window, then an accurate frequency response modulus is obtained. If not, an error in modulus results, which, for practical cases, is evident at lowest frequencies. For the user to value this accuracy of this measurement is directly comparable to value the similar error that arises when exciting the DUT with a burst or sweep in

to page 73

OLD COLONY SOUND LAB

PRODUCT SHOWCASE

LAUD

LIBERTY AUDIOSUITE

SOFTWARE

Liberty Audiosuite (LAUD) is an integrated collection of high performance audio test instruments implemented in software and making use of the versatile capabilities of the Personal Sound Architecture sound cards. LAUD instruments include MLS/FFT, a sophisticated dual-channel MLS- or impulse-based network and impedance analyzer; SINE, a versatile dual-channel gated or continuous sinewave-based network/impedance analyzer; SCOPE, a dual-channel triggerable digital oscilloscope; SPEC_AN, an FFT-based spectrum analyzer with noise source; and DIST_AN, a harmonic distortion analyzer.

Each "instrument" is equipped with a range of features and options to provide great versatility for advanced users. Yet LAUD is easy to operate, allowing new users to achieve useful results in minimal time.

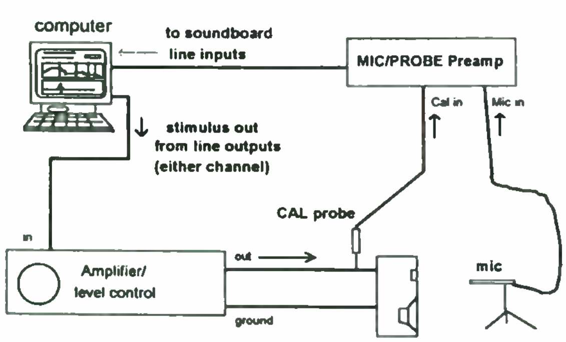
Some advanced functions and features included within various instruments in the package include a measurement and extraction system for T/S parameters; full measurement in both magnitude and phase; energy/time curve analysis; powerful data manipulation and conversion facilities; definable display axis and scalings; floating, dual on-trace data markers for reading of data values; a distortion visualizer; waterfall plots; software-controllable input and output gains; 14 sample rates from 5.5125kHz to 48kHz; dual-channel 16-bit data acquisition; built-in automeasure macros; built-in help; and much, much more!

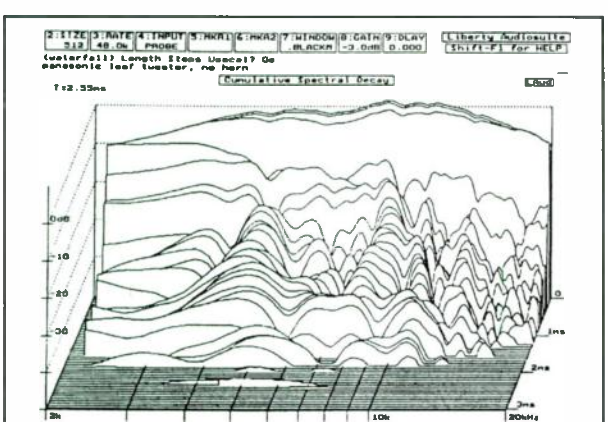
Requires 386-, 486-, or Pentium-based computer; at least 4Mb RAM; VGA or SVGA; mouse; Personal Sound Architecture DSP-type sound card, such as Orchid Soundwave 32 or 32+ and Gamewave 32+, Cardinal Digital Sound Pro 16, Wearnes Beethoven 16-DSP, Adaptec DSP, Paradise DSP16, or Echo DSP. All other types of sound cards will not function with LAUD *at all.*

Package comes complete with software (IBM 3-1/2" DS/HD DOS only) and 74-page manual in blue looseleaf binder. Please note that many measurements will require good-quality test probes, various long cables and/or adaptors, a measurement-quality mic and mic preamp (for acoustical measurements), and/or a power amplifier (for speaker work). See Accessories for LAUD and IMP.

SOF-LAU1B3G
Liberty Instruments

$269





LAUD, LAUD, LAUD—
what do I need?

There are a number of ways you can outfit yourself with LAUD, in addition to totally fabricating everything, in which case you would need to buy only the SOF-LAU1B3G software if you already had a compatible sound card (see software description). Here are two other possible ways:

Basic Unassembled, Uncalibrated Version

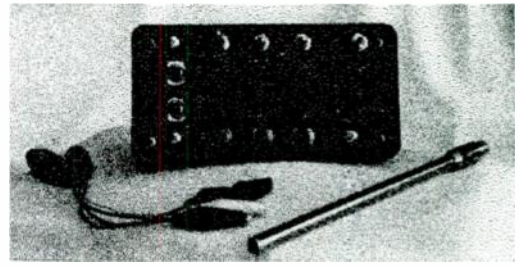
SOF-LAU1B3G	Software and manual
KW-5	Unassembled M/PP
KW-5M	Unassembled mic and wand
LIBCARD-E or other	If needed
Test probes	Make your own
Various cables	Supply yourself

Professional Assembled, Calibrated Version

SOF-LAU1B3G	Software and manual
KW-5A	Assembled M/PP
LIBMIC	Assembled, calibrated mic plus wand plus clip holder
LIBCARD-E or other	If needed
KW-5P	Assembled Test Probes Set
Various cables	Supply yourself
SOF-AID1B3G	Liberty Aids performance enhancement software (see ad elsewhere)

About calibration: Through extensive testing, we have found that about 1% of the mic capsules (also known as cartridges) we use will have behavior anomalies worth noting, while an additional 13% will be slightly aspecific but within tolerable limits. Calibration procedures compare your specific capsule with a known true response reference, so that you know if and where there is any deviation.

Accessories FOR LAUD and IMP



Old Colony Sound Lab

KW-5	Unassembled Mic/Probe Preamp only. Enables your sound card testing software. Does not include probes, mic, or wand. Silkscreened case measures 5-1/8"W x 2-5/8"H x 1-5/8"D. From *Speaker Builder* 5/95.	**$69.00**
KW-5A	Assembled Mic/Probe Preamp only	**99.00**
KW-5M	Unassembled Liberty mic plus 12" brass wand, uncalibrated	**18.00**
KW-5MA	Assembled Liberty mic plus 12"brass wand, uncalibrated	**29.00**
KW-5P	Assembled Test Probes Set	**12.50**
LIBCAP	Mic capsule with soldered leads, no wand, Liberty calibrated	**23.00**
LIBMIC	Assembled Liberty mic plus 10-1/8" wand plus custom clip/holder, Liberty calibrated. Beautifully polished nickel-plated brass wand is terminated in rugged RCA. Package includes 3-1/2" floppy containing IMP and LAUD compensation file, printout, 90-day warranty, and free recal after one year. Requires 1.5-10V to operate, such as normally supplied by IMP or M/PP. Preamp and power supply NOT included.	**129.00**
LIBCLIP	Rugged black plastic custom mic clip/holder is threaded to mount on standard 9/16" mic stand or boom. Holder is drilled to fit .283-inch-diameter LIBMIC wand.	**23.00**
LIBCARD-E	Echo DSP sound card and version 1.3 software, with manual	**179.00**
LIBCAL	Liberty calibration of Liberty wand	**Call or write for info**

LIBERTY AIDS SOFTWARE

G. R. Koonce

These handy programs from one of the country's great tinkerers will add usefulness and value to your IMP or LAUD system. While the files contain IMP names, they are actually equally usable with LAUD. IMPSPACE designs absolutely anechoic test layouts. IMPROOM designs near field and far field test layouts. IMPZOBEL designs simple and complex zobels based on input .ZMA files. IMPPCOEQ (Passive Crossover and Equalizer) places a four-branch network ahead of the system defined by .FRD and .ZMA files. And IMPEQLZR designs downslope equalizer functions used with .FRD files. In addition, the disk contains three datafiles to play with in IMPSPACE. IBM only, requires VGA 640 x 480 16-color, 280K. Mouse, math coprocessor recommended. From *Speaker Builder* 4/95.

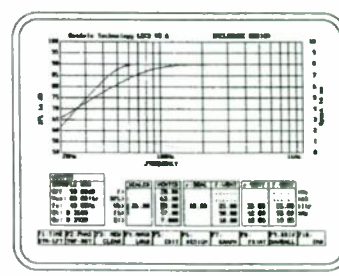
SOF-AID1B3G **$19.95**

LOUDSPEAKER ENCLOSURE AND CROSSOVER DESIGN SOFTWARE

SOF-LEC1B3G

$79.95

ENCLOSURE DESIGN SCREEN



GENDALE TECHNOLOGY

New from South Africa, LECD shows the SPL of a 2- or 3-way speaker system against frequency, based on parameters entered and enclosure chosen. Its straightforward format allows you to see what is being analyzed at all times and to access functions directly.

LECD provides for sealed, vented, sealed-vented, and vented-vented design, as well as calculates passive crossover network component values for first- through fourth-order Bessel, Butterworth, Chebyshev, Gaussian, Linear Phase, Linkwitz-Riley, and Legendre filters.The effect of inverting speaker polarity and of speaker offset can be seen and saved in files, as can all data.

