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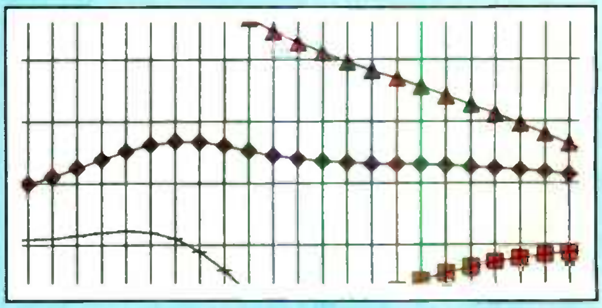
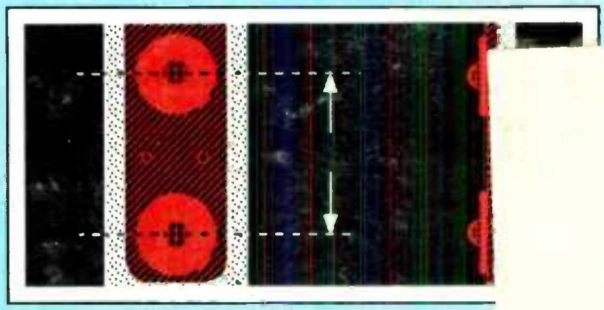
SpeakerBuilder
THE LOUDSPEAKER JOURNAL

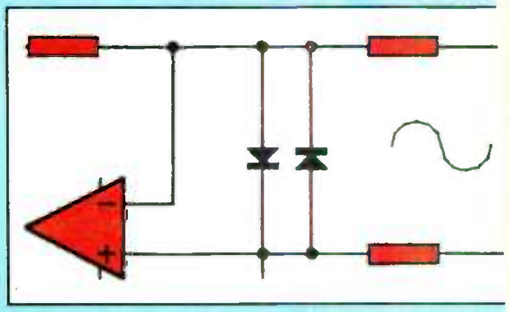
BETTER SONICS FOR CHURCH
—Bacon



UPGRADES FOR VINTAGE MAGNEPANS
—Seymour



CONVERSION FOR LOW SIGNAL MEASUREMENTS
—Silva



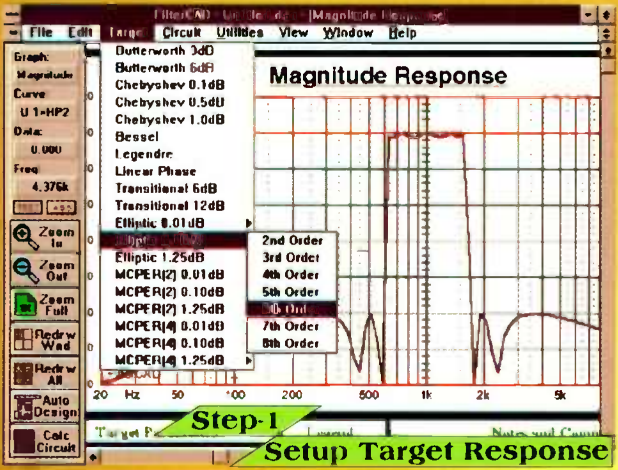
ELECTRONIC CROSSOVERS GALORE
—Borbely & Gaertner

New

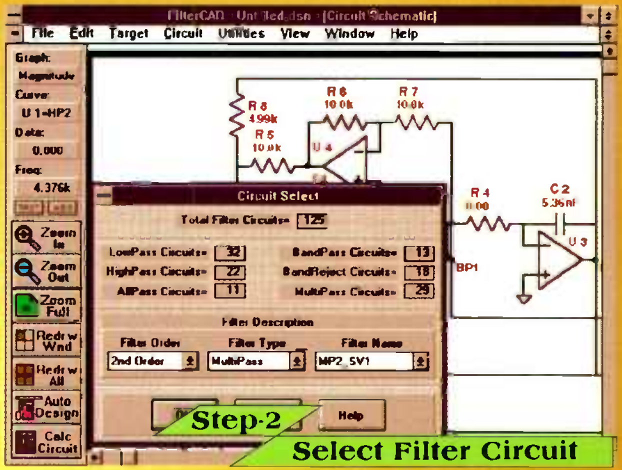
FilterCAD

Design Active Filters in 3 Easy Steps !

FilterCAD is a program specifically developed to handle the unique requirements of active filter design. In the past, filter designers had to rely on tables and equations of filter design data, or use trial-and-error analysis with general circuit simulation programs. FilterCAD provides an entirely new approach- *direct design*. FilterCAD contains all of the synthesis equations necessary to actually *design* the component values itself, in addition to providing a full target generation system for accurate comparison. With FilterCAD, designing simple or complex multi-stage filters is an easy and *very fast* 3 step process!



Step-1
Setup Target Response



Circuit Select

Total Filter Circuits= 125

LowPass Circuits= 32
BandPass Circuits= 13
HighPass Circuits= 22
BandReject Circuits= 18
AllPass Circuits= 11
MultiPass Circuits= 29

Filter Description

Filter Order: 2nd Order
Filter Type: MultiPass
Filter Name: MP2_SV1

Step-2
Select Filter Circuit

Filter Circuit Topologies

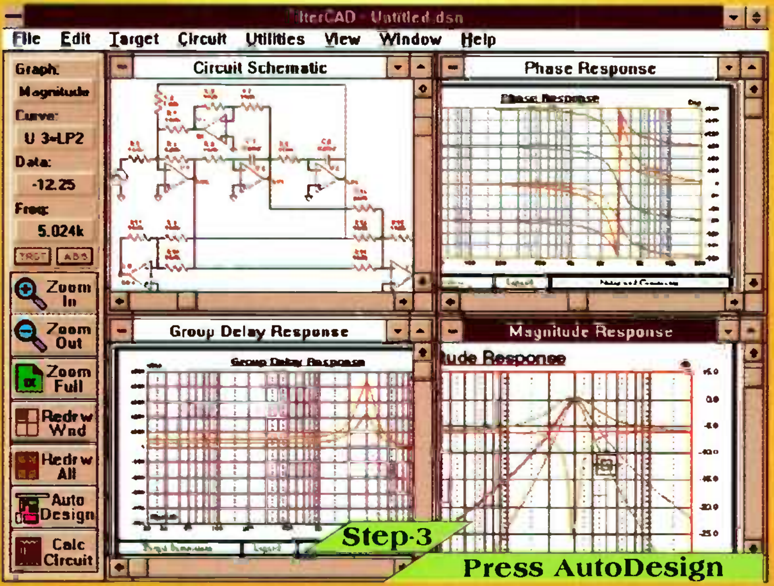
FilterCAD contains a catalog of predefined circuit topologies, from which the user can choose a particular circuit or circuits for a given design. The design equations and filter synthesis information for each of these circuits has been developed and coded into the program, which enables FilterCAD to actually design the circuit itself based on a few key component choices by the user.

Filter Circuits

- 125 different circuit topologies, covering 1st through 8th order filter designs and more.
- User controllable two-pole Op-Amp model.
- Unlimited cascade design.
- Multiple-Feedback-Loop filters to 8th order.
- RLC ladders including Elliptics to 8th order.
- RDC ladders using FDNRs to 8th order.
- Gyrator synthesis for equivalent 'L' and 'D'.
- 1st-4th order state variables and biquads.
- RLC Allpass lattice circuits.
- Twin-Tee Bandpass and Bandreject circuits.
- Wein Bandpass Bandreject circuits.
- Asymmetrical LPN/HPN bandreject circuits.
- 1st-4th order Sallen-Key LP/HP AP BP BR.
- Many other 1st and 2nd order circuits.

System Features

- Standard values: any, 1%, 5%, 10%, 20%.
- Circuit Impedance Scaling.
- Unlimited frequency range.
- User controllable analysis resolution.
- User controllable scale design.
- Custom graphs, fonts, line widths, colors.
- ABS/REL cursor readout system.
- ASCII data import / export.
- Graphics raster and vector export.
- SPICE net list generation.



Step-3
Press AutoDesign

Target Generation System

FilterCAD contains a full target creation system which enables the user to instantly generate a desired response for a particular filter design. The target response is then displayed on all magnitude, phase, and group delay graphs. Built-in standard classical filter functions are provided with automatic calculations for any transformation and frequency.

Custom Target Controls

- Magnitude, Phase, and Group Delay.
- TF Poly Order: 1-16 poles and zeros.
- Transfer Function Blocks (TFBs): 8 Max.
- TFB Parameters: Ao, Fp, Qp, Fz.
- LP1,HP1,AP1,LP2,HP2,AP2,BP1,BR1.
- TFB Enable Disable switches.
- Automatic target leveling to circuit data.

Standard Target Functions

- 1st-8th order filter functions.
- Full transformations: LP,HP,AP,BP,BR.
- Butterworth 3 dB / 6 dB (Linkwitz/Riley).
- Chebyshev 0.1 dB / 0.5 dB / 1.0 dB ripple.
- Linear Phase family.
- Bessel family.
- Legendre family.
- Transitional 6 dB / 12 dB cutoff.
- Elliptic 0.01 dB / 0.1dB / 1.25 dB ripple.
- MCPER(2) 0.01 dB / 0.1 dB / 1.25 dB.
- MCPER(4) 0.01 dB / 0.1 dB / 1.25 dB.

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System Requirements:
MS Windows 3.1, 4MB RAM minimum.
VGA or higher video card resolution.
Math Coprocessor recommended.

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Portland, OR 97224 USA
Tel: (503) 620-3044
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Good News

M&P

All Selenium products (woofers, horn drivers, horns and tweeters) are now available from M&P. The Pro Unit Woofers (12″, 15″, 18″) feature 1kW continuous power handling with up to 100dB SPL (1W/1M). Factory computer-aligned set screws allow for replacement of cone, spider, former, and voice coil. Selenium also manufactures customized speakers. M&P Technologies, Inc., 75 E. Uwchlan Ave. #128, Exton, PA 19341, (800) 355-0500, FAX (215) 524-5531.

Reader Service #115

SASAKI

Dr. Harry Olson determined that "ideal sound comes from a spherical speaker enclosure." Sasaki Acoustics' spherical glass "Clearball" allows uniform sound radiation plus extra freedom in speaker placement. Cyclops Distributors, 600 N. 12th St., Reading, PA 19604, (215) 374-4760, FAX (215) 478-9475.

Reader Service #109

M&K

The S-90 speaker system was designed to achieve a timbre match with other front-channel speakers to avoid the sonic discontinuity that occurs when sound pans across the left, center, and right channels of unmatched speakers. Miller & Kreisel Sound Corp., 10391 Jefferson Blvd., Culver City, CA 90232, (310) 204-2854, FAX (310) 202-8782.

Reader Service #102

IMPULSE

The H6 horn loudspeaker incorporates the same high-frequency unit used in the H1 and H2 models with compact design. Cabinet dimensions are 36″ × 7.5″ × 14″. Nominal impedance is 8Ω with sensitivity of 89dB (2.83V at 1M). Low and midfrequencies are produced by a doped paper coned drive unit that is back loaded by a horn. Impulse Loudspeakers, 5 High Parade, Streatham High Road, London SW16 1EX, (081) 769-5726, FAX (081) 769-0353.

Reader Service #103

POLYDAX

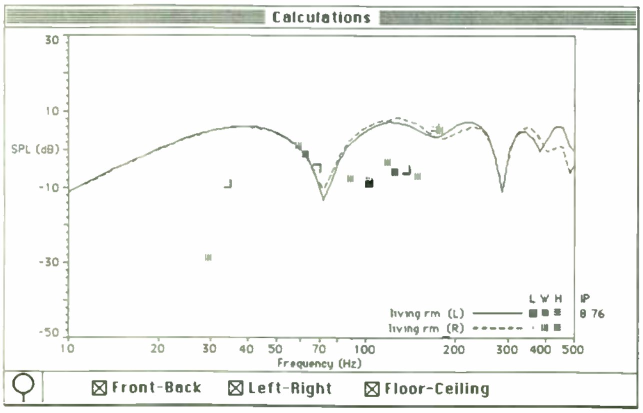
The latest line of woven Kevlar cone mid-bass speakers includes the HT100K0 (4″), HT130K0 (5¼″), HT170K0 (6½″), and HT210K0 (8″). Design features include high-loss rubber surrounds, large (20-oz.) magnet structures, and high-temperature voice coils wound on aluminum formers. Polydax Speaker Corp., 10 Upton Dr., Wilmington, MA 01887, (508) 658-0700, FAX (508) 658-0703.

Reader Service #101

SITTING DUCK SOFTWARE

Listening Room, a new program for Macintosh computers, allows users to position listener and speakers interactively and displays the magnitudes of standing waves, plus the effects of 124 early reflections on the direct response. Local optimization of listener and/or speaker positions and other new features. Sitting Duck Software, PO Box 130, Veneta, OR 97487, (503) 935-3982.

Reader Service #106



☒ Front-Back ☒ Left-Right ☒ Floor-Ceiling

SCANTEK

The new brochure on the precision Type 116 Sound Level Meter from Norsonic describes features and specifications. With an 80dB dynamic range, the meter simultaneously measures peak and RMS values of A- and C-weighted noise levels. Data is presented both numerically and graphically, and the unit is PC compatible. Scantek, Inc., 916 Gist Ave., Silver Spring, MD 20910, (301) 495-7738, FAX (301) 495-7739.

Reader Service #113

← ***Reader Service #36***

Good News

AUDIOCONTROL

The C-101 Series III octave equalizer has a built-in digital pink-noise test generator and a real-time audio spectrum analyzer. Octave-spaced sound controls allow for adjustment while pairing left and right sliders. The unit, which comes with a calibrated microphone and features an 18dB/octave Chebychev alignment subsonic filter, can be connected to any home stereo. AudioControl, 22313 70th Ave. W., Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.

Reader Service #107

MCINTOSH

Home theater with Dolby Pro Logic® and Home THX Audio are now in production, capable of reproducing movie sound tracks and stereo music with equally high quality. McIntosh Laboratory, Inc., 2 Chambers St., Binghamton, NY 13903-2699, (607) 723-3512, FAX (607) 724-0549.

Reader Service #110

■ TRI-STATE

The new catalog of home stereo, automotive, and professional speakers, parts and accessories, refoaming and reconing services is now available. Normally a $1 value, the catalog is currently being offered free to any speaker builder who calls. Tri-State Loudspeaker, 650 Franklin Ave., Aliquippa, PA 15001, (412) 375-9203.

Reader Service #112

■ DZURKO

Dzurko Acoustics is a new "High End" division of Audio Concepts. Its first product is the Shadow, having a unique "enclosure within an enclosure" upper cabinet module that eliminates out-of-phase leakage. Design refinements and attention to detail include an elegant hand-veneered enclosure and hand-damped, cloth dome tweeters and midrange drivers. Dzurko Acoustics, 901 S. 4th St., LaCrosse, WI 54601, (608) 784-4570, FAX 784-6367.

Reader Service #104

■ SCANTEK

Norsonic now offers a lightweight precision sound calibrator, Type 1251, providing a signal of 1kHz at 114dB independent of microphone equivalent volume, ambient temperature, and atmospheric pressure and humidity. Scantek, Inc., 916 Gist Ave., Silver Spring, MD 20910, (301) 495-7738, FAX (301) 495- 7739.

Reader Service #116

OPTOELECTRONICS

The new M1 frequency counter, with ten user-selectable gate times, is capable of ultra-high-speed measurements and high-resolution 10-digit measurements. The pocket-size counter is virtually hands free for frequency finding of 10Hz–2.86Hz. Optoelectronics, Inc., 5821 NE 14th Ave., Fort Lauderdale, FL 33334, (800) 327-5912, FAX (305) 771-2052.