Available graphs include the SPL/frequency response of each speaker in its enclosure; the peak cone excursions of each speaker; the on-axis SPL of the complete system, in magnitude and phase, phase delay, or group delay; the transfer function of each crossover filter section (woofer, midrange, tweeter), in magnitude and phase; the input impedance, in magnitude and angle, of each speaker, as well as at various points and of the complete system; and the equivalent square-wave time response of the crossover filters and loudspeakers.

Republic of South Africa, IBM DOS only, version 3.6. Requires 286 or better, 386+ preferred. Supports all color monitors plus Hercules. 109-page, 7-1/2" x 9" manual in yellow looseleaf.

An interesting and friendly workhorse of a package! Also available:

SOF-LEC1B3GD	LECD 3-1/2" Demo Disk (usable as credit toward later purchase of full package)	**$5.00**

OLD COLONY SOUND LAB
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Ask SB

ON-LINE QUESTIONS

By Dick Pierce
Contributing Editor

[The following is an on-line exchange between arlarnie@aol.com and Dick Pierce.—Ed.]

RE SPEAKER CABLES

Audio cable conducts a signal of bandwidth 20Hz–20kHz, as a minimum. This bandwidth represents a 1,000:1 ratio. I have been unable to think of any reason why cabling needs to conduct at such a ratio. You may not consider it important, but I submit it is at least *unique*. I may be wrong here, but just tell me: no flames are necessary.

Indeed, you are wrong. But you suffer from a misconception common in the high-end community, so you needn't feel singled out. Many people get the notion that, because audio has a frequency range of 1:1,000, or ten octaves, or three decades, or whatever the ratio is, it is somehow "special." There are no problems which make such a range unique in any way. Nor are there different principles; nor is there any magic.

People often think this "wide" bandwidth has much more information-carrying capacity because its range is so impressive. As attractive and intuitively correct as it seems, this notion is wrong.

In terms of information-carrying capacity, there is nothing unique about this particular 20kHz bandwidth. The important concept here is that the amount of information capacity, or resolution, is proportional to the bandwidth, the arithmetical difference between the lower and upper band limits—not the ratio between them. Any 20kHz bandwidth has precisely the same information capacity and resolution as any other (all else being equal). Let's look at an example.

Take the audio band between 20Hz and 20kHz. You say that, since the ratio is 1:1,000, the resolution is higher than, say, a 10–15MHz video channel, which has a ratio of only 1:1.5. In that band, however, we can fit 5MHz worth of channel capacity. An even more extreme case would be a bandwidth of 10.69–10.71MHz. Here the ratio is less than 0.2%! Yet, this is effectively the IF bandwidth for the audio information in an FM tuner.

Your insistence on "coding" notions directly asserts that, since the audio "bandwidth" is about ten billion *times wider than this, we must lose* all *of the audio information when sending via FM. Turning on a good radio station with a good tuner completely destroys the validity of that assertion and your underlying theory.*

For the pickers of nits, the FM IF bandwidth works a little differently than this. For example, it must be twice this width to broadcast stereo. There are other minor implementation details, but the underlying principles remain.

INFINITY QUESTION

But audio cable conducts a signal of potentially "infinite resolution."

This notion is simply, absolutely, and totally wrong. Audio cables are incapable of conducting infinite resolution because this requires *infinite dynamic range and infinite bandwidth. Any band-limited signal with a limited dynamic range does* not *have infinite resolution.*

EFFECTS OF CODING

All other cabling usage seems to "code" the information onto another waveform, such as AM, FM, digital, RF in general, and so on. In my opinion, this "coding" limits the resolution of the original signal at some point, making these methods incapable of infinite resolution.

Wrong—and the FM example demonstrates the fallacy of this notion. What the high-end community must understand is that "infinite resolution" simply doesn't exist. The amount of information, which is plainly the measure of a system resolution, is directly proportional to its bandwidth in frequency, not *the ratio thereof.*

MATTERS OF RELEVANCE

Even current video signals, which arguably contain more information "bits" per second, are encoded, as opposed to being direct-voltage-information related.

If it contains more "bits-per-second"—more information, call it what you will—then by definition it has more resolution. If a system's bandwidth is limited (for whatever reason) to 20kHz, while a measurable "signal" may exist in the "interval" between the information which is relevant to satisfying the constraints of that 20kHz bandwidth, it has no *relevance to that information.*

Another perspective is that if the bandwidth is limited to 20kHz, there will not be any information between the information of relevance. With a bandwidth of 20kHz, no significant change in the signal can happen at a rate faster than once every 1/40,000 of a second. It makes no difference whether the system is analog or discrete time-and-amplitude-sampled digital. The notion of continuity does not imply infinite resolution.

If some significant change occurred at a rate faster than 1/40,000 of a second, then the bandwidth would be wider than 20kHz. And it is irrelevant whether that 20kHz bandwidth is from DC to 20kHz (conventional audio) or from 1GHz to 1.000000020GHz, no significant changes will happen in either of those bands at a rate faster than 1/40,000.

This ignores the further restriction on resolution made by limited dynamic range, either from noise or quantization. It's irrelevant to the question of resolution. A system's dynamic-range limits basically set a minimum change in amplitude, below which there can be no information about the original signal due to the fundamental uncertainty imposed by the system.

Two very important concepts are resolution in time and resolution in amplitude. First, the resolution in time is a function of the system's bandwidth. The ability to distinguish two unique events in time without ambiguity is limited by the bandwidth. The wider it is, the more time resolution the system has. Second, the resolution in amplitude is a function of the system's dynamic range. The ability to distinguish two unique amplitudes without ambiguity is limited by the system's dynamic range. The wider the dynamic range, the more amplitude resolution.

You can combine these concepts and say that the total resolution of the system—its information-carrying capacity—is essentially the product of its bandwidth (e.g., 20kHz) and its dynamic range. Either must *be infinite for that system's resolution to be infinite. I challenge you to find any system having infinite resolution, because that would require infinite bandwidth or infinite dynamic range. And we all know that no such system exists, be it studio analog tape recorder, DAT recorder, cassette, amplifier, piece of wire, chunk of air, or the human ear.*

EXPERT ADVICE

[In another on-line exchange, barton@vax.sbu.ac.uk questions the nature of expertise.—Ed.]

The real experts don't disagree. It's the quasi-experts, the self-appointed experts, who seem to dominate the popular music press. The theory has been around long enough to be proven correct. How do I become an expert? Is there some exam I can take? Could I put it on my CV?

The easiest way to become an expert is to say, "I am an expert," and put it on your CV. You are then an expert until someone challenges you by asking a question that requires real *expertise.*

The more difficult way is to study a long time. Put in 20-hour days, six to seven days a week, in the trenches, to see what works and what doesn't. Say dumb things and then be forced to eat them diced into crow pie. Try things out, understand why they failed or succeeded, and, always—learn, learn, learn. Do this for about 20 years, take the result, put it on your CV, and you are an expert until someone challenges you by asking a question that requires real *expertise.*

It's the aftermath of trying to answer that question that separates the experts from the charlatans.

PERFECTING A PIONEER

A number of readers have written about their experiences with the popular 4.5" full-range Pioneer A11EC80-02F (available from Parts Express and MCM Electronics). I've used it as well and can recommend it as a quality speaker.

In response to the dearth of detailed technical information on this speaker (i.e., response curves), I provide measurements and also suggest some simple modifications to improve the performance of this and other speakers.

Figure 1 shows the response curve of an unmodified A11EC80 in anechoic half-space conditions at 1W/1m. It shows a respectably smooth curve for a $10 speaker, which makes a nice full-range if you're not concerned about the uppermost octave. Distortion (%THD) is also plotted at 4W RMS. You'll notice a distortion spike of about 6% just below the 1.5kHz response dip. There's also a sharp 7.5% distortion spike at 3.5kHz.

These spikes are due to excessive cone flexure. The 1.1kHz spike occurs because the surround begins to move out of phase with the cone body and bends the edges of the cone beyond the linear elastic limits of the paper. At 1.5kHz the surround moves the opposite direction of the cone at any given time, producing a dip. The 3.5kHz spike is due to breakup of the cone itself.

If you wiggle the opposite end of a 5' rope that is tied to a wall, you'll see a wave travel from your hand to the wall and reflect back to your hand. This is a standing wave. Now imagine a 4' rope tied to a 1' rubber strap, which is in turn tied to the wall. You can see a discontinuity in the wave where the rope is tied to the rubber strap. Depending on the materials, the strap may flap wildly or it may absorb all of the energy that comes to it and not reflect any of the wave back at all.

This is analogous to a speaker cone. The impulse originates at the voice coil and travels to the edges of the cone where we hope it is uneventfully absorbed by the surround. Unfortunately, sometimes surrounds "flap wildly," but there are some simple measures to fix that.

Many times the surround is sufficiently soft and absorbent, but, because it is too flimsy or has poor geometry, it has a tendency to flap. This situation can be greatly improved by putting a thin bead of glue around the inside edge of the surround where it meets the cone. This will often add enough stiffness to the surround to prevent the flapping and smooth the response.