Reader Service #111

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Continued on page 6

Good News

Continued from page 4

MOTOROLA
An improved midrange 100W horn driver incorporates protective circuitry that allows it to be used safely at continuous power levels of 400W. Frequency response: 800Hz–20kHz. Motorola Ceramic Products, 4800 Alameda Blvd. NE, Albuquerque, NM 87113, (505) 822-8801, FAX (505) 822-8812.
Reader Service #105

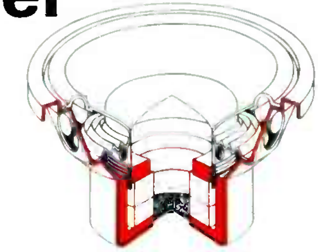
CAIG
Pro-Gold is an environmentally safe aerosol for cleaning, lubricating, and protecting gold, base metals, and other precious-metal connector surfaces. Its nonabrasive/noncorrosive formula enhances conductivity, reduces intermittent connection problems, and increases transmission quality. Caig Laboratories, Inc., 16744 W. Bernardo Dr., San Diego, CA 92127-1904, (619) 451-1799, FAX (619) 451-2799.
Reader Service #108

B&W
The Matrix 803 Series 2 loudspeaker features three 6″ drivers and a sloping shelf; cabinet dimensions have also been reduced. Frequency range is 20Hz–22kHz; sensitivity is 90dB SPL (2.83V/1M); nominal impedance is 8Ω. B&W Loudspeakers of America, PO Box 653, Buffalo, NY 14210, (800) 387-5127.
Reader Service #114

Reader Service #14 →

MW 114-S

Neodymium Magnet DPC Cone 4" Woofer



Specification

Overall Dimensions	Ø118mm (4.64") x 58mm(2.29")
Mounting Baffle Hole Diameter	Ø95mm (3.75")
Magnet System	Pot Type, Vented, Neodymium Magnet
Nominal Power Handling (Din)	150W
Transient Power - 10ms	800W
Voice Coil Diameter	54mm (2.125")
Voice Coil Type/Former	Hexatech Aluminium
Frequency Response	55-7000 Hz
FS - Resonant Frequency	65 Hz
Sensitivity 1W/1m	87 dB
Z - Nominal Impedance	8 ohms
RE - DC Resistance	5.6 ohms
LBM - Voice Coil Inductance @ 1kHz	0.47 mH
Magnetic Gap Width	1.25mm (0.050")
HE - Magnetic Gap Height	6mm (0.236")
Voice Coil Height	12mm (0.472")
X - Max. Linear Excursion	3mm
B - Flux Density	0.88T
BL Product (BXL)	6.75
Qms - Mechanical Q Factor	2.32
Qes - Electrical Q Factor	0.36
Q/T - Total Q Factor	0.31
Vas - Equivalent Cas Air Load	3.18 litres (0.113 cu. ft.)
MMS - Moving Mass	7.00gm
CMS	807μm/n
SD - Effective Cone/Dome Area	53cm² (20.86 sq. in.)
Cone/Dome Material	DPC (Damped Polymer Composite)
Nett Weight	0.500 kg

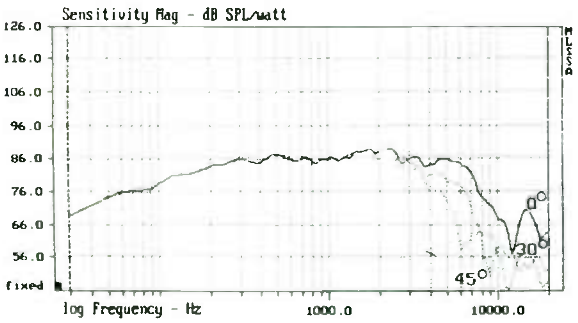
Specifications given are as after at least 45 minutes of high power, low frequency running, or 24 hours normal power operation.

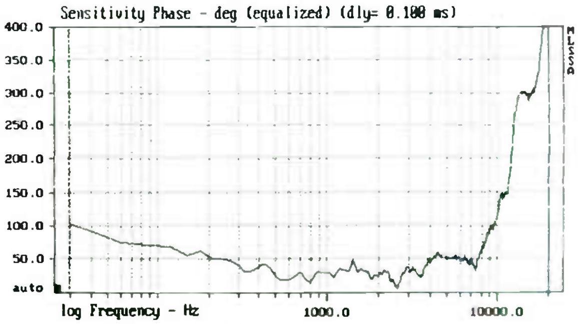
The 114-S is the first of Morel's new generation of woofers, featuring a powerful Neodymium magnet system which provides increased sensitivity, lower Qt and reduced distortion. For a 4" driver it is unique in having a large 54mm (2.125") diameter Hexatech aluminium voice coil.

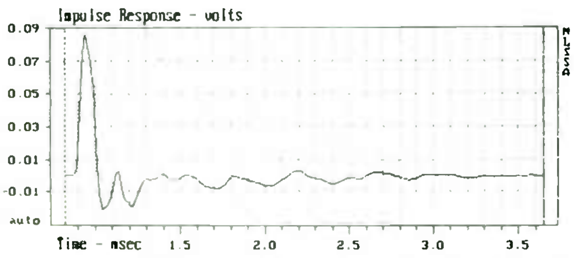
Benefits of this large voice coil diameter include a very high power handling capacity and lack of sound level compression. In addition, it allows the use of a very shallow cone profile. Coupled with the use of Damped Polymer Composite cone material and a rubber surround, this provides excellent dispersion (off-axis response), resistance to cone break-up (even at high sound pressure levels) and lack of colouration.

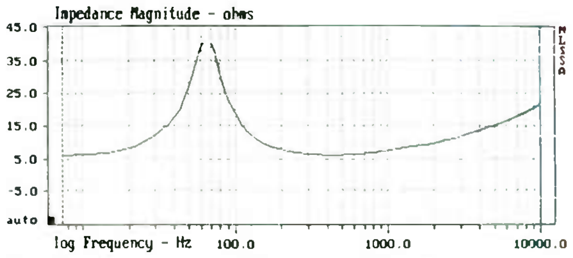
Frequency and phase response are very flat, while the roll-off is very smooth. The MW 114-S may be used either as a bass-mid range in 2-way systems, or as a mid-range in multi-way systems.

The vented magnet system is encased within a steel chassis, which improves efficiency and shields the magnet, virtually eliminating stray magnetic fields. The MW 114-S is ideal not only for high quality hi-fi, but also TV, video and surround-sound applications.









Morel operate a policy of continuous product design improvement, consequently specifications are subject to alteration without prior notice.

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Editorial

A HAPPY FIFTEENTH, *SB*

The first issue of this magazine was mailed in February of 1980, which means that the issue you hold in your hands—the first of eight for 1994—is also the first in our fifteenth year of publication. This milestone, while not a major one, merits some pause for reflection on where we came from and some looking forward into the years ahead.

My announcement that speaker articles, which for a decade had appeared at the rate of 2.5 per year in *Audio Amateur*, would be moved to a new periodical was greeted with consternation by a few readers. One California *TAA* subscriber, while loyally sending in a sub to *Speaker Builder*, predicted that neither publication would survive for more than a couple of years.

All through the seventies I had been more and more aware that the Thiele and Small articles were radically reshaping loudspeaker theory. Before *SB*'s founding a major topic in the *Journal* of the Audio Engineering Society had been loudspeaker technology. I also discovered in the late seventies that a Seattle, Washington company named Speakerlab had a mailing list of more than 90,000 names. Founding a loudspeaker journal thus seemed a worthwhile enterprise.

I believed, as I said in my initial editorial, that separating loudspeaker articles into a new magazine would enhance the attention its readers and authors would give to the topic. *Speaker Builder* has done so far beyond any hope I had for it at its founding. The intensification a periodical gains by limiting its subject matter is a result of the feedback between its editor, authors, and readers.

What is vital to this process, however, and its most valuable attribute, is the cross-fertilization of ideas. I have watched authors evolve in their thinking over the past decade and a half. Ideas set out in our earlier issues have undergone major revisions, sometimes by the selfsame author. The journey of this sort of journal is a voyage of discovery.

This magazine has attracted an astonishing array of writing, research, and construction talents. The enthusiasm our contributing editors bring to this avocation is the primary fuel for the contents. It would be difficult to overestimate the value of the education and experience our editors and authors bring to their interest in loudspeaker design and construction.

Although the computer was a technological capability we were aware of in 1980, I doubt that anyone in that year really understood or anticipated what sort of influence what came to be called the PC would have in the decade to follow, including IBM and Apple. Our acceptance of the PC and its proliferation laid the groundwork for an explosion of applications whose influence we are still only very partially realizing.

The PC has revolutionized loudspeakers. The control in manufacturing drivers has enabled a higher level of driver uniformity that was sorely needed; the ease of device design has made much more diversity possible at a far smaller development cost than ever before. Our computer capability as hobbyists in terms of loudspeaker design and measurement has become affordable at the speed of light, it seems. And just as we thought we knew what was going on, the whole enterprise of sound and sight reproduction has begun not only to merge but to migrate into computers as well.

The indirect influence of the computer on almost anything having to do with building loudspeakers is nearly impossible to catalog. Adhesives, plastics, cabinet material, damping materials, connectors, wire, screws, and tools are all evolving rapidly and making possible all manner of small, significant improvements hitherto unattainable.

Speaker Builder circulation grew by just over 20% in 1993. It is now available at more news outlets than ever. Almost 70,000 copies of the magazine were distributed last year, many of them to a steadily expanding audience overseas. I thank all of you for your investment of enthusiasm and vigor in this enterprise. You are a very rewarding audience. I hope you will join me in wishing our teenaged *SB* a happy fifteenth.—E.T.D.

← Reader Service #21

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 exisiting generation; those who dissent from the opinion, still more than those who hold it.

--JOHN STUART MILL

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About This Issue

What happens when a savvy audiophile gets the opportunity to design a sound system for a church auditorium before the cornerstone is laid? Starting on page 12, **Marc Bacon** walks you through the process and the results.

Erno Borbely and **Jean-Claude Gaertner** are well known to Audio Amateur readers for their amplifier and preamp designs—among other things. They have now turned their skills to a modular electronic crossover whose flexibility puts almost any topology within the speaker enthusiast's reach. Indeed, the configurable system allows easy variation and comparison of networks. Their offering begins on page 20.

Is all the trouble we take to carefully manage the backwave from our drivers just a waste of time? Apparently if you pay careful attention to some details, you can leave the back of the cabinet wide open. **Warren Hunt** and **Joseph Janni** have wrestled with the problem and reveal their solution starting on page 30.

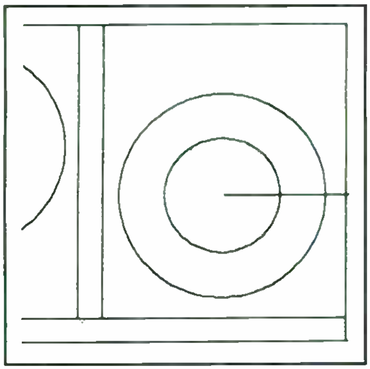
It is painful to decide your speaker is outmoded and that you should probably dispose of it. **Mark D. Seymour** had a long-time relationship with his venerable Magnepan MG-1s. Rather than junk them, he went to work with the basics to see what improvements might be possible. He details these for us beginning on page 44.

Low-frequency AC signals are seldom accurately measurable on any but the most expensive meters. That being so, Professor **Homero Silva** devised a way to convert the signals to DC for greater accuracy. His solution appears on pages 56–57.

Speaker Builder

THE LOUDSPEAKER JOURNAL

VOLUME 16 NUMBER 1 January 1994



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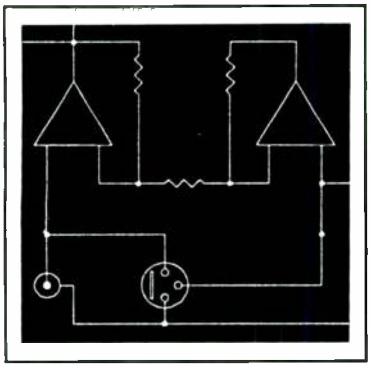

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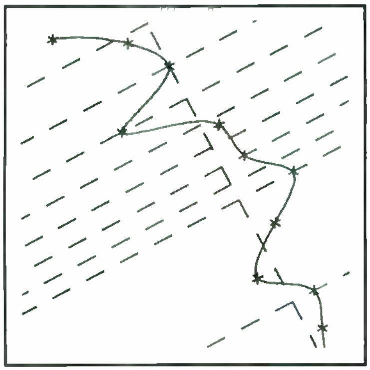

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DEPARTMENTS

SANCTUARY SONICS

By Marc Bacon

When it comes to architectural sound, much of what passes as public address and musical systems bears very little resemblance to "mid-fi," let alone "hi-fi." Accordingly, when I was privileged to participate in the design and construction of a new church in Longueuil, Quebec, I had the opportunity to improve over similar structures, and at a much lower expense.

Before designing any public address system, it is important to have a working knowledge of architectural acoustics. I highly recommend *Acoustical Designing in Architecture*, an excellent primer which requires only a knowledge of high school algebra.[1] [*See Related Products box.—Ed.*]

HOUSES OF THE HOLY

When designing a church building, it is important to prioritize the requirements for speech comprehension versus musical enjoyment. Religions which utilize organ music in large cathedrals will enjoy longer reverberation times, while others place more emphasis on the spoken word. The church I was involved in designing incorporates congregational singing, an occasional choir or instrumental ensemble, and a sermon delivered from the pulpit area. Reverberation must be sufficient to support singing, with proper treatment of sidewalls; the stage area requires a monitor system which will not generate feedback; and excellent speech comprehension is essential.

Further constraints imposed by property lines and strict zoning ordinances dictated a 62½′ × 62½′ maximum building size. In order to accommodate a 300+ seating capacity in the auditorium and still have Sunday school classroom space—plus maintain adequate acoustics—we decided to orient the auditorium on the building diagonal and move the classroom space to the basement and second story. The resulting room has an irregular contour, with no point more than 46′ from the stage, yet with corner-to-corner distances of over 74′. Its only drawback is the lack of parallel side walls to aid congregational singing. Diffusion is excellent, however, and flutter echo nonexistent.

To provide a sense of spaciousness, we set the ceiling height at 14′, with a stage height of 37½″. Construction materials included suspended 2′ × 4′ foam panels for the ceiling, gypsum board on steel studs for the walls, and a concrete floor covered by commercial low-pile rug in the seating areas and asphalt tile in the aisles. In preference to pews, we used individual fabric-covered chairs due to their cost advantage and placement flexibility. An added benefit is that these chairs have a similar acoustical effect whether the auditorium is fully occupied or nearly empty (see the sidebar, "Architectural Acoustic Calculations.")

In designing the musical system, I chose not to follow the time-honored practice of mounting two large, powerful horn-loaded speakers in the front, with directivity calculated to cover the audience. Due to the auditorium's wide fan shape, the angle of coverage is too great to support a convincing stereo image. In addition, the distance from the stage to each listener is

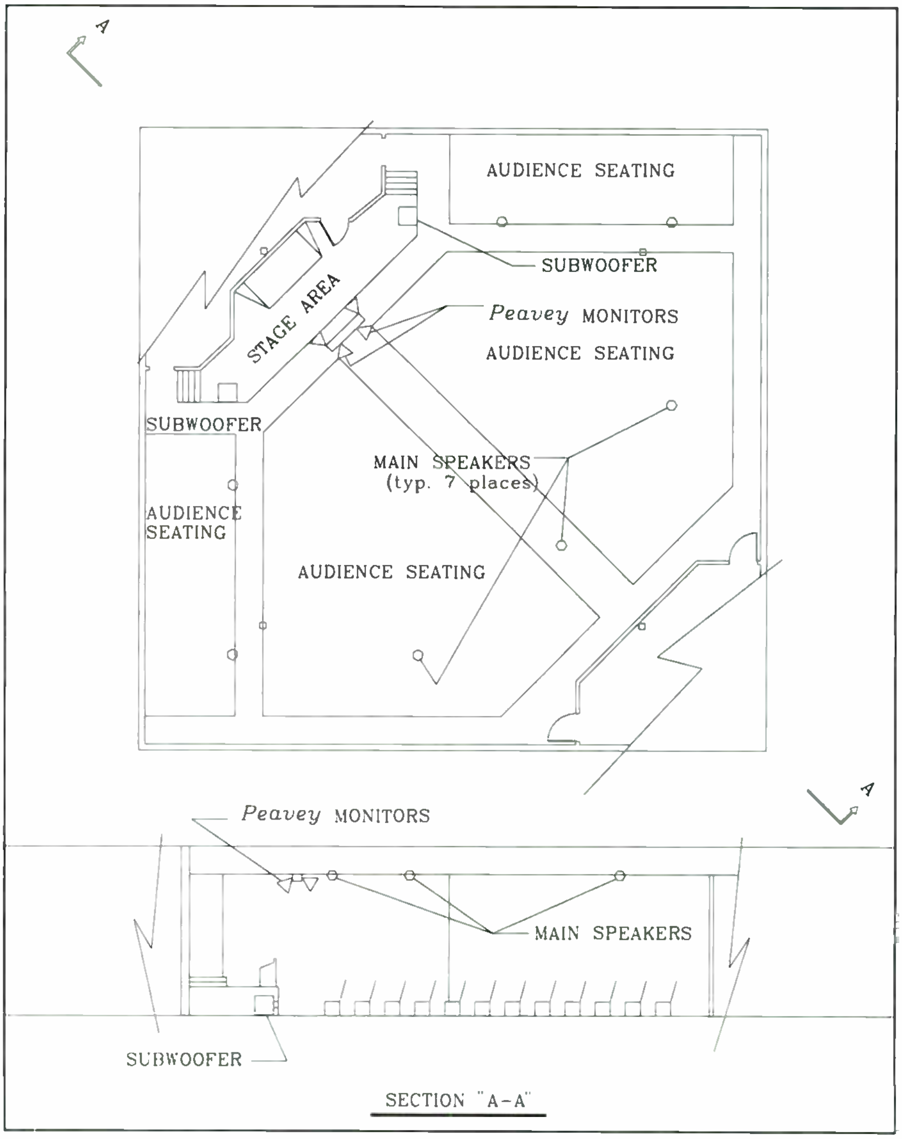

FIGURE 1: The auditorium plan, overhead and side views.

ABOUT THE AUTHOR

Mark Bacon is a professional engineer who works as a consultant in the areas of materials engineering, welding, and quality assurance. Author of several *SB* articles and the Loudspeaker Design Powersheet, he takes an avid interest in speaker design, and especially in system modeling and loudspeaker/room/ear interaction.

insufficient to warrant a high-sensitivity system. Furthermore, I believe discretion to be de rigueur in a church (i.e., the less apparent the better).

MONITORY RESTRAINTS

The final configuration is shown in *Fig. 1*. Two subwoofers are located under the stage area for frequencies below 85Hz. Using a large bandpass box with the port leading to the front of the stage, only two 8″ grille cloths are apparent. The bulk of the musical spectrum is communicated through ceiling-mounted two-way speakers, which are designed to cover a cone of area in a way that minimizes variations in average direct sound levels (*Fig. 2*). Each ceiling speaker is an acoustic suspension unit, with a shallow rear box attached to an MDF front and painted the same color as the ceiling panel acoustical tiles.

To maintain some semblance of sonic perspective, while providing stage monitors, we utilized four existing Peavy monitors (not hi-fi, but small and paid for): two facing the center of the auditorium and two returning sound to the stage area. We incorporated them into a ceiling-mounted bar which also provided stage spot illumination. The monitors' sound level was lowered to give an even SPL over the audience and stage.

Loudspeaker sensitivity was not an issue, since the church already possessed a good mixer and a 200W/channel amplifier. We chose to use mono sound, routing the output of one channel to the subwoofers and the other to the main speakers. Although unconventional, this approach allowed an overall 8Ω/channel load, as well as a single console-mounted passive crossover, with no transformers to degrade sound in the signal path. It also permitted active balancing of the subwoofers' level to the main speakers without using an active crossover.

Based on the listeners' proximity to the stage, the nonparallel walls, and the multiple sound sources resulting from the various speakers, our goal was to create a diffuse soundfield which would be sonically pleasing despite a lack of stereo perspective. As the human ear is notoriously inept at locating sound sources in the vertical plane, we thought the monitor speakers and short stage-to-listener distance might provide some focus.

After considerable computer simulation using the Loudspeaker Design Powersheet, I chose Vieta 13″ drivers for the subwoofers. Discussions with Elliot Zalayet at Zalytron and Kimon Bellas at ORCA revealed that an even more outstanding version of this already superb air mover recently had been developed.

The L120/F3 has improved magnetic system linearity and an incredibly low 16Hz resonant frequency. With its massive, ma-

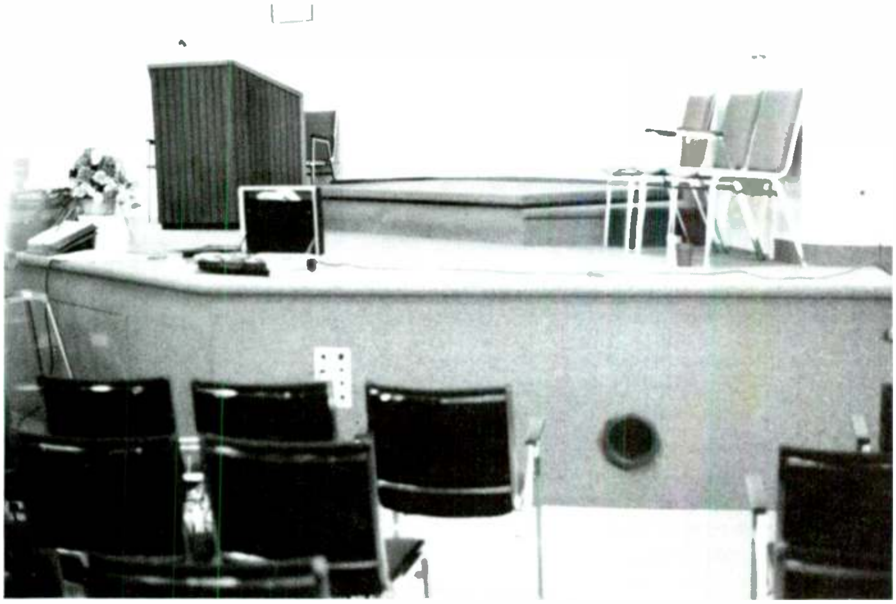

PHOTO 1: View of subwoofer port built into front of stage area.

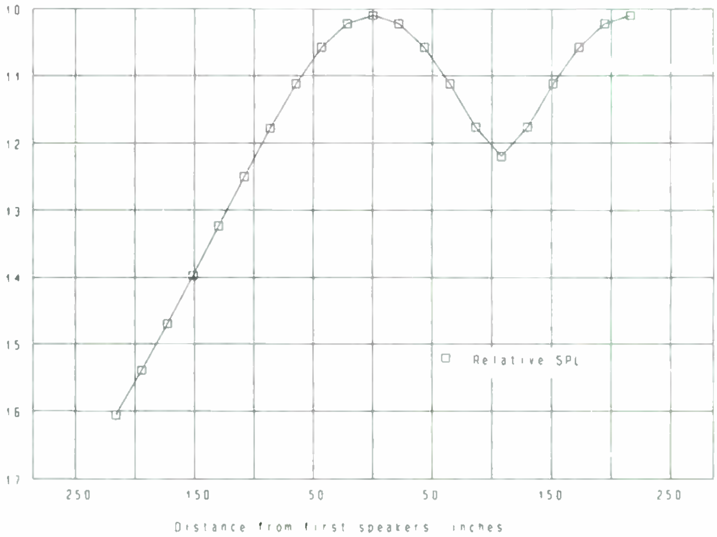

FIGURE 2: Relative SPL between the two ceiling speakers, ignoring room effects.

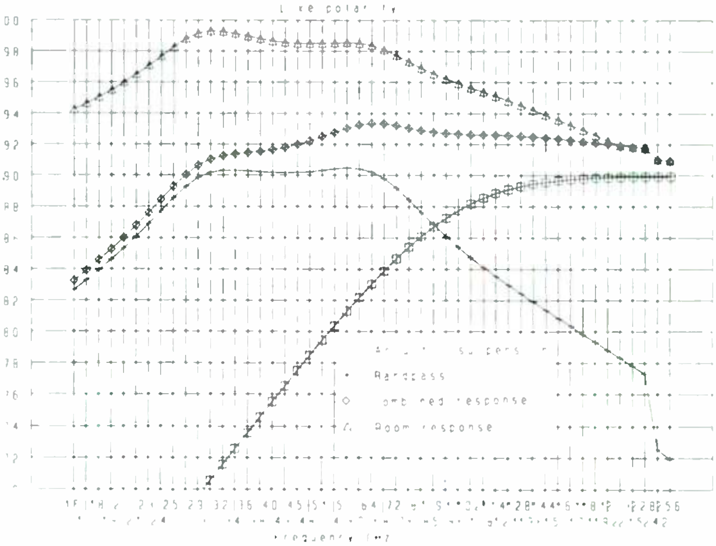

FIGURE 3: LDP predictions of subwoofer/main speaker frequency response.

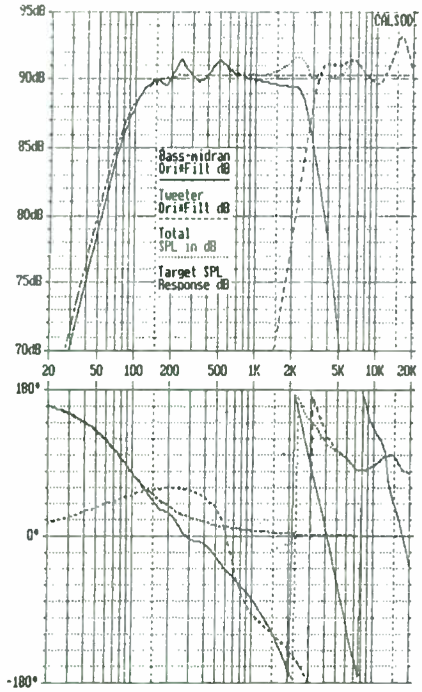

FIGURE 4: CALSOD predictions of main speaker frequency response and input impedance.

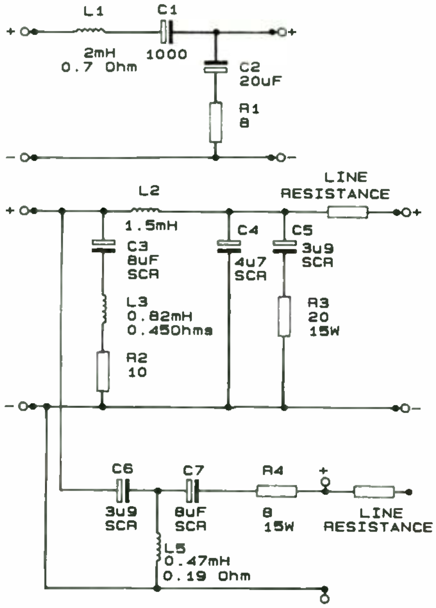

FIGURE 5: Crossover configuration.

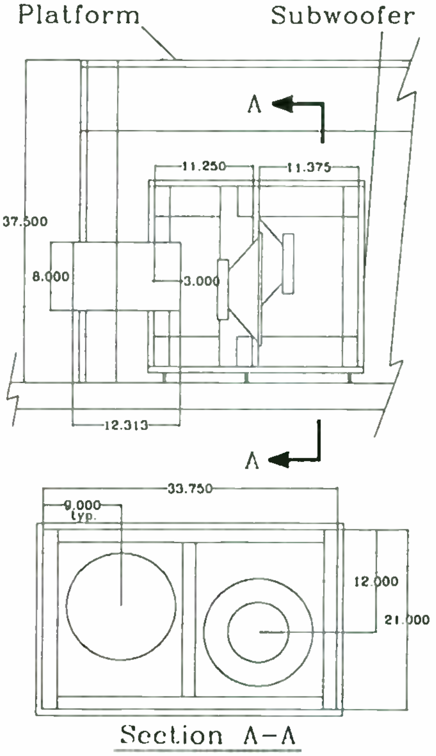

FIGURE 6: Subwoofer box design.

TABLE 1

SUBWOOFER BOX CONSTRUCTION AND MATERIALS

Material:

1 1⁄16″ particleboard on 2″ × 4″ bracing, with all joints glued using construction adhesive.

Construction notes:

1. Support on wood screws and insulate from floor with fiberglass.
2. Mount Vieta woofers face-to-face.
3. Reinforce baffle board between woofers with 2″ × 6″ brace.
4. Measure actual Sonotube diameter and adjust length to suit.
5. Line sides of rear box with 1″ fiberglass.

Specifications:

Impedance 4Ω
400W RMS
SPL 93.1dB 2.83V/1M
F_L = 22Hz
F_O = 43Hz
F_H = 85Hz
Q_B = 0.9
S = 0.55

chined die-cast aluminum frame, ample rear opening, thick (over ⅛″) highly damped cone, long travel, and uniquely vented voice coil, it is an amazing value. Given a box which allows them to work, these drivers will deliver effortless, ample, low-distortion bass notes. On sustained notes with a large amplifier, you can feel the wind from the 8-inch-diameter ports from 4′ away. (Note: Since the time this article was submitted, Vieta drivers are no longer imported by ORCA. Focal has developed a drop-in replacement which I have not yet tried, but which they claim has even lower distortion.)

For woofer/midranges, I chose the Focal 6V013. These units have an uncanny midrange that is immediately coherent and three-dimensional, even when unbaffled. When placed in a 20-liter ported box, they have surprisingly clean bass output, although they are incapable of reproducing the lowest octave. The price/performance ratio is outstanding, the rolloff easily controlled by a simple crossover, and the off-axis response would make any serious listener happy. Furthermore, they are less affected by box reflections than some thinner-coned Kevlar midranges. As with all the Focal drivers I've seen, the craftsmanship is superb.

The tweeter sets the timbre for most musical instruments and reproduces the frequencies necessary for speech comprehension, so it is critical to the tonal balance. I chose the Focal T90Ti02, a very musical tweeter with a concave shape resulting in outstanding off-axis performance. The resonance typical of many metal-domed drivers has been pushed up into the range of inaudibility; the titanium dome is further damped with a 7-micron layer of titanium oxide. According to ORCA, extensive Finite Element Analysis (FEA) of magnetic circuits, suspension, voice coil formers, and the like, has been used in the driver's design. Although they compress more than the Morel MDT30 when overdriven by low-frequency sound, they are much more analytical to my ears, with a certain "sweetness" that is hard to define, yet extremely pleasant.

UNIDENTIFIED SOURCE

Using the Loudspeaker Design Powersheet and CALSOD, I designed the subwoofer boxes and ceiling units, as well as an extension speaker.[2,3] The results are shown in *Figs. 3–8*.

I chose a fourth-order Linkwitz-Riley (L-R) acoustical crossover between the midranges and the tweeters because of its good lobing pattern, low Q, and good tweeter protection. In developing the circuit, I used the resistances of the wire (14-gauge Signet music wire from Zalytron) as a lumped parameter before the crossover, and an impedance-compensating circuit to reduce the inevitable peak around the crossover. To keep series resistance low, I used two conductors in each direction for the subwoofers, and one conductor for each midrange and tweeter.

A single console-mounted passive crossover reduced the cost substantially compared with individual crossovers mounted in each box; however, it required bi-wiring the entire church directly from the crossover, which was an 11-hour job for two people. Although the crossover is a fourth-order L-R, the electrical part of it is not, so it must be combined with the driver response to obtain the desired transfer function. Crossover parts included SCR polypropylene capacitors, Intertechnik coils, Signet wire, and massive ORCA binding posts. While going to a lower-order crossover and cheaper parts may be tempting, I firmly believe this to be false economy for audiophile systems.

The subwoofer boxes, which seemed mon-

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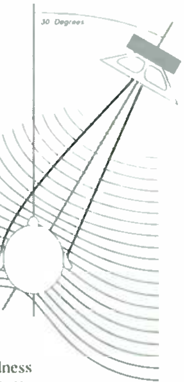


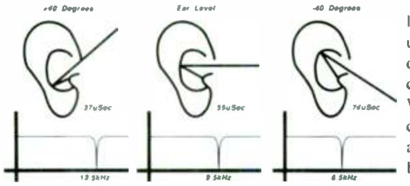
The acoustic image produced by a stereo system is a sound painting. It exists in your mind because you can localize the sound in space. Headphone listening is unable to present a normal audio image because much of the spacial localization information is missing. In order to understand what HeadRoom does and how it corrects localization, you must first understand the psychoacoustic cues your mind uses to localize sound.

Imagine you are listening to a pair of speakers, and you turn off the left speaker. Both ears continue to hear the right speaker. But the right ear is a little closer to the speaker, so it hears the sound slightly before the left ear. This time difference between ears is called the Inter-Aural Time Difference (ITD). ITD is the primary cue your mind uses to localize sound left-to-right (lateralization).

The near ear also hears the sound slightly louder. This is because the far ear is in the acoustic shadow of the head. The loudness difference between ears is called the Inter-Aural Amplitude Difference (IAD).

ITD significantly overwhelms the importance of IAD as a lateralizing cue.

The Pinna is the outside part of your ear. The part you can see. The folds and ridges of the Pinna create complex interactions with the high frequency components of incoming sound. The exact nature of the relationship between Pinna effects and auditory localization is not completely understood. However, it is generally agreed that the Pinnae create high frequency reflections around your ear, heard and decoded by your mind as up-down localization clues.



There are a variety of other localization cues your mind uses. Your visual perception of physical surroundings is compared to what you hear. Walls, ceiling, floor, your clothes, your posture, etc. are all used to evaluate sound timbre and location.

But the most important cues of all are the acoustic changes heard with head movement:

Your acoustic memory is very short, so to evaluate sound accurately you need to switch quickly between a reference sound and an unknown sound. Your mind accurately senses acoustic differences only in that brief moment of transition. This is why it is better to A-B switch between audio equipment when performing quantitative objective comparisons without test equipment. (Please do not missread this as an endorsment of ABX audio reviewing. I firmly believe in the validity and importance of the subjective musical experience). To overcome short term audio memory problems during sound localization, your mind continuously compares head movements with changes in the IAD, ITD, and Pinna cues. The changing head position is the reference against which all localization cues are perceived and understood.

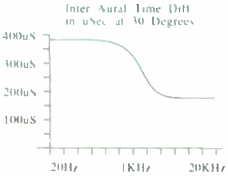
Hardware to duplicate all the above cues in headphones sufficiently to recreate normal listening would be extraordinarily complicated. It would have to be adjustable for each individual, it would have to take into account your head movements while listening, and it would have to be calibrated for each individuals ear shape.