This is the case with the A11EC80. I applied a small bead of Elmer's School Glue around the crimp in the surround, and then smoothed it with my finger *(Fig. 2)*. I put the surround under a heat lamp for a few minutes until the glue dried, then measured the speaker again. Notice that the 1.5kHz dip has all but disappeared, and so has the distortion spike *(Fig. 3)*. Apparently, the surround now does a better job of absorbing standing waves at 3.5kHz, since the distortion there has fallen as well.

This simple modification works for many woofers and mids. It won't turn a sow's ear

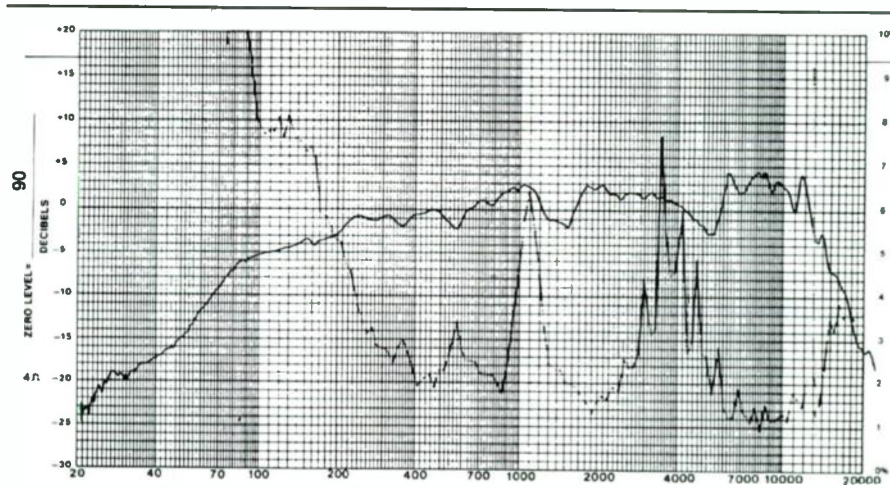


FIGURE 1: Untreated Pioneer A11EC80. Total harmonic distortion 70Hz–16kHz/0–10% scale at 8V RMS (4W).

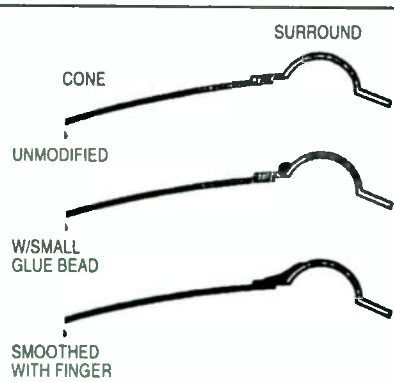


FIGURE 2: Cone cross-section: a) unmodified; b) with small glue bead; c) smoothed with finger.

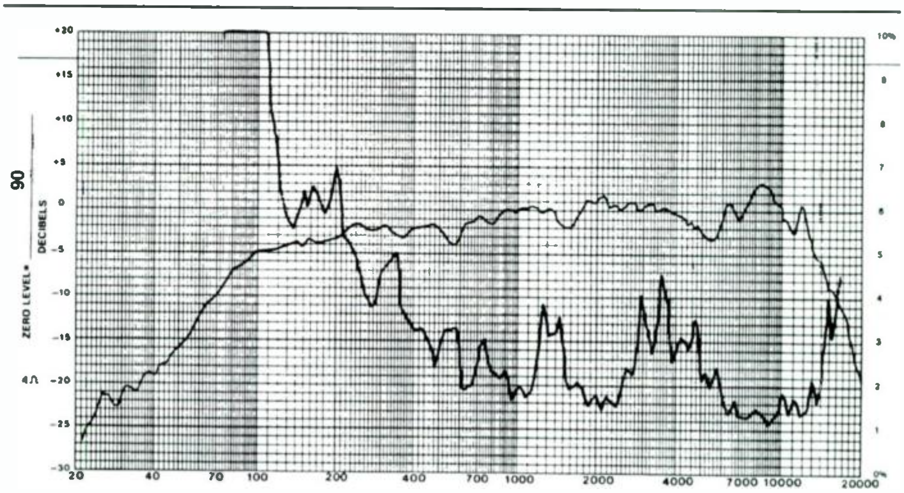


FIGURE 3: With edge treatment. Total harmonic distortion 70Hz–16kHz/0–10% scale, at 8V RMS (4W).

into a silk purse, but it sure can make a substantial improvement, turning a mediocre speaker into something quite acceptable. Certainly, you could modify the A11EC80, as I have, add an inexpensive tweeter, good crossover, and ten-liter box, and have a very pleasing small pair of speakers for $50.

Speaker manufacturers have been using this technique with success for years. Now you can, too.

Perry Sink
Berwyn, IL 60402

Mike/Probe Preamp

Continued from page 22

metal-type drills, or, better yet, punches. A lettering pattern for the cover is shown in *Fig. 6*. Of course, you can mount your switches and connectors off the board for use with a different enclosure.

OPERATION

After being switched on with S1, the Mike/Probe Preamp takes about ten seconds to stabilize before use, as the coupling and bypass capacitors charge and the op amp settles. Operation is relatively straightforward. If you require the Meas input of the soundboard to come from the mike, set S4 to the mike position. For an electrical measurement rather than an acoustical measurement, set S4 to PROBE.

S2 and S3 select the attenuation for channels 2 (Cal) and 1 (Meas). The center-off position is 0dB (or no) attenuation. If looking at signals which peak above 1.4V, switch to –20dB attenuation (the 10× position). If that still is not enough, switch to –40dB (100×). When used with 47.5k resistor probes, the Mike/Probe Preamp will be protected for peak voltages of over 100V. But....

Caution! The voltage protection levels of the circuit are not meant to imply suitability for regular use of the device in measuring high voltages. This preamp is *not* intended for measuring AC line voltages, internal nodes in tube circuitry, electrostatic panels, or similar high-voltage circuits. The Mike/Probe Preamp's insulation and construction is intended for normal use in measuring peak (DC and AC combined) levels normally below 50V. When working with or measuring any voltages found internal to electronic circuitry, you should observe appropriate safety measures.

Two other cautions: (1) Don't feed the Mike/Probe Preamp's probe inputs with anything other than a 47.5kΩ resistor probe. Although the gain will be increased if the series resistor value is reduced, the circuit protection will be limited to lower levels. (2) Never connect the probe's ground lead to any point which isn't the same potential as the sound card's ground. In particular, power amplifier output terminals marked minus (–) are not necessarily ground, particularly in the cases of tube amplifiers or bridge amplifiers. In general, you should not use these types of amplifiers with the Mike/Probe Preamp.

CONCLUSION

Although they restrict computer-based audio measurement systems to PCs which can take plug-in cards (ruling out most portables), multimedia sound cards can be used for very high-performance audio measurements in fixed installations. Several software packages are available for working with various sound cards. For example (caution: shameless plug follows), Liberty Audiosuite, working with some of the more advanced programmable DSP sound cards, allows measurements of anechoic response (via pulse, MLS, or gated sine wave), impedance, energy-time curves (and RT60), spectrum, distortion, Thiele and Small parameters, waterfall plots, and dual-trace oscilloscope readings at 14 different sample rates up to 48kHz.

While the sound cards provide the sophisticated acquisition gear, and good software programs transform the cards into lab-grade instruments, the addition of the Mike/Probe Preamp can make this combination really practical and compatible with real-world situations. At the same time, it preserves the exceptional resolution and accuracy of which such a system is capable.

RELATED PRODUCTS AVAILABLE FROM OLD COLONY SOUND LAB

ITEM	DESCRIPTION	RETAIL
KW-5	Unassembled Mike/Probe Preamp, not including mike	$69
KW-5A	Assembled M/PP, not including mike	$99
KW-5M	Unassembled Liberty mike and wand, uncalibrated	$18
KW-5MA	Assembled Liberty mike and wand, uncalibrated	$29
KW-5P	Assembled test probes	$12.50
LIBCAP	Mike capsule w/leads, Liberty calibrated	$23
LIBMIC	Assembled mike, wand, and holder, Liberty calibrated	$129
LIBCARD-E	Echo sound card and software	$179
SOF-LAU1B3G	Liberty Audiosuite software and manual	$269

Shipping: USA, add 7%; other surface, add 14%; other air, add 25%.

SOURCES

Liberty Instruments
PO Box 1454, West Chester, OH 45071
Tel./FAX (513) 755-0252

Old Colony Sound Lab
PO Box 243, Dept. SB5, Peterborough, NH 03458
(603) 924-6371/6526, FAX (603) 924-9467

Pox Humana

IN THE MANNER OF GALILEO OR, HO-HUM, THE OLD WORLD ORDER

By Dick Pierce

In a typical response to others' failed attempts to audibly discern improvements due to some obscure tweak or a magic wood disk stuck to the wall, Michael LaBella writes:

"My impression is you may have created a series of assumptions as to your system's resolving capabilities, not only on a hardware/software basis, but also as it is affected by the AC power supply system, and the physical environment in which the hardware system resides. In other words, was your system a valid measuring tool which produced results consistent with those experienced by other people's systems?"