Such solutions exist however. NASA has a system that duplicates virtually all psychoacoustic cues for pilots and astronauts: termed a convolvatron. The system synthesizes the out-of-head audio experience sufficiently for pilots to hear sound from their coordinate directions. For example, if an enemy plane is behind him, the threat warning signal sounds like it comes from that direction. If the pilot turns his head, a sensor in the helmet measures the head movement, and the warning signal continues to come from the correct direction. This gives pilots a more natural cognitive environment in which to work. This experimental system requires that the pilot go through an ear calibration session in an anechoic chamber.

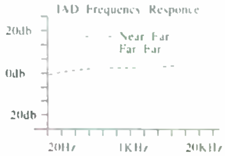
The HeadRoom solution is not quite as complex, it strives to recreate only ITD and IAD information. HeadRoom, however, is much less expensive.

When first considering this project, there was a tendency to try to model other localization cues (like reverberance). We have resisted this temptation. We designed HeadRoom to process the audio signal only the amount required to recreate the natural delays and equalization you hear with a pair of speakers.

So . . . let's go back to turning off one speaker. In headphones, if you turn off the left channel, only the right ear hears the remaining audio. Consider that this means that any audio information that is only on one channel, is only heard in one ear. Your mind doesn't like this, your brain wants to hear the audio in both ears with a time delay between ears. This is exactly what HeadRoom does. It provides a cross-feed delay and EQ signal path between channels. In other words, the right channel goes directly to the right ear, but it also gets mixed into the left channel via the electronic delay and equalization circuit. We call this circuit an "Audio Image Processor."



The interaural time difference for 30 degree off center sound is about 300 uSec. But different frequencies travel around to the far ear at different rates. Low frequencies set up a standing wave at the surface of the head and take longer to get around to the far ear than the high frequencies. You can visualize this like a rock in a river: water rushing up to the rock feels the back pressure of the water next to the rock. To get around the rock, the water has to travel farther than the actual diameter of the rock. The delay generated by the Audio Image Processor, therefore, is not a fixed time delay. It delays low frequencies for longer than the high frequencies.



It is intuitively obvious that the ear farthest away from the speaker does not hear sound as loud as the ear closest to the speaker. What is not quite so obvious is the frequency dependence of this phenomena. Low frequencies wrap themselves around your head as they go by, so they experience only modest attenuation as they travel to the far ear. As you get higher in frequency (~1KHz), the acoustic wave has a more difficult time wrapping around your head. Soon the sound just travels right past your head, and the far ear is literally in the shadow of the head. As frequencies get very high (~5KHZ), acoustic energy begins to act like packets and wants to travel along surfaces (Literally: The Skin Effect). When these high frequency sounds reach your head, they follow your skin surface around to the far ear and restore frequency response. The result is the IAD response curve has a dip centered around 2 KHz.

The HeadRoom crossfeed circuit does not exactly match the typical IAD curve. It would be difficult to build an analog filter with a dip at 2 KHz and retain the correct ITD response of the filter. But nature has been kind to us.

When you delay an audio signal and mix it back with itself, you create a notch filter. The center frequency of the notch will be that frequency which has a half cycle time equal to the delay. (You may have to think about that for a sec). A significant amount of audio is the same on both the left and right channel, so we create a notch filter in the mono part of the audio. But the resulting frequency response is very close to the response required by the natural IAD.

Either we were lucky, or nature is beautiful. Well, actually, we knew what we were doing, but the numbers did have to be close enough for the right tweak.

An additional contributor to IAD is the slightly better high frequency response of the ear closest to the speaker. As a sound source moves toward the side, the high frequency response of the ear on that side improves. The Premium and Supreme HeadRoom include a Near-Ear Emphasis Filter.

Without HeadRoom

With HeadRoom

Headphones deliver audio that seems to come from inside the head. HeadRoom does not solve "In-Head-Localization." To do this you must recreate all localization cues. But HeadRoom does correctly "Lateralize" the audio image. Typically, when listening to headphones, there is a strong central image with two sound blobs to the left and right. This happens because the difference signal audio tends to snap all the way over to the ears. And the mono signal is heard to form a strong center image. These lateralizing problems are almost totally corrected by HeadRoom because the difference signal information arrives at both ears with correct timing and amplitude differences.

It is worthy to note that the very best recordings (Chesky, Dorian, Delos, Audio Quest, M.A. Mapleshade, Three Blind Mice, etc.) are made with very few and very expensive microphones, and do contain some timing information. But, these recordings remain engineered for speaker reproduction, not headphones. HeadRoom does a surprisingly good job imaging these recordings on headphones.

The image when listening to HeadRoom fills the left-right space smoothly. Instruments tend to be more clearly localized at one point in space. Reverberance seems to be more natural and behind the instruments. A depth dimension becomes apparent. Your head seems to become a coherent listening space, a more natural audio environment. The sound has dimension within your head.

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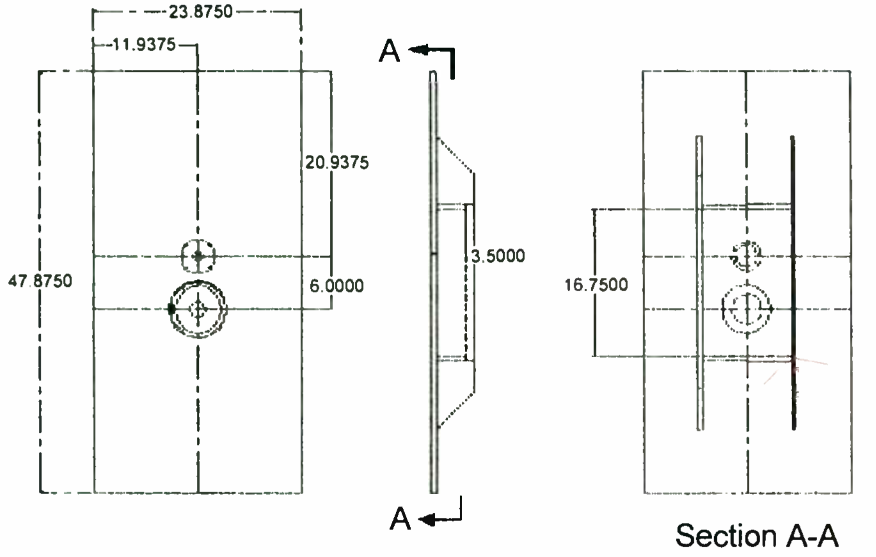

FIGURE 7: Ceiling box design.

TABLE 2

CEILING BOX CONSTRUCTION AND MATERIALS

Material:
½″ particle board, except for back, which is $1\frac{1}{16}$particle board.

Construction notes:

1. All joints are glued and screwed.
2. Use grille cloth or window screen on quarter round frame for grille.
3. Paint to match acoustic ceiling.
4. Fill box with fiberglass insulation. Specifications:

Impedance 8Ω
75W music power
F3 85 Hz
SPL 90 dB 2.83V/1m
$Q_{TC} = 0.7$

Continued from page 14

strous on my workbench, are totally invisible under the stage area. To prevent unpleasant bass sounds from transmitting to the basement, every joint is glued and reinforced with 2″ × 4″ building lumber. In addition, the boxes are isolated from the floor on large screws set on a mat of insulation. The ports are 8-inch-diameter Sonotubes set 3″ into the subwoofers (to ensure that my formula for end correction applied), but project from the front of the boxes to the front of the stage. Thus, the box is entirely isolated from the surrounding plane surfaces.

The result is quite good, with better definition, transients, and extended response than many audiophile systems, although the imaging is not as convincing as my home system. The cost is just over $5 per installed seat, not counting my time, which is a bargain compared to professional installations. Response

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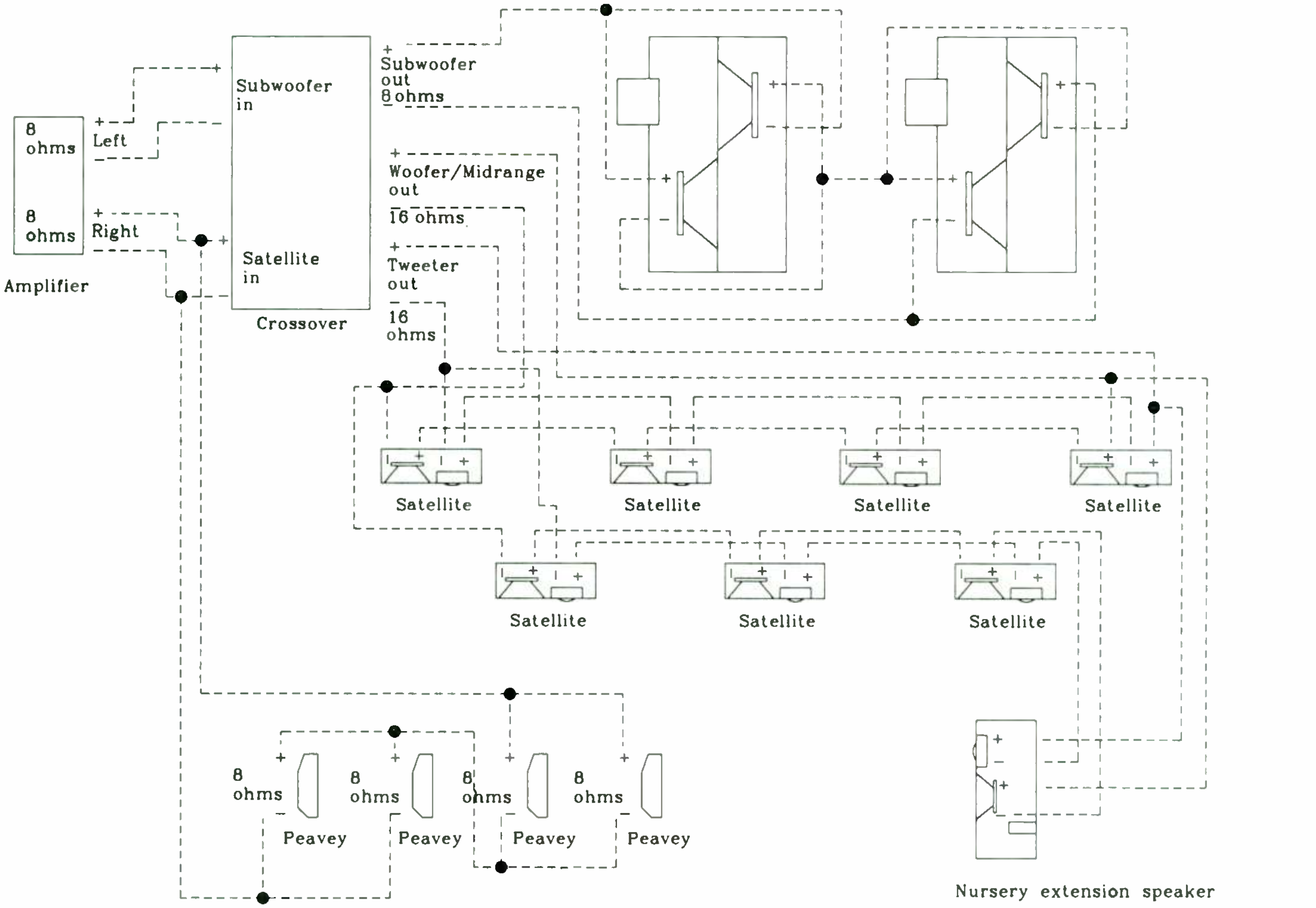

FIGURE 8: Circuit wiring diagram.

RELATED PRODUCTS

The following are available from Old Colony Sound Lab, PO Box 243, Peterborough NH 03458, (603) 924-6371, FAX (603) 924-9467.

#BKAC4	*Acoustical Designing in Architecture*	$27.95
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#SOF-PSH1B3	(same) 1 × 3½″	$49.95
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#SOF-CAL2B4G	CALSOD WITH GRAPHICS Standard version 1.30 for IBM 1 × 3½″	$69.95
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Continued from page 16

under the middle rear speaker at a seated ear level is ±2dB from 100Hz–15kHz referenced to 1kHz. Apart from a narrow (1′ wide by 8′ long) band behind each column, pink noise is within 1.5dB of center throughout the entire listening area. Intelligibility at 70–80dB is certainly preferable to earsplitting PA speakers, visibly mounted up front for all who sit near them to hate.

The sound balance makes it difficult to locate the source as either the front of the stage

ARCHITECTURAL ACOUSTIC CALCULATIONS

1. Calculate reverberation time (p. 138, Equation 8.17)*:

$$t_{60} = \frac{0.049V}{-2.30S\ \log_{10}(1-\zeta) + 4mv}$$

where:

t_{60} = Time in seconds for a 60dB decay in sound
V = Room volume, ft.^{3}
S = Total surface area of the room
ζ = Average absorption coefficient of the room
m = Attenuation coefficient of the air

For the church, results are (*Fig. A*):

Frequency (Hz)	Empty Room	Full Room	Optimal for Speech
128	0.718	0.751	1.100
256	0.558	0.523	0.980
512	0.404	0.363	0.820
1,024	0.371	0.325	0.820
2,048	0.368	0.320	0.820
4,096	0.343	0.300	0.820

The general characteristic is a very "dead" auditorium, but having excellent speech comprehension (syllable articulation is over 87%). A distributed low-level system is required to support music. The trade-off for such an environment is lack of fullness for congregational singing, somewhat compensated for by the hard, reflective walls.

2. Estimate diffusion characteristics: Consider a point source of sound on the stage and reflect rays off the walls. Then, using a CAD drafting program, observe if sound foci exist. Valid for frequencies over approximately 250Hz. The church showed excellent diffusion. A low-level distributed sound system also aids in diffusion.

3. Volume per seat (p. 170): Acceptable figures range from a low of 125 ft.^{3}/seat to as much as 175 ft.^{3}/seat. The church has 132 ft.^{3}/seat. Sound is normally better and construction is less expensive toward the lower end of the scale.

4. Calculate transmission loss and noise reduction factors (pp. 216, 217):

$$\text{Noise insulation factor} = 10\log_{10} a/T\ \text{dB}$$

where:

a = Total number of units of absorption in the room
T = $\Sigma T_1S_1 + T_2S_2 + T_3S_3$
$T_{1,2,3}$ = Transmission coefficients of room boundaries with areas S_1, S_2, and S_3

For the church, the noise insulation factor is 38dB.

The average A-weighted noise varies between 58 and 65dB at this location. Sound levels in the unoccupied church are thus between 20 and 37dB, which is *extremely* quiet. The concrete floor and ceiling, brick facing, double doors, heavily insulated walls, and thick thermopane windows provide excellent insulation.

5. Calculate steady-state SPL for one acoustic watt (p. 125, Equation 8.6):

$$Lp = 10\log_{10} 1/a + 136\text{dB}$$

where:

Lp= SPL for one acoustic watt
a = Total absorption in Sabins

For the church, Lp = 99.6dB.

With an average loudspeaker efficiency of 0.08, the sound level for one electrical watt is 78dB. Thus, maximum sustained sound level with the 200W/channel amplifier is 101dB throughout the building—too much for comfort. Maximum before exceeding speaker thermal rating at 7 × 75W = 105dB.

6. Calculate angle from sound source to listener (*Fig. B*):

Angle for front listener = 228.8°
Angle for rear listener = 7.7°
Difference = 21.1°

This is more than the 15° desirable as a good figure, so grazing attenuation is greatly reduced.

*References are to page numbers in *Acoustical Designing in Architecture.*

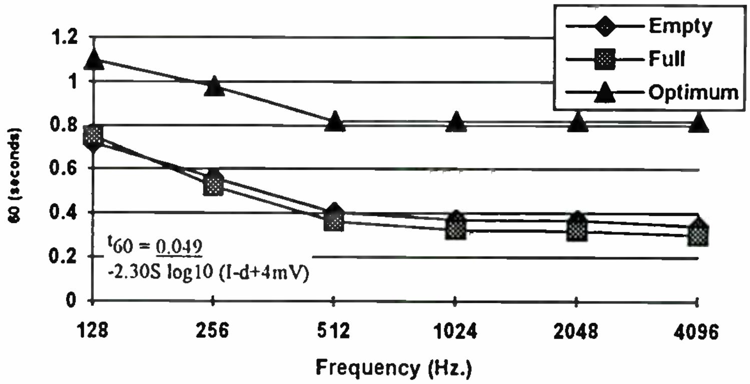

FIGURE A: Reverberation time of Longueuil church.

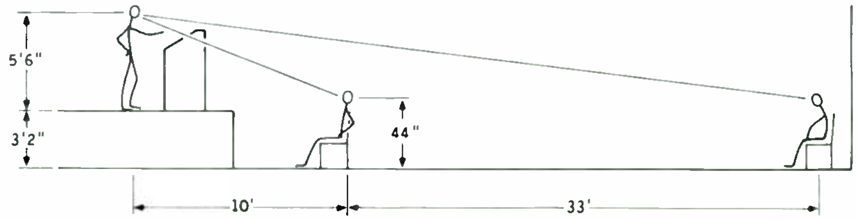

FIGURE B: Cross-section of auditorium.

(subwoofers) or the ceiling (main speakers). Flush-mounted ceiling speakers offer the added advantage of not suffering from lower-midrange dip, as they radiate into half-spherical space.

HOW TO SUCCEED

While none of the congregation had heard anything like the Vietas' bass output, no one complained of boominess or overbearing bass, either. Playing test tones during installation was fun: at 20Hz, the 3″ concrete floor resonated; at 30Hz, the windows vibrated; at 40Hz, the gyproc set up a roar; and at 50Hz, the ventilation ducts buzzed—all with about 93dB steady sine waves. Most important, that lower octave provided the foundation so vital to sacred music. The buzzing does not, of course, occur with most music. The Vietas were far from being stressed, even with all the amplifier power I had on hand to play the Telarc 1812 Overture cannon.

Good sound in public places *is* attainable. All it takes is application of basic acoustics, some of the speaker-building knowledge you have acquired at home, and a few simple modeling and measurement tools. You should avoid undersized wires, stamped perforated baffles, whizzer cones (rather than quality tweeters and crossovers), tiny impedance-matching transformers, and no rear chamber. Unlike many sound reinforcement speaker manufacturers, you should also avoid using a huge driver with small excursion limits in a small box, which is crossed over much too high to a compression horn whose only virtue is high sensitivity.

You will succeed, because speakers are your hobby and not your livelihood, so you can afford to spend the time to plan and execute your work properly. I encourage all of you to take advantage of any opportunity to participate in a similar undertaking.

REFERENCES

1. Knudsen, Vern O. and Cyril M. Harris, *Acoustical Designing in Architecture*, Acoustical Society of America, 1978.

2. Bacon, Marc, "The Danielle, Part I," *SB* 4/92, p. 22.

3. Bacon, Marc, "The Danielle, Part II," *SB* 5/92, p. 34.

SOURCES

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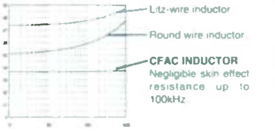

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Reader Service #40

World Radio History

MODULAR ACTIVE CROSSOVERS

By Erno Borbely and Jean-Claude Gaertner

The laws of physics are such that reproducing the entire audio frequency range with one transducer is practically impossible. We must, therefore, rely on filters to divide the audio spectrum into smaller ranges for loudspeakers. For cost reasons, most of these filters are passive. They allow us to get by with only one amplifier, and the whole setup seems very simple.

Calculating a passive filter is a difficult task for an amateur. Effectively, the loudspeaker's impedance varies considerably as a function of frequency, and this affects the results of the calculations. Utilization of networks that correct the impedance (and, more importantly, software like CALSOD and LEAP) facilitate these calculations.

Active filters, which require a separate amplifier for each band, are generally reserved for high-end speakers. Obviously, this costs more than a single amplifier, but the approach has several advantages: you can use loudspeakers with different characteristics; the amplifiers can be adapted to the particular frequency range they are supposed to reproduce; each speaker is driven directly from the amplifier's output with only a cable in between; filter element calculation is very simple and can be accomplished with textbook formulas.

A number of active filter structures are available, each having its own advantages and disadvantages. We chose a second-order structure called the Sallen-Key. The independence between the filter's low-pass (LP) and high-pass (HP) sections allows you to adapt the slope and cutoff frequency to suit your particular speaker. We will propose several filter applications for practical speaker systems.

All filter types theoretically can be implemented with the circuit proposed here, but only Bessels, Butterworths, and Linkwitz-Rileys are of interest for audio. The Bessel (also called Thomson) offers the best transient response. The overall slope around the cutoff frequency, however, is only about 4.5dB/octave, with the 6dB/octave slope per order achieved only several octaves beyond the cutoff frequency. The Butterworth offers

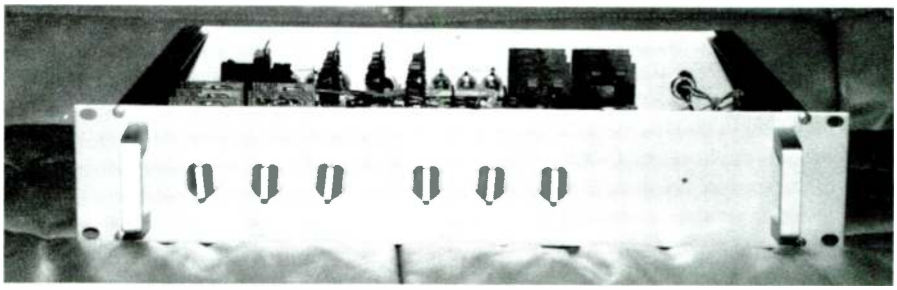


PHOTO 1: Front view of the crossover. We used an aluminum 2U 19' Fisher box with an internal chassis.

PHOTO 2: Rear view of the crossover. XLR is used for symmetrical, and Teflon® isolated cinch for asymmetrical input. Note that on this prototype we didn't mount the XLR plug that would be needed for symmetrical output.

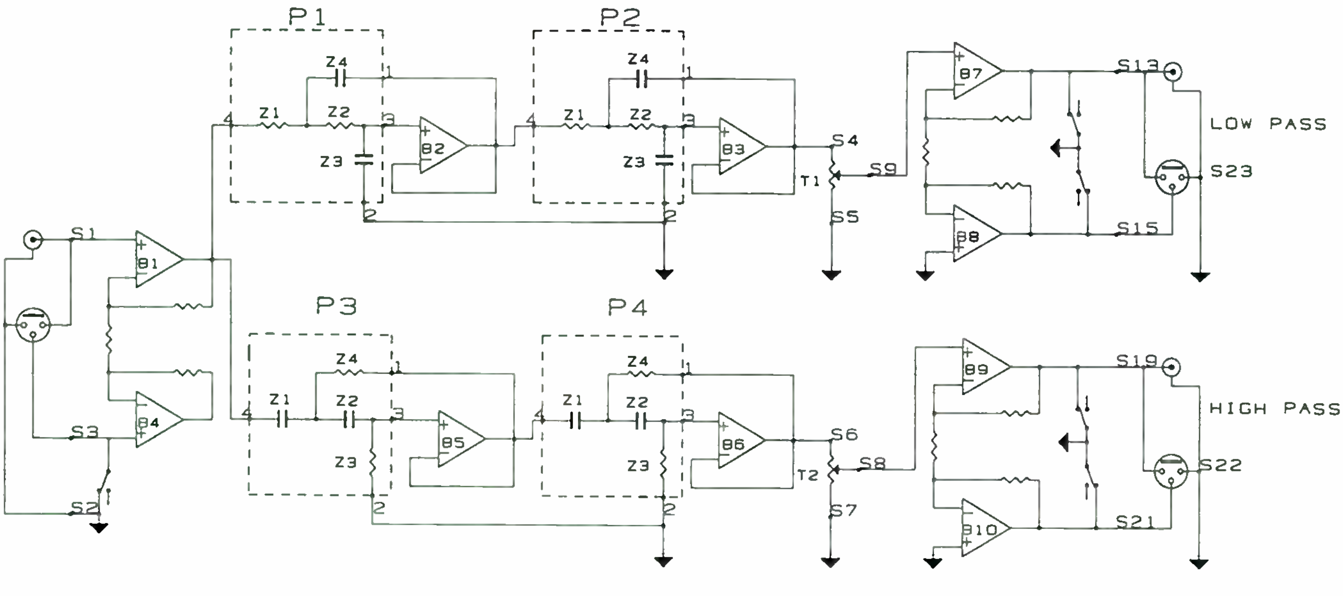


FIGURE 1: Two-way crossover block schematic. Power supply is ±24V. Switch on mute.

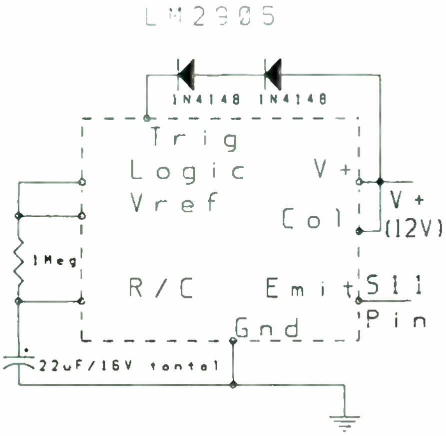


FIGURE 2: Switch-on delay schematic. The switch-on delay is R/C (about 22 seconds in this case).

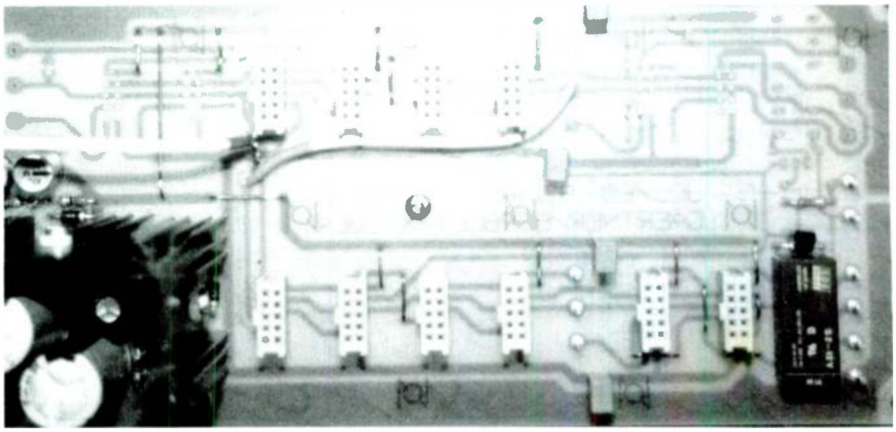


PHOTO 3: The three-way extension motherboard. Note the two shielded links: the first is to bring the buffered signal from the main two-way motherboard; the other is to link the high- and low-pass sections on the three-way extension motherboard.

ness, bandstop slope, and transient response. The Linkwitz-Riley, an even-order filter, combines all the qualities of the Butterworth with good phase response around the cutoff frequency.

CIRCUIT DESCRIPTION

The JC/EB-193/141 is a universal active filter designed for two- and three-way loudspeakers. It is implemented on a motherboard, with plug-in or solderable buffers and crossover networks. (Each motherboard serves one channel.) In the basic two-way mode, it provides a common input buffer, the LP and HP filter sections, and the output buffers. As a three-way extension board, it uses the filter sections and the output buffers. The filter slope is selectable in 6dB increments between 6 and 24dB. Input and output buffers can be configured for either single-ended or balanced operation.

Figure 1 is the block schematic for a two-way crossover. B1/B4 form a balanced-to-single converter and serve as a common input buffer for the filter sections. When used with single-ended sources, B4's input should be shorted to ground (S3 to S2). Alternatively, it can be removed from the board.

The crossover's LP section is formed with B2 and B3, with their networks P1 and P2. B5 and B6, with P3 and P4, form the HP section. As shown, each buffer and its associated network represents a basic second-order Sallen-Key filter; therefore, a second-order crossover

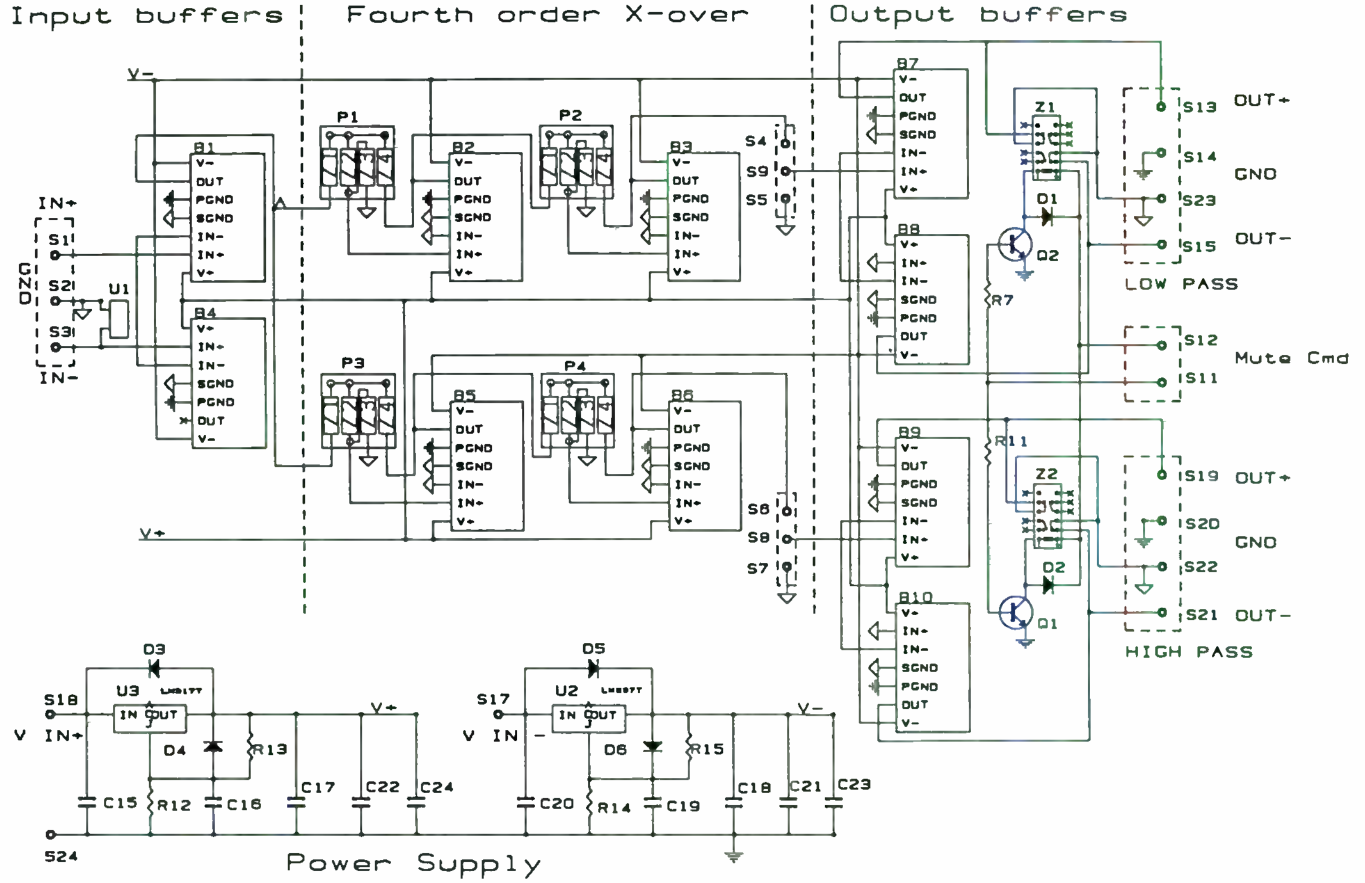


FIGURE 3: Two-way crossover schematic and wiring diagram.

can be implemented with only one buffer/network. A third-order filter requires one second- and one first-order network. This can be implemented with either P1/B2 or P3/B5, and Z2/Z3 of P2 or P4. B3 (B6) is needed only if you use the optional volume control P1 (P2). For a fourth-order crossover, you will need all four buffers and networks.

The single-to-balanced converter for the output is comprised of B7/B8 and B9/B10. They also serve as output buffers. For single-ended output, B8 and B10 can be omitted. A mute relay installed at the output will, when activated, short the output signal to ground. We recommend doing so in order to avoid DC thumps at the filter's output when powering up and down. *Figure 2* shows a possible switch-on delay.

Figure 3 is the two-way crossover's schematic and wiring diagram. The buffers are shown as blocks with pin assignments. Note that signal ground and power ground are completely separated, allowing you the flexibility of implementing your own grounding scheme. (You must, however, remember to connect the two grounds somewhere—either on the board or at the power supply.) The motherboard has built-in 317/337-type regulators, which you can omit when using off-the-board regulators. Q1 and Q2 are the relay drivers for the mute circuit. The mute relay, an SDS S2 12V, shorts the output of the buffers to ground. An instantaneous or delayed mute command is supplied to the board through Pins S11 and S12: S11 is the mute command pin (the relay is activated with a voltage higher than 2V);

CALCULATING THE CROSSOVER NETWORK

We wish to build a two-way crossover with a cutoff frequency of 2kHz, and will calculate all the values of the passive network.

$$F_C = \frac{1}{2 \times \pi \times R_O \times C_O} \qquad (1)$$

According to equation (1) we have:

$$2{,}000 = \frac{1}{2 \times 3.1416 \times R_O \times C_O} \rightarrow R_O \times C_O = \frac{1}{6.2832 \times 2{,}000} = 7.95773 \times 10^{-5} \qquad (2)$$

LOW-PASS SECTION

First-order:
We choose a value for C_0 = 5,600pF and according to (2) R_0 = 14,210Ω. So, we will use P1 (*Fig. 1*) and $Z_2 = R_0$ = 14,210Ω (13k + 1k21), $Z_3 = C_0$ = 5,600pF.