Lessee, the cable (substitute digital, blue tack, speaker stand, and so on, or call it a "thingy" for the moment) argument goes something like this (in the style of a Galilean dialogue):

Propono: "The difference between thingies is obvious and dramatic. It's not difficult to hear the difference, so profound as it is, anybody can!"

Observo: "Well, I set up an experiment in my system using my music, and I am puzzled and chagrined that I could not reliably hear this difference of which you speak."

Propono: "Aha! Obviously, your equipment is not of sufficient, uh, Resolving Power™ to reveal the difference."

Observo: "But, kind sir, I spent a veritable king's ransom to buy Resolving Power, yet I cannot say with the same conviction as thee which was which."

Propono: "Then, it is obvious the music you listen to is to blame. It lacks the right kind of Information to exploit the Resolving Power and reveal the differences which are obvious to me."

Observo: "But then I am confused, because it was that music which indeed has the right kind of Information to which I was listening that led to my findings, that I could not hear the difference. What lack I, prithee?"

Propono: "Obviously, your environment clouds the subtle distinctions between the thingies and leads to your confusion."

Observo: "But I strove to achieve the Right Environment, one with no clouds, and still the subtle distinctions elude me."

Propono: "Yes, the obvious differences of which I speak are so subtle that it requires only the best-trained ear to distinguish and appreciate these differences."

Observo: "Yet, after long, arduous hours of listening and ear training, I am at a loss to hear a difference that I trust to be real."

Propono: "Then, alas, you are deaf, because you cannot hear these differences, ones which are so apparent to me. I mourn for your loss, but not much, because you are clearly an audio Philistine. Please draw the curtain on your way out."

Observo: "Before I go, let me ask: you say these huge obvious differences are so subtle that I can't hear them. I am confused further because first you said the differences are so striking anyone can hear them, but now, maybe they're so subtle that only an anointed few are blessed with the ability. What are the facts?"

Propono: "Facts? There are no facts, only TRVTH™, and only I and a numbered few are in possession of the TRVTH. Please go then."

Observo: "By the way, while we were talking, I switched thingies behind your back, and you failed to notice any difference. Do you care to comment?"

Propono: "Your deafness is so profound, your skepticism so pervasive, your lack of faith so entrenched, that even I can't hear the differences that are so dramatically subtle, so subtly obvious. Begone with thee!"

Observo: "Good-bye, then, Propono. You are so full of it."

Now, some might take umbrage at the notion of a hypothetical Galilean-style dialogue between a proponent of some audio magic and a bewildered disciple unable to share in the enlightenment. One of the objections might be that such dialogues would not, *nay*, could not occur in real life, and thus my whole premise is ridiculous and stupid.

Rather than dredge up the innumerable times such a dialogue *has* occurred on electronic networks, for instance, it might be informative to relate a real example of one that occurred in the *Journal* of the Audio Engineering Society. In fact, it is a classic upon which I directly based my satire.

J. Diamond fired the first round when he presented a talk at the 66th convention of the AES, May 6–9, 1980 ("Human Stress Provoked by Digitalized Recordings"). Dr. Diamond asserted that digital encoding caused stress in listeners. This was "tested" in a method "in which volunteers had their arms pushed down more easily by the speaker during the digitally recorded passages" (N. Morgan, "Comments on Human Stress Provoked by Digitalized Recordings," *JAES* Vol. 28, No. 9, p. 613, Sept. 1980). The test did not include any of the well-established techniques for measuring muscular stress such as galvanic skin resistance or electroencephalogram measurements.

Despite this, Dr. Diamond states clearly (*db Magazine*, Jan. 1981) that his method promises results for nearly any practitioner, and that the effects are profound and easily detected. So, he has met the criteria of the first stage of my example by suggesting an effect that is, in the proponent's words, of large enough magnitude and of small enough ambiguity to be easily detectable by all (or, at least most).

Now, along comes Mr. Morgan with a double-blind test (in his *JAES* rebuttal letter) that not only uses the "pushing test" methods recommended by Dr. Diamond, but also includes other, more objective measures, such as galvanic skin resistance. The results indicated no statistically significant differences between the phenomenon alluded to by Dr. Diamond and pure random chance. Thus, the second stage of my little Galilean scenario has been met: someone else couldn't detect the "obvious" results.

To add fuel to the fire, the Detroit AES Section conducted its own tests, with Dr. Diamond coaching one person via phone, and careful double-blind techniques also used. The session's conclusion was "for all groups and all individuals, there was no statistically significant correlation between analog or digital sound and a weak or strong deltoid response ("Detroit Tries Digital Test," News of the Sections, *JAES*, Vol. 29, No. 5, May 1981).

The test results were anything but profound and easily detectable by any practitioner (in other words, unconditional). With the ball squarely in Dr. Diamond's court and the stage set for the next chapter in our Galilean dialogue, we await his response.

In "Comments on Stress Test" (JAES, Vol. 29, No. 10, p. 749, Oct. 1981), he states, "There is a level of expertise required to carry out the test successfully." This "level of expertise" requirement disagrees with his original assertion.

In further contradiction, he invents "magic," to account for the fact that only he can detect the phenomenon:

"I have found, for reasons of which I am unaware, that increasing the length of lead between pickup and preamplifier, or preamplifier and amplifier, would have the effect, under many circumstances, of overcoming the digital effect." He adds, "On an average day I perform the test 350 to 500 times, and I have done this for many years. This is certainly more than the eleven or so 'push tests' [which conformed to the demonstration he gave in 1980] of Morgan or the few tests that Clark performed [using the method described by Dr. Diamond himself]."

He admits, "Parameters such as EEG and GSR did not reveal the problem. Of this I am well aware. To date the problem is revealed only by the test I advocate."

Finally, he says, "Only when they attend to all the variables as I do in my research, and when they develop a level of expertise in performing the test, will it be possible to compare their results with mine."

And thus he has achieved the final stage of our little *Gradus ad Parnassum*. He has thoroughly discredited his own assertion by heaping qualification upon qualification in an attempt to disprove consistent objective findings. His method is highly subjective. To quote Mr. Clark's reply to Dr. Diamond's criticisms:

"Once this effect was thought to be so strong and obvious that a single paragraph encouraged you to get started in deltoid testing. The effect could even be demonstrated in five minutes, at an AES paper session. Now it requires a uniquely qualified sound system and a tester trained in person by Diamond. Clearly, Diamond is disqualifying anyone but himself and others who get results he likes from running this experiment."

So here we have, in evidence against several people's assertions, a real and unambiguous example of a ridiculous scenario. Surprisingly, though, I must agree that the scenario *is* ridiculous. Not, as many contend, because of its vanishingly low probability, but for just the opposite reason, because of the depressing regularity such a dialogue appears in the popular and professional press and electronic network forums.

SB *Mailbox*

VENT TUNING

In the article "The Third Dimension: Symmetrically Loaded" by Jean Margerand (*SB* 6/88, p. 29), I am unable to follow algebraically how he gets from

$$F_B^2 = \frac{3 \times 10^4 \times S_V}{L_V \times V_F} - 0.9\sqrt{S_V}$$

to

$$L_V^2 = \frac{3 \times 10^4 \times S_V}{V_F \times F_O^2} - 0.9\sqrt{S_V}$$

Can someone explain this?

Jim Mayles
Wheeling, WV 26003

Contributing Editor Robert Bullock responds:

The first formula is dimensionally incorrect as it stands. To correct it F_B^2 and L_V should be interchanged. The second formula is then obtained from the first by the trivial substitution of F_O for F_B. The dimensionally corrected formula, however, still appears to be incorrect.

Using formulas (5.38), p. 129, and (5.47), p. 133, of Acoustics *by Beranek and a little algebra yields*

$$L_V = \frac{C^2 S_V}{4\pi^2 F_B^2 V_F} - k\sqrt{S_V}$$

where the variables are as in Margerand, C is the speed of sound, and k is a constant equal to $1.226\sqrt{\pi}$, $1.463\sqrt{\pi}$, or $1.7\sqrt{\pi}$, depending on whether neither, one, or both ends of the vent are flanged. I am not able to obtain Margerand's coefficients from this formula for any choice of units or constants.

I hope this helps.

IMP QUESTIONS

Like most speaker builders, I work in a crowded workshop with limited space, and making measurements with the IMP, trying to calculate approximately where the first reflection exists, is quite difficult even with all the furniture along the walls to lengthen the time between the first reflection and the mike. The room is pretty well damped with acoustic panel board on all four walls, which increases the difficulty. So to aid in finding the first reflection I use a piece of flat hard Masonite™ to cause my second wiggle on the plot. How do I know if the reflection is possible transient motion of the driver, so it doesn't introduce major error in the measurement by cutting off some transient information?

Also, will mike location change the phase angle at different frequencies? If the mike is placed exactly x feet (wave length at a certain frequency) away from the speaker at the point at which that certain frequency wave has been completely acoustically created, would the apparent phase be 0°? If the mike is placed 1.129′ away from a driver reproducing 1kHz, is the phase shift at zero? Also, why does the graph only have ±90° shifts. How are these graphs read at a phase shift above 90°?

When I use the IMP on T/S parameters, I can't force a DVM reading into R_E, which would yield a more accurate (DC) ohm driver reading. I'm also having trouble changing the effective driver diameter displayed. Finally, what does the distance command do, and how can it be used?

Brian Kuder
Parma, OH 44134

Bill Waslo responds:

There is no easy way to absolutely verify that the speaker's main transient has died down sufficiently before the echo arrival. It *depends on how quickly the transient decays and the distance to the reflecting surfaces. And then you must decide: are the cabinet edges' reflections part of the transient decay or part of the echo reflections?*

To make matters worse, the decay is all a matter of degree: to be completely accurate, the transient never dies out; it will decay eternally (or practically, until it is into therma noise). Knowing the mechanism of a speaker's operation, we know that there is no means within the speaker for a second large burst of energy after a long delay (there's no place in the speaker to store such a long sequence of sound). So we can be assured that the energy from the speaker, after all the drivers' initial peaks arrive at the measuring or listening position, will continue to decay as time increases.

If you see a relatively "dead" period (about 0.2ms long) in the impulse response, followed by a sudden high peak, you can be fairly certain the peak is an echo coming after the response is pretty well decayed. But there isn't a good way to achieve an anechoic response in very tight spaces; anything within about a wavelength at a given frequency is essentially part of the speaker at that frequency. If you know you must cut into the tail of the transient, use a ".BLACKMAN" window to minimize the effects.

You are correct about the phase angle of the acoustical response. It will roll with increasing delay, and the speed of sound covering the distance from speaker to mike causes a delay. To remove this, use the F9 key. For instance, if you measure with the mike 1.129m from the speaker reference plane (whatever you determine that to be), and keep the time marker #1 all the way to the left (so the true delay from speaker to mike is left intact) during the measurement and subsequent FFT, then press F9 to remove this delay and enter the value:

$$(1.129[m])/(345[m/s]) * (1{,}000[ms/s]) = 3.27ms$$

The graphs have only labels at the 90° grids, and it is understood that the upper and lower edges of the graph are +180° and −180°, respectively. Note that there is no difference (mathematically, or any other way) between 180° and −180°, so the plot rolls around, and therefore all possible angles can be displayed. If you go one degree more negative than −180, you are at +179, which is the same as −181. You can also read the angles by placing a marker on the curve. Readouts on the screen display the angle at the marker in text.

You must do the T/S parameter extraction immediately after setting this value. Enter the value, back up one menu via Esc and then press Extract to get the parameters. If you leave the AutoMeasure T/S menu area, it will revert to "Extract" mode for the DC resistance also. The driver diameter changes and stays changed more normally. Just press Inches and enter the number of inches effective diameter (usually 80% of nominal diameter), then Enter, and it should use it.

Distance is a utility which calculates acoustic distance by use of a measured phase curve. You can use it to find the apparent position of the speaker in different parts of the frequency range. As described above, the delay caused by sound going over a distance causes measured phase to slope downwards and "roll." The excess slope is proportional to delay. If you can assume that the speaker is minimum phase in a certain frequency region (most single drivers, without crossover, are minimum phase; almost no multidriver speaker systems are), you can calculate the drivers' phase response at the 0-delay reference plane using the Hilbert transform.

Distance calculates the Hilbert transform of the current frequency response plot to find the minimum phase curve. Then it figures out what kind of delay would need to be added to *get a phase slope as close as possible to that of the responses' measured phase curve slope, in the region between the two frequency markers.*

Try this: measure a tweeter or upper-midrange driver without crossover and use a Cal. Leave the #1 time marker at the far left of the trace, so true delay is measured, but use the #2 marker to get rid of echoes. Use FFT to frequency response, then set the two frequency response markers at the edges of the flat region of the measured response. Use Display Show Distance. If you do this correctly, the distance displayed should be very close to the distance between the microphone capsule and the drivers' pole piece (for a normal dynamic driver).

GOING PUBLIC

I read "A Public-Area Sound System" by Hilary Paprocki (*SB* 4/94, p. 12) with great interest. I applaud his involvement with the local community college and share some of his sentiments.

Since I designed the FA136 driver, I was especially pleased that he liked it and the results. This model was intended to be an advance in commercial sound, and his conclusions indicate we hit the target.

I would like to offer some information regarding the tweeter and crossover. Yes, some audiophiles seem to like a strong treble balance. I assume that to be a matter of personal taste, compensation for natural high-frequency hearing loss that comes with aging, or compensation for Fletcher-Munson loss at lower listening levels. Other audiophiles prefer the top end slightly rolled off for a "softer, more pleasant" sound.

Since personal tastes vary and the acoustical environment where the speaker will be used varies considerably as well, I designed the FA136 to be a flat response device. In anechoic measurements the response is ±3dB from 100Hz to beyond 16kHz. With the crossover modifications that Mr. Paprocki implemented, the response drops 6dB above 5kHz, which is fine if either personal taste or room acoustics are better served by so doing. Another user with different tastes and a room filled with drapes and overstuffed furniture might wish for more tweeter output.

I do not wish to imply that there is a right or wrong here except for how well a speaker pleases a listener in a given situation. I designed the FA136 to be flat so it might please tastes that fall in the middle in average acoustic environments. Given the hard ceilings, walls, and thin carpet described by Mr. Paprocki, I might well have made a similar change.

Michael E. Lamm
Design Engineer
Atlas/Soundolier
Fenton, MO 63026

Hilary Paprocki responds:

We are honored to be addressed by Mr. Lamm. He has designed an awfully nice driver, and no resonable person could find fault with his design philosophy.

It's particularly significant that he discusses context-based system voicing, since I was recently criticized in these pages for very distinctly increasing the treble output of another scientifically-tested "flat" loudspeaker system. If that doesn't underline his point, I don't know what would.

It's a wonder any commercial loudspeaker is able to find wide acceptance and success given the wide variety of end environments. All of this makes Mr. Lamm's FA136 all the more admirable an accomplishment. Frankly, I had hoped the designer of this driver would reveal himself so that I could congratulate him. Great work!

PORT ADJUSTMENT FORMULA

While browsing through some back issues, I noticed Pete Taylor's comments (*SB* 5/93, p. 16) about the tuning formula:

$$L_V = \Delta f_B \times 2L_V/f_B$$

He is not the first one to notice that it isn't very accurate. I have used this "safe" approach for years. When used in conjunction with the "bare box" formula, which always suggests a tube with excess length and a lower than desired frequency, this formula gives a ΔL_V that is too small. So you cut off a little, and then a little more. I have continued to use the formula, maybe because I used a slide rule until far into the calculator and computer era.

It occurred to me that others might not be so tolerant of rough estimates, so I took another look at port tuning. In checking box tuning, George Augspurger's VNTWRK program is ideal.* Running some tuning variations through it, I noticed that the formula was most erratic on short ducts. For a duct with the length equal to the diameter, the results are far off the mark.

Taking the duct diameter/length ratio into consideration, here is an alternative formula:

$$L_V = -\Delta f_B \times 2 \times (L_V + ((d/L_V)^2 \times L))/f_B$$

I compared the corrections and final L_V, using this formula, against VNTWRK and achieved the following results:

V_B ft³	d	L_V	f_B Hz	f_B^2 Hz	L_V new	L_V (VNTWRK)
0.5	1″	2″	37.1	40	1.61″	1.58″
0.5	2″	6″	45.44	50	4.66″	4.61″
1	2″	3″	40.64	45	2.07″	2.08″
1	3″	8″	41.1	45	6.27″	6.18″
2	4″	9″	35.64	40	6.36″	6.32″

Here is the result, rather typical, of the first example above, using the old formula:

$$0.5(V_B)\ \ 40(f_B)\ 2.36\ (L_V)$$

As you can see, 2.36″ is not very close to Augspurger's 1.58″.

The new formula works well with round ducts and is accurate for a duct of square cross-section when you use the equivalent diameter of a circle with the same area instead of the rectangular measurements. If you have a vent of rectangular shape that is not square, use the following formula:

$$f_B = -\Delta f_B \times 2 \times (L_V + ((1.13H/L_V)^2 \times L_V))/f_B$$

where H is the narrow dimension of the slot.

Here are some results:

V_B ft^3	Port	f_B Hz	f_B^2 Hz	L_V new	L_V (VNTWRK)
2	2″ × 8″	44.02	50	4.9″	4.79″
3	1″ × 16″	35.63	40	6.00″	5.93″
4	3″ × 6″	35.36	40	3.08″	3.13″

The formulas above work well for most vents, but they are not useful where $L_V < d$. Here is an example:

V_B ft^3	d	L_V	f_B Hz	f_B^2 Hz	L_V new	L_V (VNTWRK)
4	4″	0.75″	41.7	30	12.7″	5.17″

You are not likely to encounter this situation anyway.

If anyone has any other suggestions on this, I'd be glad to hear them.