Second-order Butterworth:
Table A gives m_1 = 0.7071 q_1 = 1.4142. *Figure A2* gives $C_1 = m_1C_0$, $C_2 = q_1C_0$, so $C_2 = 2C_1$. We choose C_1 = 10nF, so C_2 = 20nF. $C_1 = m_1C_0 = 0.7071C_0$, so C_0 = 14.142nF and equation (2) gives R_0 = 5,627Ω (5k62). We will use P1 and $Z_1 = Z_2 = R_0$ = 5,620Ω, $Z_3 = C_1$ = 10nF, Z4 = C2 = 20nF (10nF+10nF).

Third-order Butterworth:
Table A gives $m_1 = 0.5$, $m_2 = 1$, $q_1 = 2$. *Figure A3* gives $C_1 = m_1C_0$, $C_3 = m_2C_0$, $C_2 = q_1C_0$. We can see that $C_2 = 2C_3 = 4C_1$. We choose C_2 = 20nF, C_3 = 10nF, C_1 = 5,000pF. $C_1 = m_1C_0$, so C_0 = 2,500pF and equation (2) gives R_0 = 7,957Ω. P1 and P2 will be needed. P1: $Z_1 = Z_2 = R_0$ = 7,960Ω (6k81+1k15), $Z_3 = C_1$ = 5,030pF (4,700pF+330pF), $Z_4 = C_2$ = 20nF (10nF+10nF). P2: $Z_2 = R_0$ = 7,960Ω (6k81+1k15), $Z_3 = C_3$ = 10nF.

Fourth-order Butterworth:
Table A gives $m_1 = 0.9238$, $m_2 = 0.3826$, $q_1 = 1.0823$, $q_2 = 2.6131$. *Figure A4* gives $C_1 = m_1C_0$, $C_3 = m_2C_0$, $C_2 = q_1C_0$, $C_4 = q_2C_0$. We choose C_4 = 22nF, so C_0 = 8.419nF and equation (2) gives R_0 = 9,452Ω, C_1 = 7.777nF, C_3 = 3.22nF, C_2 = 9.11nF. P1 and P2 will be needed. P1: $Z_1 = Z_2 = R_0$ = 9,452Ω (8k25+1k21), $Z_3 = C_1$ = 7.77nF(6.8nF+1nF), $Z_4 = C_2$ = 9.11nF (8.2nF+1nF). P2: $Z_1 = Z_2 = R_0$ = 9,452Ω (8k25+1k21), $Z_3 = C_3$ = 3.22nF (2.2nF+1nF), $Z_4 = C_4$ = 22nF.

Fourth-order Linkwitz-Riley:
Two second-order Butterworths in cascade. We have already calculated the values, so passive network P1 and P2 will be the same, as you can see in *Fig. A5*. P1: $Z_1 = Z_2 = R_0$ = 5,620Ω, $Z_3 = C_1$ = 10nF, $Z_4 = C_2$ = 20nF(10nF+10nF). P2: $Z_1 = Z_2 = R_0$ = 5,620Ω, $Z_3 = C_1$ = 10nF, $Z_4 = C_2$ = 20nF(10nF+10nF).

HIGH-PASS SECTION

First-order:
We choose a value for C_0 = 5,600pF and according to (2) R_0 = 14,210Ω. So we will use P3 (*Fig. 1*) and $Z_2 = C_0$ = 5,600pF $Z_3 = R_0$ = 14,210Ω (13k+1k21).

Second-order Butterworth:
Table A gives m_1 = 0.7071 q_1 = 1.4142. *Figure B2* gives $R_1 = R_0/m_1$, $R_2 = R_0/q_1$. We choose C_0 = 5,600pF, according to equation (2) R_0 = 14,210Ω, R_1 = 2,0096Ω, R_2 = 10,048Ω. We will use P3 (*Fig. 1*) and $Z_1 = Z_2 = C_0$ = 5,600pF, $Z_3 = R_1$ = 20k, $Z_4 = R_2$ = 10k.

Third-order Butterworth:
Table A gives $m_1 = 0.5$, $m_2 = 1$, $q_1 = 2$. *Fig. B3* gives $R_1 = R_0/m_1$, $R_3 = R_0/m_2$, $R_2 = R_0/q_1$. We choose C_0 = 6,800pF according to equation (2) R_0 = 11,702Ω, R_1 = 23,405Ω, $R_3 = R_0$ = 11,702Ω, R_2 = 5,851Ω. P3 and P4 will be needed (*Fig. 1*). P3: $Z_1 = Z_2 = C_0$ = 6,800pF, $Z_3 = R_1$ = 23,405Ω (22k1+1k3), $Z_4 = R_2$ = 5,851Ω (5k62+221). P4: $Z_2 = C_0$ = 5,600pF, $Z_3 = R_3$ = 11,702Ω (11k+681).

Fourth-order Butterworth:
Table 1 gives $m_1 = 0.9238$, $m_2 = 0.3826$, q_1

TABLE A

CROSSOVER NETWORK EQUATIONS

BUTTERWORTH (see note 1)

Order	m	q	equation
first (6dB/octave)	0	0	
second (12dB/octave)	$m_1 = 0.7071$	$q_1 = 1.4142$	$(p^2 + 1.4142p + 1)$
third (18dB/octave)	$m_1 = 0.5$	$q_1 = 2$	$(p^2 + p + 1)(p + 1)$
	$m_2 = 0.1$		
fourth (24dB/octave)	$m_1 = 0.9238$	$q_1 = 1.0823$	
	$m_1 = 0.3826$	$q_2 = 2.6131$	$(p^2 + 1.8477p + 1)$
			$(p^2 + 0.7653p + 1)$

LINKWITZ-RILEY (see note 1)

Order	m	q	equation
fourth (24dB/octave)	$m_1 = 0.7071$	$q_1 = 1.4142$	$(p^2 + 1.4142p + 1)$
			$(p^2 + 1.4142p + 1)$

BESSEL (see note 2)

Order	m	q	equation
first (6dB/octave)	0	0	
second (12dB/octave)	$m_1 = 0.6808$	$q_1 = 1.9077$	$(0.6180p^2 + 1.3616p + 1)$
third (18dB/octave)	$m_1 = 0.4998$	$q_1 = 1.9547$	
	$m_2 = 0.7560$		$(0.4771p^2 + 0.9996p + 1)$
			$(0.756p + 1)$
fourth (24dB/octave)	$m_1 = 0.3871$	$q_1 = 1.0048$	
	$m_1 = 0.6698$	$q_2 = 0.7298$	$(0.3889p^2 + 0.7742p + 1)$
			$(0.4889p^2 + 1.3396p + 1)$

Note 1: The coefficients in m and q are valid for low- and high-pass filters. All coefficients and equations are from the book *Filtre actif* by Paul Bildstein, Editions Radio, 9 rue Jacob 75006 Paris.
Note 2: For Bessel, the coefficients m and q are only valid for a low-pass filter (ibid., p. 54).

S12 is the power supply pin (+12V for the recommended relay).

Figure 4 is the block schematic for the three-way extension board; *Fig. 5* is the schematic and wiring diagram. The circuit consists of an LP (B2, B3) and an HP section (B5, B6), connected in cascade and forming a bandpass filter. The input signal is taken from the input buffer's output on the two-way filter board. The bandpass filter's output is converted to a balanced signal by B9 and B10. You can remove B10 from the board if the balanced output is not needed. Second- and third-order filters can be implemented as previously described.

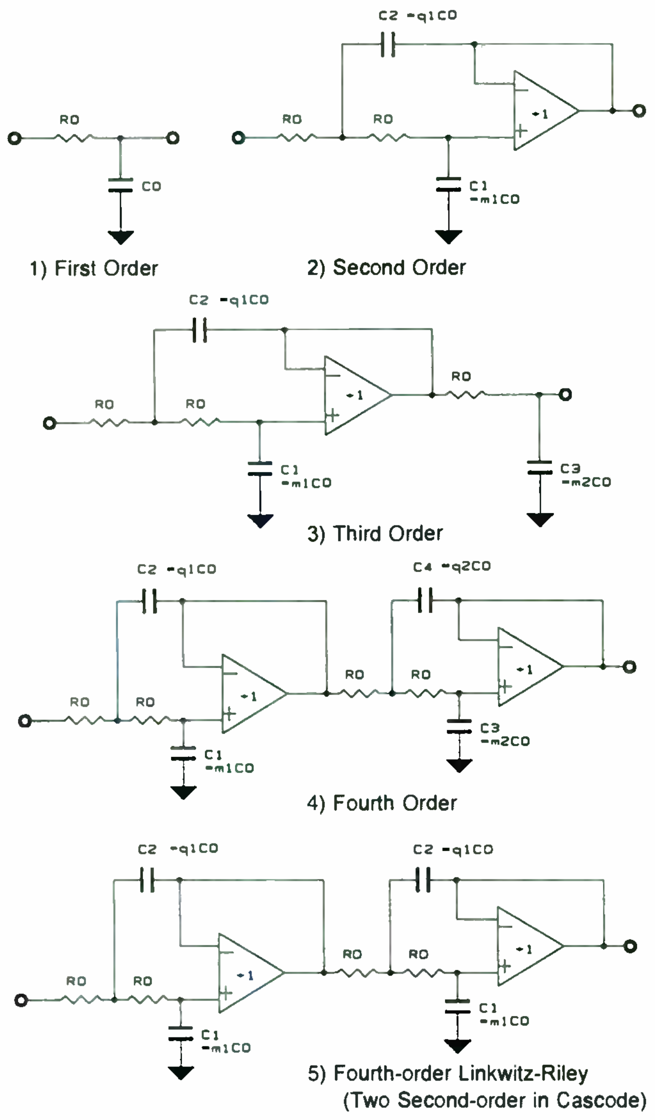
We used ORCAD for all schematics and the new LAYO1 software for the PC board layouts. When we tested the laser printer check plot and the Gerber files, we found that the lines and pads appeared thinner on the check plot than on the screen. Both outputs, however, are usable. For the final prototypes, we used the Gerber output files with very good results. [*Note: LAYO1, an impressive package at an affordable price, is available from Old Colony Sound Lab—see the Availability Box.—Ed.*]

On the motherboard, the regulator heatsinks are grounded. Consequently, the latter must be insulated with mica or other insulator materials. The motherboard is configured as a two-way crossover (bold lines indicate the jumpers). This board is fully populated, with all four networks and all ten buffers installed. In many cases, you will be using fewer than the full complement. Except

= 1.0823, q_2 = 2.6131. *Figure B4* gives $R_1 = R_0/m_1$, $R_3 = R_0/m_2$, $R_2 = R_0/q_1$, $R_4 = R_0/q_2$. We choose C_0 = 6,800pF according to equation (2) R_0 = 11,702Ω, R_1 = 12,668Ω, R_3 = 30,587Ω, R_2 = 10,812Ω, R_4 = 4,478Ω. P3 and P4 will be needed (*Fig. 1*). P3: $Z_1 = Z_2 = C_0$ = 6,800pF, $Z_3 = R_1$ = 12,668Ω (12k1+562), $Z_4 = R_2$ = 10,812Ω (10k+825). P4: $Z_1 = Z_2 = C_0$ = 6,800pF, $Z_3 = R_3$ = 30,587Ω (30k1+475), $Z_4 = R_4$ = 4,478Ω (4k32+150).

Fourth-order Linkwitz-Riley:
Two second-order Butterworths in cascade. We have already calculated the values, so the passive network P3 and P4 will be the same, as you can see in *Fig. B5*. P3: $Z_1 = Z_2 = C_0$ = 5,600pF, $Z_3 = R_1$ = 20k, $Z_4 = R_2$ = 10k. P4: $Z_1 = Z_2 = C_0$ = 5,600pF, $Z_3 = R_1$ = 20k, $Z_4 = R_2$ = 10k.

As another example, look at *Fig. 7*, which shows the actual response of one of the prototypes with a cutoff frequency of 140Hz.



1) First Order

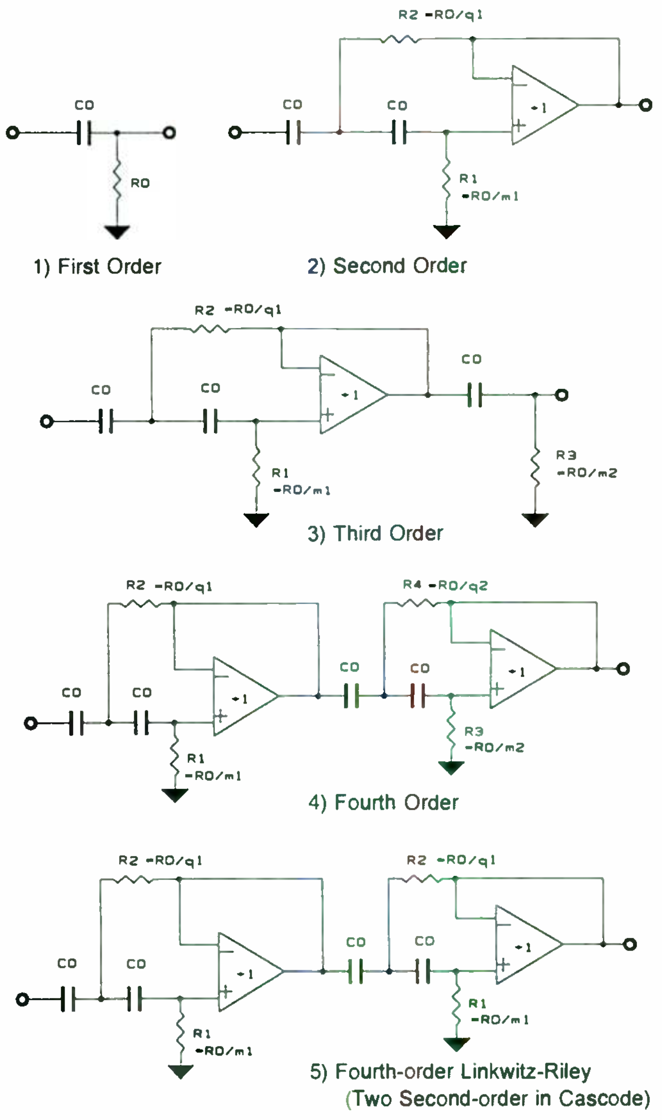
2) Second Order

3) Third Order

4) Fourth Order

5) Fourth-order Linkwitz-Riley (Two Second-order in Cascode)

FIGURE A: Low-pass structure.



1) First Order

2) Second Order

3) Third Order

4) Fourth Order

5) Fourth-order Linkwitz-Riley (Two Second-order in Cascode)

FIGURE B: High-pass structure.

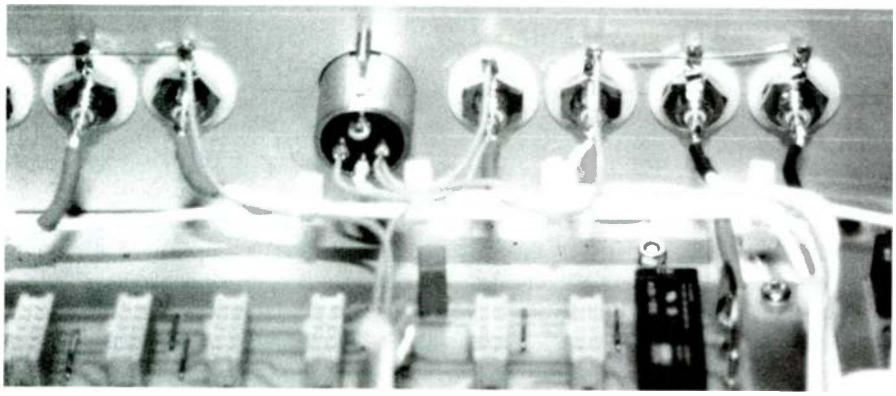


PHOTO 4: The rear panel. Note the way we made the ground connections; the ground input pin (the cinch on the right of the XLR) is connected to the signal ground on the motherboard. The three ground output pins are connected together and soldered to the star ground. Signal ground and power ground are connected at one position only for each channel. The best we found was to connect Pin S14 and S23.

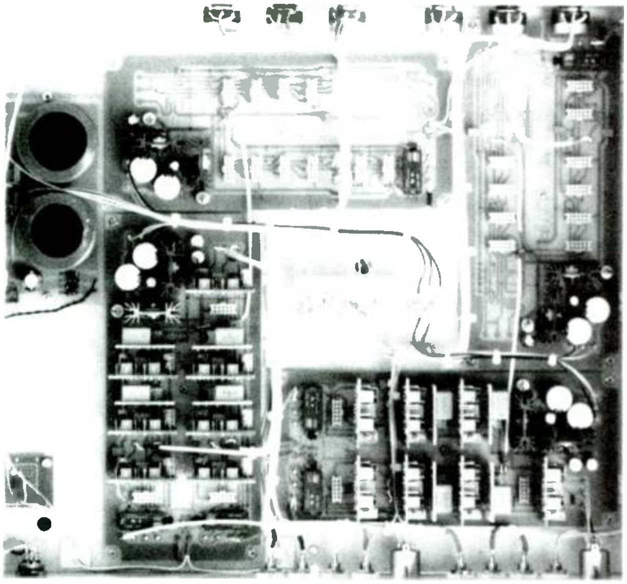


PHOTO 5: Top view of the measured crossover in a two-way asymmetrical input/output system, with the needed buffer modules and passive network. To go to three-way or symmetric, just plug in module and play.

for B4, B8, and B10, which are dedicated for balanced operation, you must always install a jumper between input and output for the missing unit. Otherwise, the next unit will not receive a signal. If you have installed connectors for each unit, but don't use all of them, you can install a wire jumper in the connector itself. [*Note: Board patterns and stuffing guides are available upon request. See the Availability Box for details.—Ed.*]

The volume control also needs bypassing if not used, as do the muting relays at the output. If you don't need the muting function, don't install them. The points marked X and Y on the stuffing guides indicate the location of the interconnection cable between the two motherboards (for a three-way system), which should be shielded. When the motherboard is used as a three-way extension board or bandpass filter, the optional volume control is connected to Pins S6, S7, and S8. Again, those units not used must be bypassed with a jumper.

The motherboard setup procedure is very simple. Simply check that regulators U2 and U3 are working. Apply a ±28V unregulated voltage to Pins S17, S18, and S24, and check the output voltages across filter capacitors C17 and C18. They should be close to ±24V. The buffers should be checked separately (see the BUF 124 setup procedure). Overall frequency response of the filter sections should be checked with buffers and networks installed.

UNITY-GAIN BUFFER

The heart of the crossover is the buffer, around which you build the filter sections. The schematic of the discrete BUF 124 is shown in *Fig. 6*. Originally developed as an input and tape buffer, it has been updated and optimized for filter operation. It features very low distortion and a rise-and-fall time of 200ns.

The BUF 124 two-stage circuit consists of a JFET-cascode input stage and a bipolar-cascode second stage—both pure Class A. Due to the cascode-connected JFETs, which operate at 2mA drain current, the input stage features very low input capacitance and high

Continued on page 26

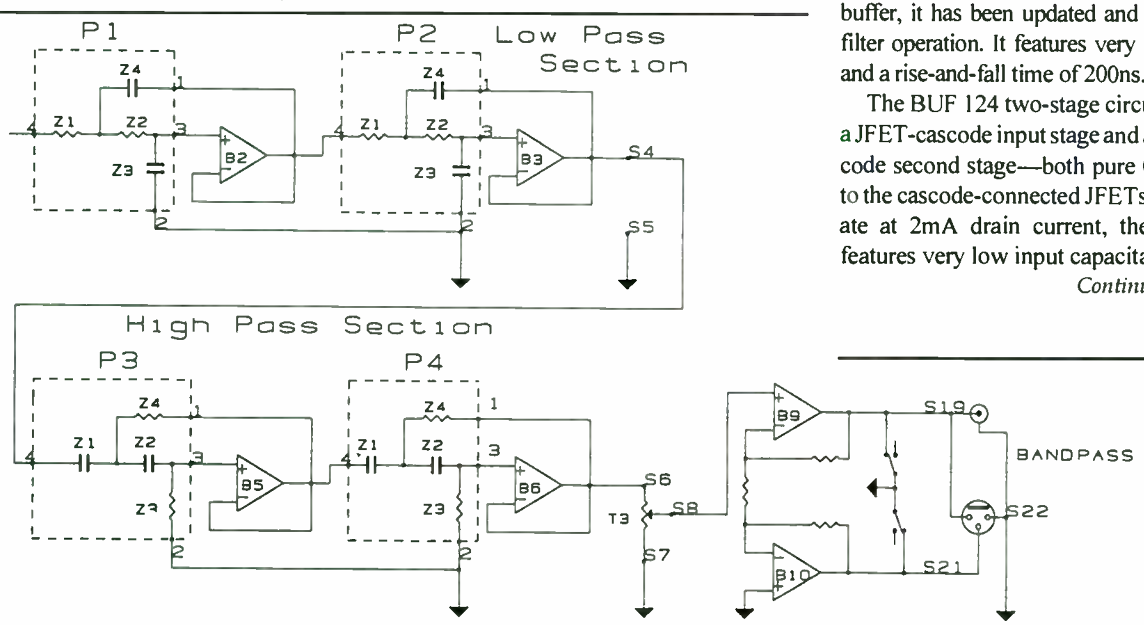


FIGURE 4: Three-way extension board block schematic. Power supply is ±24V. Switch on mute.

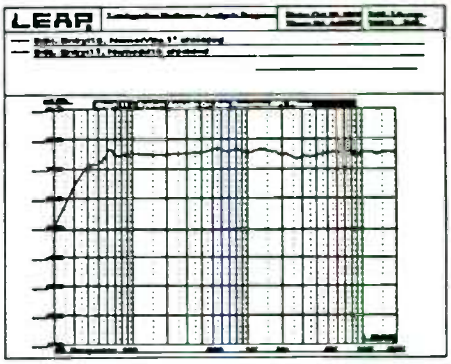
THE CENTER OF IT ALL

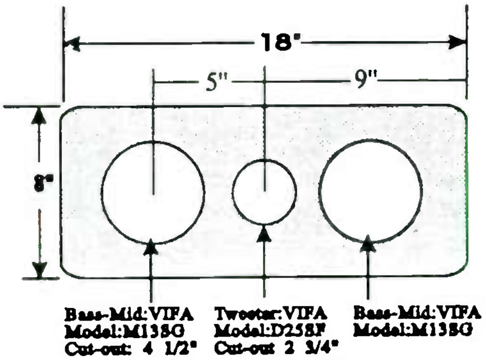
The center channel, often thought of as responsible for only reproducing dialogue, is actually responsible for reproducing about two thirds of the total acoustic energy of a typical movie. Much of the film's music track and a great deal of the sound effects originate from the center channel. This leaves the front, main channel mostly responsible for augmenting the center channel's foundation with far left and right information; and this leaves the rear channel responsible for spatial ambience. This being the case, the center channel must be well designed and constructed with high quality components, and when done well, it should have a similar tonal balance to that of the front, main speakers. We at A&S Speakers following these guidelines, have developed a top quality center speaker. Vifa components, among the most respected and widely used in the high-end market, are used in our center speaker as well as all of our A/V satellite speakers. This insures excellent quality and tonal consistency in our A/V packages.

THE CENTER CHANNEL

TECHNICAL SPECIFICATIONS

Frequency response: 55 Hz - 20kHz +/-3db
Sensitivity: 88 db
Nominal Impedance: 8 ohm
Thermal power capability: 100 watts RMS
Crossover frequency: 2800 Hz





FRONT VIEW

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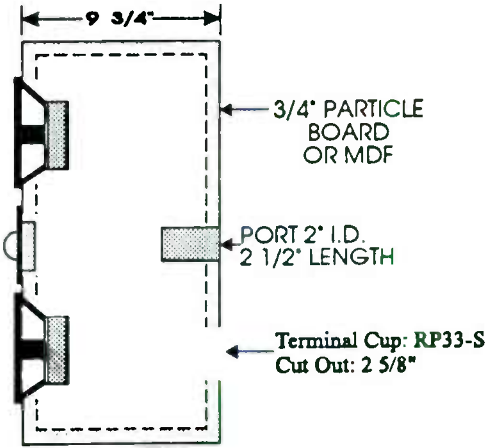

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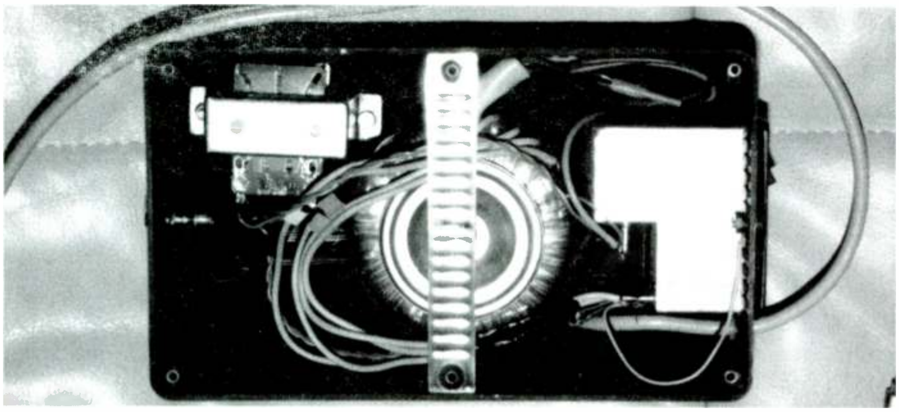


PHOTO 6: External power supply box with a conventional transformer (top left) for the relays and a toroidal transformer for the buffers. We used an AC filter, seen in the right of the photo.

Continued from page 24

common-mode voltage swing. Q1 and Q3 must be matched to ±10% of I_{DSS} (either the "BL" or the "V" group can be used).

To adjust the DC offset at the output, P1 = 200Ω. R4/R5 and R'4/R'5 are connected in series and in parallel with P1, respectively, and are used to set the current in the input stage to 2mA. You can check the current by connecting a DVM across R3 (or R6) and measuring the voltage drop across the resistor, which at 2mA is 2.8V. If the drain current is higher than 2mA (i.e., the voltage drop is >2.8V), you must connect R4 and R5 in the circuit to reduce it. If it is less than 2mA, connect R'4 and R'5 in parallel with P1 to increase the current.

The second stage operates at approximately 15mA. When the input stage is operating at 2mA, this current is set up automatically to the correct value. D1 and D2 are used to bias the cascode transistors Q6 and Q7 relative to the emitter voltage of Q5 and Q8. R8 forces a current of approximately 1mA through the reference diodes.

R10, R16, and C4 form the buffer's feedback network. In unity-gain application (i.e., when the buffer is used as a filter), R16 is not connected to ground. It is used only when the buffer is connected as a balanced-to-single or single-to-balanced converter. In balanced operation, the gain of the input and output buffer is higher than unity. R9 and C6 form the amplifier's output network, isolating the buffer from capacitive loads. Since it works in pure Class A, the buffer is virtually short-circuit proof—you can short its output with a relay for muting purposes.

The buffer is connected to the motherboard through a 10-pin gold-plated connector. You can solder it to the motherboard or install a mating connector on the board and plug it in. [*Note: The BUF 124 is available as a kit. See the Availability Box.—Ed.*]

BUFFER SETUP PROCEDURE

Test each buffer module separately, if possible, before installing it on the motherboard. This simplifies measurements, adjustments, and any necessary component changes. If you have access to a scope, connect it to the buffer's output and check for radio frequency (RF) oscillations. If you have complete audio instrumentation in your workshop, perform the usual gain, frequency response, noise, total harmonic distortion (THD), and intermodulation distortion (IM) measurements. Inputs should be shorted under DC measurements/adjustments.

Before testing, set P1 to mid-position. Short signal ground to power supply ground

Continued on page 28

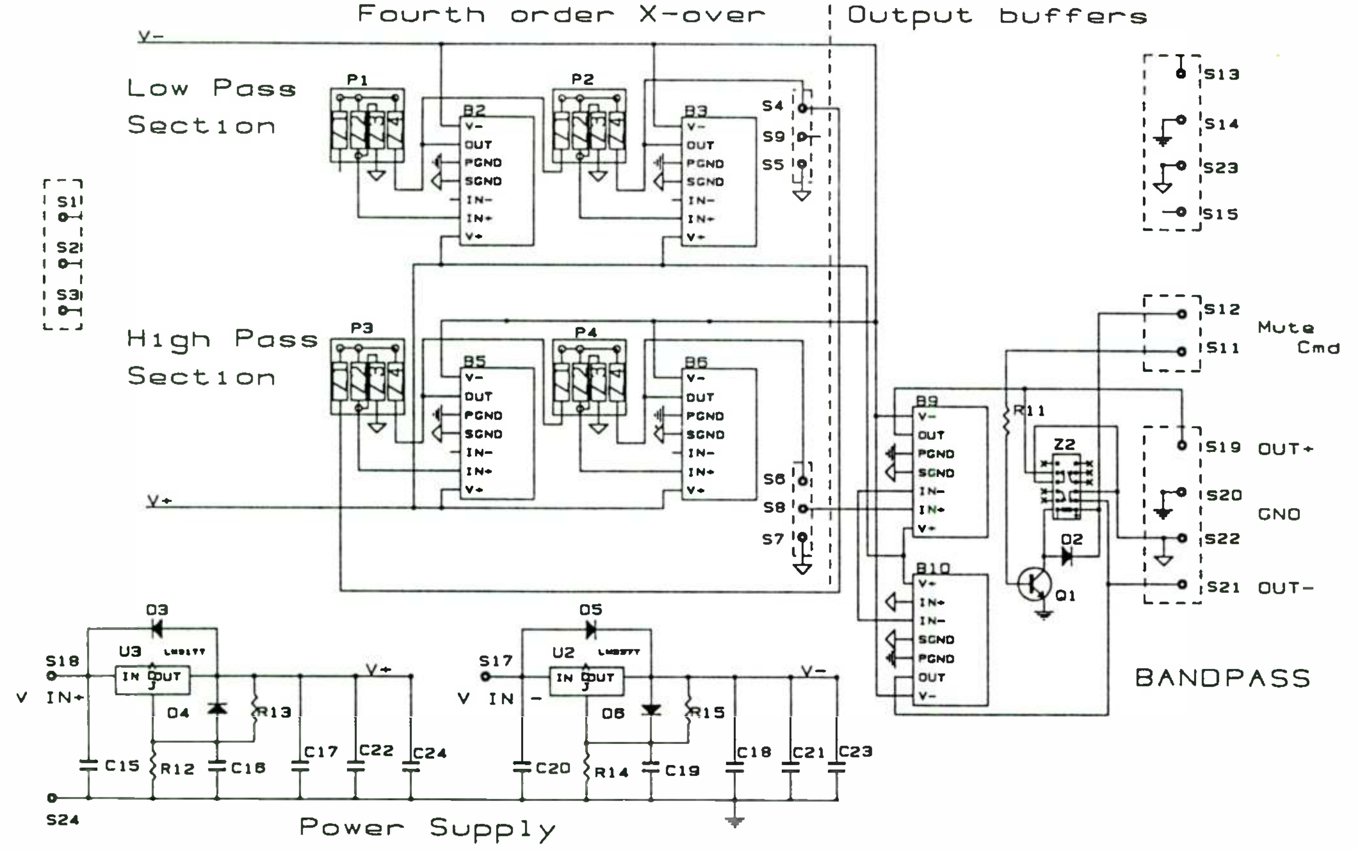


FIGURE 5: Three-way extension board schematic and wiring diagram.

BLACK HOLE

High efficiency dampening products from ORCA

A better speaker damping material...

If you've been building speakers for some time, you know how much guesswork goes with speaker damping and stuffing. The choices seem endless: fiberglass, wool, Dacron, flat foam, convoluted foam, felt, tar, plus various "magic" compounds that you're invited to brush or pour into your new cabinets. Everyone has their own recipe, and who knows if it's a recipe for disaster? Or what effects the vapors emitted by these chemicals might have on the glues that bond your woofer surround to its cone and chassis? In this era of costly, space-age drivers and computer-assisted design, we think such risks are totally unacceptable. So we went to work to find the ideal solution.

The problems are fairly well-known: a driver transforms electrical energy into mechanical energy. This mechanical energy is transformed into acoustical energy which is radiated to the outside of the cabinet - the useful front wave - and to the inside - the sometimes-useful back wave. Unfortunately, it is also transmitted though the frame of the driver to the cabinet itself, which acts as a very large "cone" of very small excursion. This means that the spurious resonances and vibrations of the cabinet have to be controlled in a predictable and reproduceable way. That's how we came to BLACK HOLE 5 and the BLACK HOLE PAD.

First, THE PAD. It's a thin (1/16 inch) black flexible viscoelastic damping material (filled vinyl copolymer) with maximum performance between 50 and 100 degrees F (we hope that that covers the temperature range of your listening room) and excellent flame resistance - it meets UL94 V-O. Thanks to its outstanding damping characteristics, THE PAD will dramatically reduce the vibration energy stored in the walls to which it is applied.