David Weems
Box 99
Newtonia, MO 64853

**Available through SB's service division, Old Colony.—Ed.*

BACK TO THE FUTURE

Dick Campbell saw the Bose Auditioner© as a sign of audio's future ("Peering Into the Future," *SB* 2/95, p. 9). To me, it was déjà vu. This device suffers from the same fatal flaw that plagued nearly all of the previous audio image enhancement tricks: it does not allow the freedom of position and movement that listeners enjoy at live performances. People listen through loudspeakers because they want unconfined and unconnected listening. In this respect, the device is a total failure and represents no progress.

Mr. Campbell reported, "In most cases, the subject could not tell the difference" between the Bose Auditioner and the live sound over a limited bandwidth. Considering this statement and Philip Witham's articles describing his densely spaced multiple-channel array ("The Linear-Array Chronicles," *SB* 5/94, 8/94, and 1/95), I see a growing consensus that the weakest link in audio reproduction is in image reproduction rather than the fidelity of the channels. We are reaching a point of diminishing returns for the improvements in distortion and bandwidth, but the subject of imaging has not been widely explored outside of the limits of two channels. Now that we are free of the constraint that held us to two channels, the LP, we can move forward.

Necessity is the mother of invention, and freedom of listening position is a commercial necessity in a movie theater. So far, the most successful advancements in this area have come from the engineers at Dolby Laboratories rather than the patent attorneys at Bose. The Bose people deserve credit for making better sound more marketable through improved packaging, but Dolby extracted stereo sound with wide dynamics from dictation tapes and surround sound with a wide listening area from two channels. Dolby raised the ceiling while Bose raised the floor of sonic performance. Dolby also proselytized its technology rather than miserly hoarding it and stifling experimentation.

It took several decades for stereo to replace monophonic recording as the mass market's standard. After several more decades, home theater has become synonymous with more than two channels. As home theater systems improve, the desire to play music over them will force the 5.1 channel standard on the music industry. After all, opera and ballet are a form of multimedia experiences. Dolby's AC3 is the product signaling audio's future, but the most successful products allow a single software inventory and let the consumer decide when and if to upgrade. With the means available to produce a 5.1 channel CD with negligible loss of performance when played on existing equipment, I predict 5.1 channel systems will become the next standard.

Dennis L. Green
Detroit, MI 48235

Contributing Editor Richard Campbell responds:

In my short editorial I may have used words suggesting that the Bose Auditioner was some sort of "audio image enhancement trick." It is certainly not that. The aural image produced in the space around your head when you are in the chin rest is done by digital signal processing techniques whose foundation is based on absolutely solid acoustical engineering and mathematical modeling. It is no trick any more than the Fourier transform is a trick to flop from the time domain to the frequency domain in signal analysis.

Indeed, the device does not allow the listener to move around because that is not its purpose. Any significant head movement invalidates the algorithms used to do the real-time convolution for that exact seat location. The digital input data in the time domain is derived from the collection of rays arriving at a particular receiver from known angles in a geometric room computer model created using the Bose Modeler© program. If you moved to a different location in the auditorium, then another appropriate receiver would need to be modeled and then selected for auditioning.

The Bose Auditioner is intended to be used by acousticians and sound system designers to auralize a seat in a hall whose architecture may exist only in a computer. It is a design and troubleshooting tool, not a consumer product such as home theater systems. Acoustical designers need reverberation accuracy (and accurate early reflections, initial time delay gap, EDT, T-20, LEF, IACC, C80, STI, and so forth), as opposed to simply bathing in the wash of Dolby 5.1© or any other entertainment system. Those are fun and enjoyable at home, but they are not work.

In home theater systems, the sound tends to back up the picture since our visual sense is absolutely supreme over our aural sense. This has been shown in numerous psychological experiments where the senses are "crossed" and the visual cues take over after a tiny vertigo-like interval. I was simply "peering into the future" of acoustic measurements and hall evaluation, in my editorial, and not into the wonderful world of surround sound, THX, Dolby 5.1, and all that is exciting in today's virtual audio environment.

SLM MOD

Thank you for providing "knowledge-thirsty audio types" with a fine publication. Your insightful articles and the opportunity to profit from other audio experimenters' experiences are priceless assets.

I would like to point out an error in the schematic in "Converting Radio Shack's Sound Level Meter to Millivolt Use" (*SB* 4/94, p. 12). Author Carrington's additional circuit shows a 15pF capacitor, which he later corrected in *SB* 6/94 (p. 53) to 150pF connected in series with the RCA connector. The text correctly indicates that the capacitor should be connected between the RCA positive connection and the wiper of the trimmer resistor as a shunting capacitor.

I modified my Sound Level Meter according to the text, but ignored the schematic (*Fig. 1*). The meter is now a useful companion to Mr. D'Appolito's microphone. Before reading the article, I used the R/S meter as supplied, except for the addition of a mini audio jack. The audio connector provided access to the meter's internal amplifier while bypassing the built-in microphone when a 3.5mm male audio plug was connected to the mini audio jack without the need for a switch. This allowed me to read the output of Mitey Mike on the sound level meter only as uncalibrated relative measurements due to the extra gain provided by Mitey Mike.

I am very pleased with the results from this simple modification, but a clarification is in order to encourage those who found it difficult to successfully modify and calibrate their own meters.

Publio Morera
Merrick, NY 11566

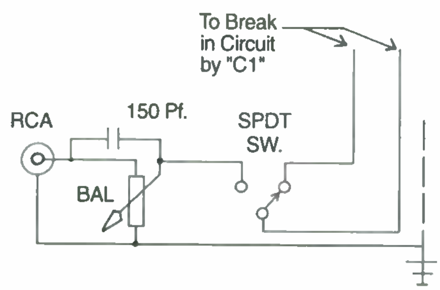


FIGURE 1: Reader Morera's correction for the Sound Level Meter.

C.L. Paul Carrington responds:

I am pleased Mr. Morera found the article on the modification of Radio Shack's SLM useful. My letter published in SB 6/94 corrected the value and location of the capacitor in the schematic to comply with the article text, and Mr. Morera's diagram (Fig. 1) will remove any doubt about what was intended. His use of a closed-circuit mini jack is a useful simplification. I use RCA phono jacks as standard audio input and output connectors for my home-built test equipment so, in my case, the switch was necessary.

WHERE'S THE 1570B BEEF?

My friend purchased a pair of Altec 1570Bs. Can you provide any information on upgrading these units, or explain the trans input or strapping the output?

Renee May
Los Angeles, CA 90038

Readers who can help with information to these questions can write to Speaker Builder, *Editorial Dept., PO Box 494, Peterborough, NH 03458, FAX (603) 924-9467.*

Technology Watch

END-FIRE RADIATORS

By Peter Muxlow

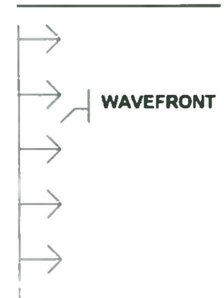
When speech reinforcement is needed in a reverberant environment, the loudspeakers require highly directional properties. This focuses the sound and minimizes the spill onto reflecting surfaces—walls, ceiling, and floor—which would add to the reverberation.

Sound reinforcement systems can be divided into two groups: high-level distribution, in which a single loudspeaker cluster (or a small number) radiates over the entire area with a relatively high sound pressure level, and low-level, where a large number of units operates at a relatively low sound pressure. Normally, horns or line-source loudspeakers (a series of speakers arranged in a line) are used because of their directional properties. If a large number of units is needed, however, this can become expensive. One low-cost alternative is the end-fire array.

PATTERN MAKING

If we apply the same signal to each unit of a line-source loudspeaker, the sound radiated will have a directional pattern dependent upon the number of speakers, the length of the line, and the wavelength of the sound. A radiation pattern is created by the destructive and constructive interference between the speakers. With a vertical line or column, the polar pattern is broad in the horizontal plane and shallow in the vertical plane.

By introducing a



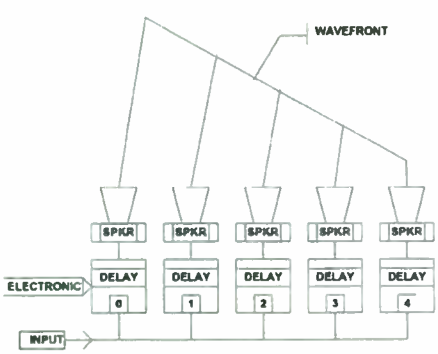


FIGURE 1: Altering the direction and shape of the wavefront by introducing a time-delayed signal.

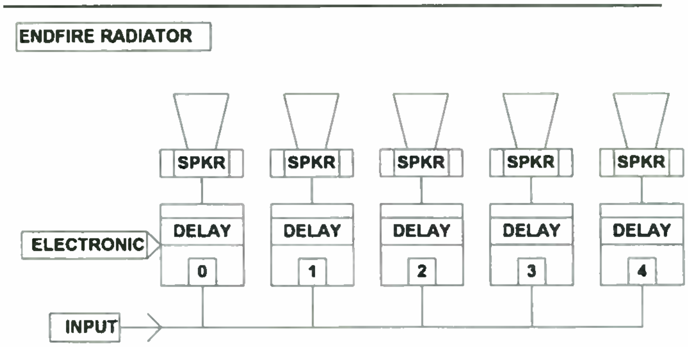


FIGURE 2: If the electronic delay is made equal to the time for the radiated sound wave to travel from one speaker to the next, the radiated waves will add phase.

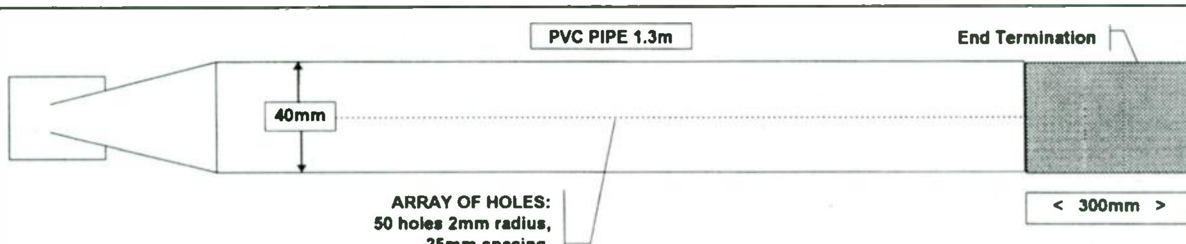


FIGURE 3: Horn compression driver TOA TV25L, 1W input, 130dB output into a plane wave tube. The End Termination is a wedge of open-cell plastic foam which reduces reflections back up the pipe. Input level of 120dB to the radiator (pipe) produces 70dB SPL output at 3.5m.

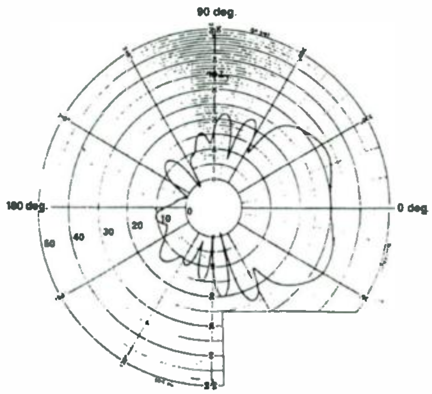


FIGURE 4: Polar pattern of prototype: 50 holes of 2mm radius and 25mm spacing at 800Hz. Measured directivity.

time-delayed signal to each unit, we alter the direction and shape of the wavefront (*Fig. 1*). When the time delay equals the time the radiated sound wave takes to travel from one loudspeaker to the next, the sound waves add phase to the line and radiate off the end of the array. Hence the term end-fire radiators (*Fig. 2*). The reciprocal transducer is the directional line, or rifle microphone, developed by Olson.

To create this radiation pattern with one loudspeaker, connect a compression horn driver to a pipe which has holes drilled in it (*Fig. 3*). Each hole acts as an individual driver or source. The time delay required between the sources is provided by the velocity of sound inside the pipe—the same speed as the outside wave propagation (344ms-1).

Holland and Fahy have developed a mathematical model using these principles.[1] It predicts the on-axis frequency response and directivity. The authors have achieved a good correlation with the prototype of *Fig. 3* and their mathematical model.

The parameters for the prototype, operating as a low-level distribution system, are:

1. A 60dB SPL at the listener's ears. With a sound pressure level of 120dB at the input of the radiator (the pipe), it radiated 70dB SPL at a distance of 3.5m.

2. A frequency range of 1–5kHz, with the polar pattern shown in *Fig. 4*.

This seems to be a good, low-cost alternative. If you wish to design your own, Holland and Fahy's article provides the necessary data, with other ideas for achieving similar results. A suitable driver might be the Motorola KSN1188A piezo, described in *SB* 1/95 ("High Quality Use of Motorola's Piezo Driver," p. 18).

REFERENCE

1. K.R. Holland and F.J. Fahy, "A Low Cost End-Fire Acoustic Radiator," *JAES*, July 1991.

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Horn Design Program

from page 19

above, there are several variables you must consider, based on your knowledge of horn design. I advise you to consult the references and any horn literature mentioned in previous issues of *Speaker Builder*. Most important of all, some horn enthusiasts willing to exchange and share ideas are a good source of information.

I'd like to thank Dr. Edgar for his help in writing this article and also Brian Smith for sparking my interest in computer-assisted horn design.

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Speaker Builder needs reader feedback in the form of letters, questions, queries, and tips. Tell us your problems and concerns, your reaction to articles, your latest audio construction adventure. Our departments answer reader questions and let you share your knowledge and experience or show off your handiwork. Whether you're seeking help with a problem or offering a solution, *SB* wants to hear from you. Remember, it's your magazine, and your contribution can help make it even better. Send your entries to:

Speaker Builder
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Madisound BBS

Continued from page 34

on the Madisound BBS, so if I overlooked your favorite program or a better choice in a particular category, please forgive the oversight. The "Letters" column of *SB* is the ideal place for you to comment on your experiences with these programs.

Now it's your turn to explore the Madisound BBS files, so turn on your computer, dial the BBS at (608) 836-9473, and log on for some of the nicest, free programs available anywhere. It's like having Christmas all year round!

CLIO Review

Continued from page 53

place of a steady-state sinusoid.

Mauro Bigi and Maurizio Jacchia
Audiomatica s.r.l.

AUTHOR'S RESPONSE

After reading Audiomatica's comments on my report, I noticed that the term "acoustical" is somewhat vague, so I'll try to explain it.

It's perfectly true that, in this case, the FFT resolution is 6.25Hz and this is a mathematical fact. But with a 10ms time window, the theoretical lowest significant frequency is 100Hz. The 8,192-point FFT gives you a 6.25Hz resolution (at 51.2kHz sample frequency). You have an uncertainty of 100Hz. What this means is that between 300Hz and 500Hz the amplitude accuracy is only three points. Again, this is not a CLIO limitation, but applies to all systems.

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Pair JBL 2235H, $260; pair Dynaudio D21AF, $50; Focal 8K415S, $50; pair Aria3 cabinets, $100. Jerry, (601) 264-6971.

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SPEAKER BUILDER CLASSIFIED ORDER FORM

TRADE— for any business or private party, selling equipment, supplies, or services *FOR PROFIT.* **FOR SALE**— to sell ***personal*** equipment or supplies. **WANTED**— to find ***personal*** equipment, supplies, or services. **CLUBS**— to help readers start and maintain club membership.

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QUESTIONS? Write or fax us at the address below.

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16	17	18	19	20
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26	27	28	29	30
31	32	33	34	35
36	37	38	39	40
41	42	43	44	45
46	47	48	49	50

Please charge to my: ❑ DISCOVER ❑ MASTERCARD ❑ VISA

CARD NUMBER EXP. DATE

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NAME

COMPANY

STREET & NO.

CITY STATE ZIP

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PO Box 494, Peterborough, NH 03458

FAX 603-924-9467
Data 603-924-6230

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CLASSIFIED INFORMATION

Four types of Classified Advertising are available in *Speaker Builder*:

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WANTED: Help readers find equipment or services.

TRADE: For any business or private party selling equipment, supplies or services for profit.

CLUBS: Aid readers in starting a club or finding new members. Specific guidelines apply to *Club* advertising. Please write to the Ad Department for terms. Don't forget, include a #10 self-addressed stamped envelope.

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Receive *free For Sale* and *Wanted* ads, 50 word maximum, each additional word $.20, $10 minimum. For *Club* ads, follow instructions at the bottom of the Clubs listings. Please submit only one ad per category per issue.

Please include your name, address, and telephone number. If TRADE please indicate number of insertions on the ad. All free ads are run only once, and then discarded. ***Ad questions, copy and copy changes cannot and will not be answered on the phone.*** All correspondence must be in writing addressed to:

Speaker Builder
Classified Department
PO Box 494
Peterborough, NH 03458-0494

Audio Control C-131 equalizers, Pioneer series 20 crossover. Joe, (617) 254-0697, (Boston, MA).

Dynaudio D28af, 17W75, Scan-Speak D2905, 18W8544, 18W8543, original QUAD tweeter panel. Richard Konopaski, 644 County Road 8, Stapels, ON Canada N0P 2J0, (519) 687-2040.

Schematics or service info needed to repair Gelosa (Italian) model #G.1/1110-U tube power amp. Will pay any costs incurred to copy, fax, etc. Thank you. Don, (609) 768-2439 days or (609) 665-6512 eves.

Bang & Olufsen model 200 mike. California Institute of Technology Music Lab needs gifts of three B&O stereo ribbon mikes with undamaged ribbons. Before shipping, contact James Boyk, (310) 475-8261, FAX (310) 470-9140, E-mail: boyk@ccosun.caltech.edu.

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Philips/Magnavox CDV488 player. Robert Irwin, 936 Pearson Rd., Paradise, CA 95969, (916) 877-7734 (leave message).

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Ad Index

CLUBS

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

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PO Box 13058, Scottsdale, AZ 85267
Paul Christos, (602) 971-3979

CALIFORNIA

Tube Audio Enthusiasts
65 Washington St. #137,
Santa Clara, CA 95050
John Atwood, FAX (408) 985-2006

Greater South Bay Audiophile Society
Larry Fisher, (310) 599-6579;
Dave Clark, (310) 427-4207

Los Angeles Loudspeaker Designers Group
Geoffrey, (213) 965-0449;
Eduard, (310) 395-5196

West Valley Audio Society
Woodland Hills
Barry Kohan, (818) 225-1341

COLORADO

Colorado Audio Society
1941 S. Grant St.,
Denver, CO 80210
(303) 733-1613

CONNECTICUT

James Addison
171 Hartford Rd., A-7,
New Britain, CT 06053

Connecticut Audio Society
129 Newgate Rd., E. Granby, CT 06026
Richard Thompson, (203) 653-7873

FLORIDA

Angel Rivera
Bradenton, FL 34206
(813) 792-3870

GEORGIA

Atlanta Audio Society, Inc.
4266 Roswell Rd. N.E., K-4,
Atlanta, GA 30342-3738
Chuck Bruce, (404) 876-5659;
John Morrison, (404) 491-1553

ILLINOIS

Chicago Audio Society
PO Box 313, Barrington, IL 60011
Brian Walsh, (708) 382-8433,
E-mail sysop@nybble.com

Prairie State Audio Construction Society
20 Wildwood Tr., Cary, IL 60013
Tom, (708) 248-3377 days,
(708) 516-0170 eves.

MASSACHUSETTS

The Boston Audio Society
PO Box 211, Boston, MA 02126-0002
E. Brad Meyer, phone/FAX (617) 259-9684,
E-mail 72365.75@compuserve.com

MICHIGAN

Southeastern Michigan Woofer and Tweeter Marching Society
PO Box 721464, Berkley, MI 48072-0464
(810) 544-8453, E-mail
aa259@detroit.freenet.org

MINNESOTA

Audio Society of Minnesota
PO Box 32293, Fridley, MN 55432
(612) 825-6806

NEVADA (Las Vegas)

Tarcisio Valente, (702) 435-7451

NEW JERSEY

New Jersey Audio Society
209 Second St., Middlesex, NJ 08846
Frank J. Alles, (908) 424-0463; Valerie Kurlychek, (908) 206-0924

NEW YORK

The Audio Syndrome
Nassau and Suffolk counties
Roy Harris, (516) 489-9576

Catskill and Adirondack Audio Society
PO Box 144, Hannacroix, NY 12087
(518) 756-9894

Daniel Phinney
29 Brian Dr., Rochester, NY 14624
(716) 594-8118, 5-9 p.m. EST

Long Island/Westchester County
Publio Morera, (516) 868-8863

Western New York Audio Society
PO Box 312, N. Tonawanda, NY 14150
Denny Fritz

NORTH CAROLINA

Piedmont Audio Society
1004 Olive Chapel Rd., Apex, NC 27502
Kevin Carter, (919) 387-0911

TENNESSEE

Memphis Area Audio Society
8182 Wind Valley Cove, Memphis, TN 38125
J.J. McBride, (901) 756-6831

TEXAS (Sweetwater)

Rick, (915) 676-7360

UTAH (Salt Lake City)

Wasatch Audio
Edward Aho, (801) 364-4204

WASHINGTON

Pacificnorthwest Audio Society
Box 435, Mercer Island, WA 98040
Ed Yang, (206) 232-6466;
Gill Loring, (206) 937-4705

Vintage Audio Listeners and Valve Enthusiasts
1127 NW Brite Star Ln., Poulsbo, WA 98370
(206) 697-1936

CANADA

Greg Boboski
70 Sumter Cres., Winnipeg, MB R2T 1B9
Canada
(204) 694-7024

Montreal Speaker Builder Club
4701 Jeanne Mance, Montreal,
PQ, H2V 4J5 Canada
Andrew McCrae, (514) 281-7954

CHILE

Christian Bargsted
Los Gomeros 1542, Vitacura, Santiago
(011) 562-538-0638

GERMANY

ESL Builders Group
Gunter Roehricht
Bühlerstr.21, 71034 Böblingen, Germany

SOUTH AFRICA

Hi-Fi Club of Cape Town
PO Box 18262, Wynberg 7824, South Africa
Chris Clarke, FAX (011) 27-21-7618862, E-mail chrisc@iaccess.8a

UNITED KINGDOM

London Live DIY Hi-Fi Circle
Brian Stenning, (011) 44-81-748-7489

If your club is not represented or if you are interested in starting up a club, simply send the information (club name, contact person, address and phone/FAX number) to:

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Clubs
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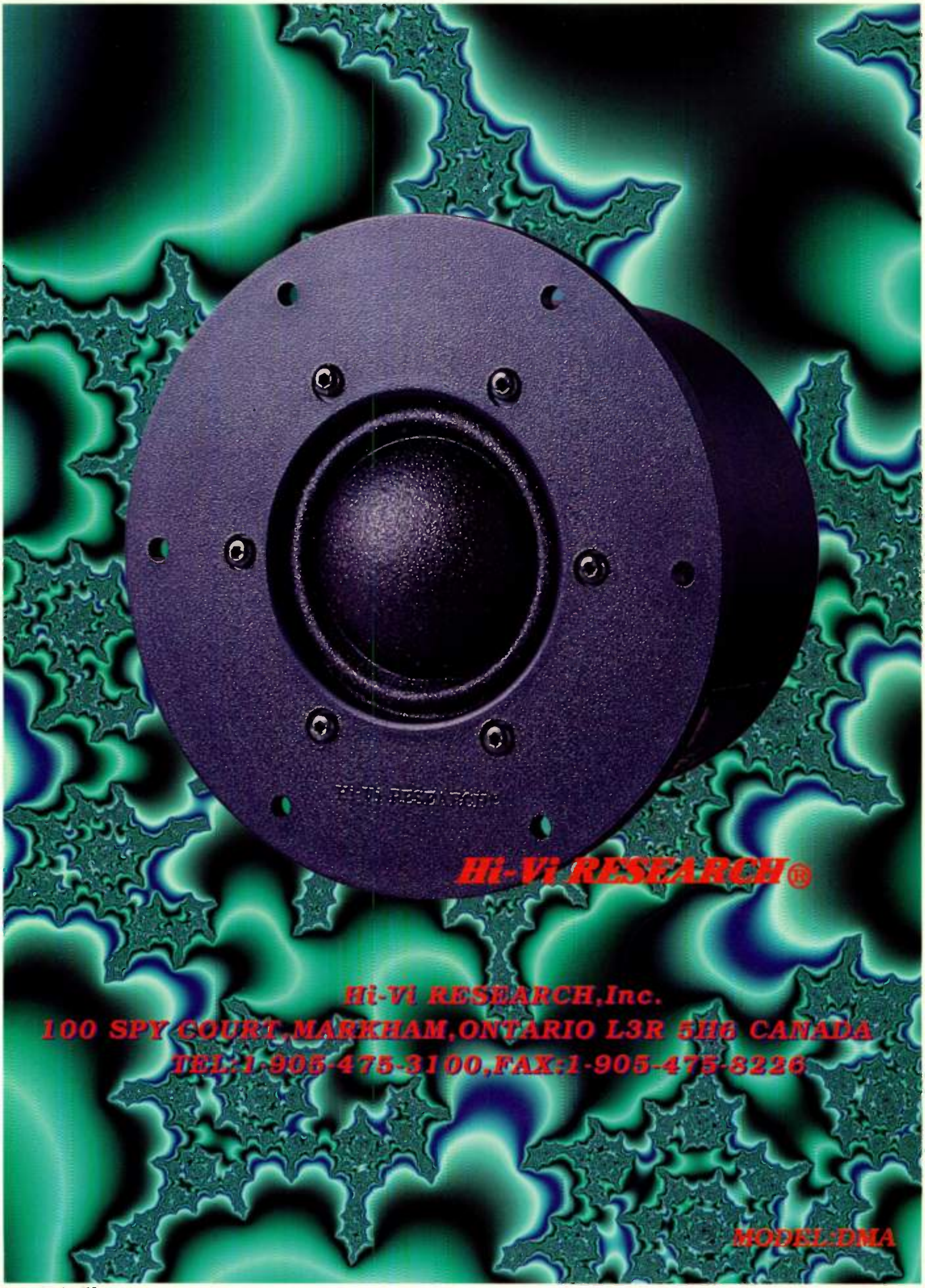
Peterborough, NH 03458
FAX (603) 924-9467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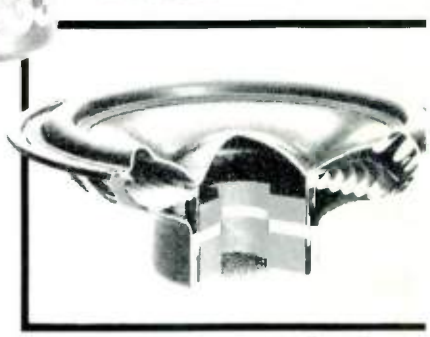

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Brookline, MA 02146
Tel: (617) 277-6663
FAX: (617) 277-2415

Reader Service #1