Easy to cut and apply, THE PAD has a pressure-sensitive adhesive back: simply peel off the release paper and press hard onto a clean surface. You can use THE PAD on just about anything you suspect of vibrating: driver frames, thin panels like car doors, and, of course, the walls of your speaker cabinets. And it can be used to recess a driver without using a router: just laminate enough layers to match the thickness of the driver frame and apply to the front baffle. Finally, it is the ideal material for "constrained layer" wall construction, where two panels are laminated on each side of a damping material for optimum transmission loss. Because THE PAD has a fine grain leather finish, you can wrap an entire cabinet exterior and give it an attractive appearance at the same time!

For applications which require **maximum damping, isolation and absorption,** we've developed BLACK HOLE 5. One and 3/8" thick, BLACK HOLE 5 is a high-loss laminate that provides optimum acoustical damping performance. It consists of five layers:

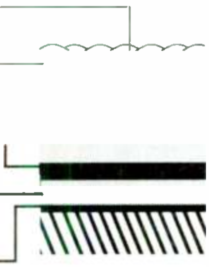
Thin diamond-pattern embossing, densified with a polyurethane film surface. This unique surface layer dramatically improves the performance of the whole acoustical system, especially the lower mid-range and mid-bass frequencies where simple acoustical foam loses its effectiveness.

One-inch deep polyester urethane foam, structurally optimized for acoustical damping. Highly effective at "soaking" maximum sound energy with minimum thickness.

Barrier septum, 1/8 inch thick. Made of limp flexible vinyl copolymer loaded with non-lead inorganic fillers, it is a "dead wall" that isolates the vibrations in the walls of your cabinet from the vibrations created inside the enclosure.

Polyester urethane flexible open-cell foam, 1/4 inch thick. Thanks to special vibration-isolation characteristics, it decouples the vibrating structure (the wall) from the rest of the damping system, thus optimizing performance.

High-loss vibration damping material, same as The Pad. It is strongly bonded to the cabinet wall with pressure sensitive adhesive.



These layers are laminated using an adhesive-free mechanical and thermal process, thus optimizing performance and eliminating the risk of solvent fume damage. BLACK HOLE 5 can be used in any enclosure, as well as for acoustical panels to improve the characteristics of your listening room. *YOU PROVIDE THE MUSIC; BLACK HOLE FIVE WILL TAKE CARE OF THE NOISE!*

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AΞON Cables

New from ORCA!

AX-ON (Greek axon, axis): that part of a nerve cell through which impulses travel away from the cell body. AXON 8 speaker cable combines outstanding design features with component quality usually associated with the most expensive cable. With eight AXON 1 solid-core conductors and utilizing mylar/ polypropylene construction, AXON 8 offers outstanding performance for amp-speaker connections and perfectionist internal speaker wiring. Our superb AXON 1 AWG 20 solid core conductor is also available separately. Oxygen-free and 99.997% pure, it is ideal for most internal wiring applications.

Outer insulation: UL approved TPE
Cable geometry: non interleaved spiral
Individual conductor insulation: 105 degree Celsius, UL approved PVC
Cable equivalent gauge: total - AWG 11, 2 conductors - AWG 17, 4 conductors - AWG 14
Individual conductors: solid core AWG 20 copper, long-grain and ultra-soft, free of all contaminants and oxygen.
Cable core: crushed polypropylene
Inner envelope: mylar film

Reader Service #32

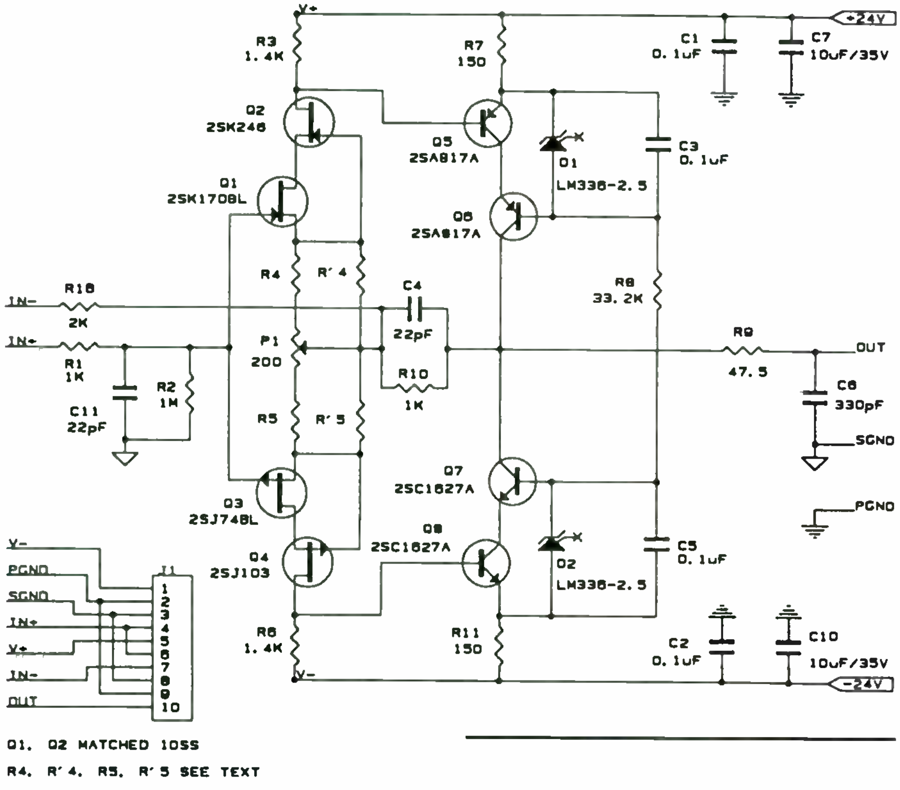


FIGURE 6: BUF-124 schematic.

TABLE 1

BUF 124 PARTS LIST

PART	DESCRIPTION
Resistors	
R1	1k*
R2	1M
R3, 6	1.4k
R4, 5	Select for I_D = 2mA
R4′, 5′	Select for I_D = 2mA
R7, 11	150
R8	33.2k
R9	47.5
R16	2k (For balanced operation)
Trimpot	
P1	200 multiturn cermet
Capacitors	
C	22p/630V PP, MICA, COG
C1,2,3,5	0.1µ/63V WIMA MKS-2
C4	22p/630V PP, MICA, COG
C6	330p/630V PP
C7, 10	10µF/35V TA
Semiconductors	
Q1	2SK170BL**
Q2	2SK246
Q3	2SJ74BL**
Q4	2SJ103
Q5, 6	2SA817A
Q7, 8	2SC1627A
D1, 2	LM336Z-2.5

Miscellaneous

10-pin connector, 3M type number: 2510-5002

BUF 124 PC board

*All resistors 0.5W/1% metal film, Resista MK-2 or equivalent.

**Q1 and Q3 are matched to 10% of I_{DSS}.

Continued from page 26

at the output. Connect ±24V regulated supply to the module and perform the following measurements/adjustments:

1. Connect a voltmeter across R3 (or R6) and measure the voltage drop. It should be 2.8–3V. If it is less than 2.8V, install R4′ and R5′. If the voltage is more than 3V, install R4 and R5. (The resistor values depend upon the I_{DSS} of Q1 and Q3. Some experimentation will be necessary to determine the correct values.) The BUF 124 kit is sold with matched/marked Q1/Q3, which makes selecting the right resistor values easy.
2. Connect a millivoltmeter to the amplifier's output, and, with P1, adjust the offset to 0V.

CROSSOVER NETWORKS

The crossover networks are mounted on plug-in PC boards. Separate PC boards are available for the low- and high-pass networks. The crossover networks are connected to the motherboard through the same kind of connectors as the buffers. For calculation of the crossover network, refer to the sidebar "Calculating the Crossover Network."

AVAILABILITY BOX

To obtain prints of layouts and stuffing guides, please send a 9″ × 12″ manilla SASE with postage for 2 oz. (international readers, please include postal coupons) to: *SB*, PO Box 494, Dept. EB, Peterborough, NH 03458-0494.

Remarkable LAYO1 PC board design software is available in packages priced as low as $99 for 4,000-point capability. For a free color information packet, contact Old Colony Sound Lab, PO Box 243, Dept. EB, Peterborough, NH 03458; (603) 924-6371, FAX (603) 924-9467.

BUF 124 KIT. The BUF 124 is sold as a single board. Kit includes drilled PC board, all resistors, capacitors, and semiconductors. Components are packed in plastic bags and are marked with component number and/or value. We reserve the right to substitute components of equal quality. Contact: Borbely Audio, Melchior Fanger Strasse 34A, 82205 Neu-Gilching, Federal Republic of Germany, 011-49-8105-5291, FAX 011-49-8105-24605.

2SK170BL/2SJ74BL FETs are also available separately, matched and marked with I_{DSS}. Contact Borbely Audio for information.

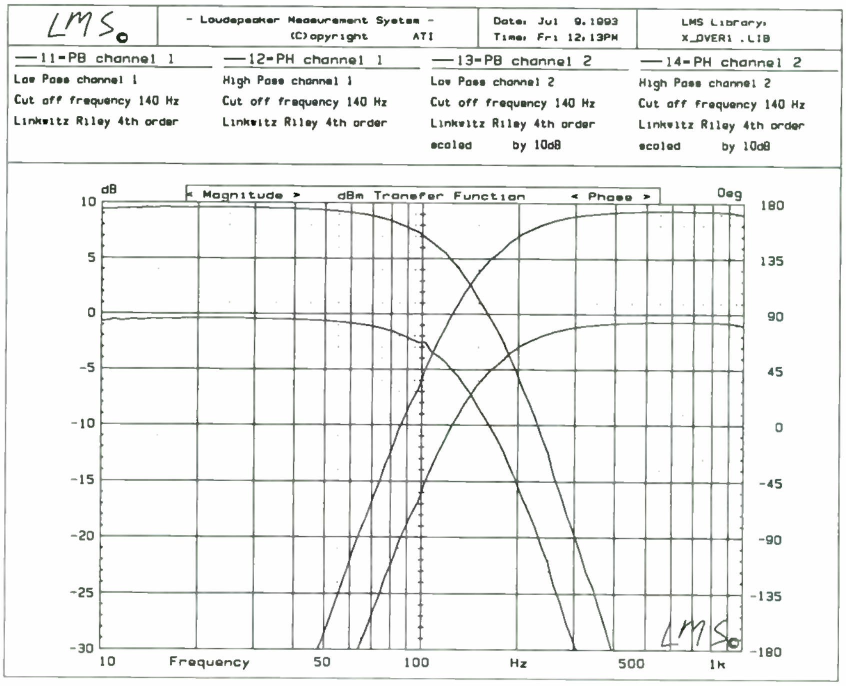


FIGURE 7: Response of the crossover populated as a two-way system. The cutoff frequency is 140Hz in a fourth-order Linkwitz-Riley mode. Only one value of capacitor and two for the resistors was needed: 0.1µF/160V WIMA MKP2; 7.5k and 15k 1% metal film resistors.

TABLE 2

TWO-WAY MOTHERBOARD PARTS LIST

PART	DESCRIPTION
Resistors	
R7, 11	47k
R12, 14	2,210
R13, 15	121
Capacitors	
C15, 20	1,000µF/35V radial
C16–19	47µF/35V radial
C21–24	0.1µF/100V WIMA MKS02
Semiconductors	
Q1, 2	BC337 or equivalent
U2	M337 TO220
U3	LM317 TO220
D1–6	1N4007 or equivalent
Miscellaneous	
B1–5	
B6–10	
P1–4	10-connector 3M ref. 8510-4500
Z1, 2	SDS Relay S2-12V
Heatsink	Schaffner ref. WA 337-25, 4 or 38, 1
U1	2-pin removable jumper

TABLE 3

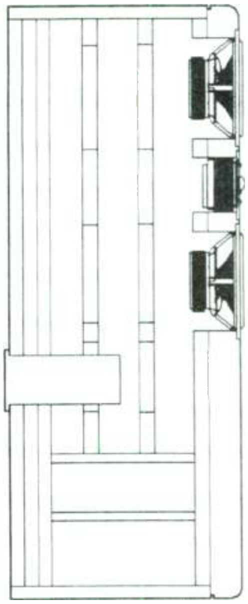
THREE-WAY MOTHERBOARD PARTS LIST

PART	DESCRIPTION
Resistors	
R11	47k
R12, 14	2,210
R13, 15	121
Capacitors	
C15, 20	1,000µF/35V radial
C16–19	47µF/35V radial
C21–24	0.1µF/100V WIMA MKS02
Semiconductors	
Q1	BC337 or equivalent
U2	LM337 TO220
U3	LM317 TO220
D2–6	1N4007 or equivalent
Miscellaneous	
B2–6, 9, 10	
P1–4	10-connector 3M ref. 8510-4500
Z2	SDS Relay S2-12V
Heatsink	Schaffner ref. WA 337-25, 4 or 38, 1

MOUTH

Word of mouth helps us grow, and our growth means a stronger publication that can do more of the things that need doing in the pursuit of better audio systems. If you have friends, associates, relatives or even enemies who share your enthusiasm for *SB*, either let us send you prospectuses to pass along to them or tell us their names and addresses, and we will send the word along. THANKS.

POWER



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World Radio History

A FULL-RANGE OPEN-BAFFLE SYSTEM

By Warren Hunt and Joseph Janni

Of the various speaker system designs, the open-baffle is one of the most misunderstood. We have built a full-range open-baffle system that performs superbly. It operates as a dipole radiator at all frequencies, yet has astonishing deep-bass performance.

We have always found particularly appealing sound qualities in unenclosed speakers (Apogee, Audiostatic, Carver, Magnepan, Martin-Logan, Quad, among others), which share a common design feature: rather exotic technology is used to produce their superb sound. Unfortunately, this technology is usually too difficult for an amateur to work with. We were determined, however, to reproduce these sound qualities with conventional, readily available technologies. As we have not found our design approach discussed anywhere in the audiophile literature, we will describe it in considerable detail.

OPEN ADVANTAGES

One of the best approaches to enclosure design is to eliminate as much of the structure as possible, and an open baffle is the simplest solution. It consists of one or more drivers mounted on a supporting structure that is usually flat and completely open in the rear. The baffle does not attempt to restrict the rear sound emissions in any way, nor is it an enclosure in the conventional sense.

An open baffle is easier to construct than traditional enclosures. More so than the sealed box, and certainly much simpler than vented, ducted-port, transmission-line, or folded-horn enclosures.

Although front panel vibrations occur in any design, open baffles have fewer panels to vibrate than conventional six-sided enclosures. The vibrations are caused by the action/reaction of the front panel to the direct motion of the driver cones. With a total weight of 60 lbs. and a speaker moving mass of 0.3 lbs., the action/reaction ratio of our open baffles is very favorable.

ABOUT THE AUTHORS

Warren Hunt is a technical supervisor at a film and video editing company in Hollywood, with an associate's degree from the Manhattan Community College of New York. He has been interested in speaker design and woodworking for several years, and has built a variety of cabinets and enclosures. Joseph Janni is the chief scientist of a large government laboratory. He holds a PhD in engineering from the University of New Mexico, and has been designing speaker systems for over twenty years.

PHOTO 1: The front of the open baffle final prototype, showing the placement of the individual drivers and the side panels.

Conventional enclosures have additional sources of panel motions caused by internal pressure increases and decreases that do not occur in open baffles. These vibrations are quite audible and can add undesirable colorations—"cabinet talk"—which can be almost as loud as the primary speaker output. For examples of these effects, refer to the speaker reviews in any recent issue of *Stereophile* magazine, where cabinet vibrations are measured separately.

Our design has excellent imaging, primarily because open baffles are dipole radiators. Dipoles emit sound equally in the forward and rearward directions, but the rear emission is 180° out-of-phase. By directing more sound toward the listener and away from the side walls, they improve the imaging. We think our design produces sufficient wall reflections to provide pleasing room ambience, but not so much as to degrade imaging. The rear emission reflects off the back wall, which adds a very pleasing and spacious quality to the sound at mid and high frequencies. For this article, we have defined the back wall as being directly behind the speaker system.

Another significant factor is that the preferentially forward and rearward emission pattern of open baffles allows the acoustics of the recording to predominate over those of the listening room. This consideration is particularly important for superior-quality recordings.

Our open baffle is inherently less sensitive to room placement than conventional enclosures due to its dipole radiation pattern, and because we used multiple drivers.[1] Soft-dome tweeters plus three cone drivers are aligned vertically, with the third driver near the floor as a subwoofer. Two identical tweeters, one forward- and one rearward-firing out-of-phase, are located midway between the two 10″ cone drivers in a D'Appolito configuration.

We experimented with locations as close as 8″ and as far as 3′ from the back wall, and verified our conclusions with several measurements and listening tests. Although the closer location performed well, the sound changed somewhat in character, with more deep bass and improved imaging in the 3′ location. At 2.5′ and beyond, the sound assumed a warmer and richer quality. Our system was not very sensitive to side wall distance. We probably obtained these encouraging results because an open baffle minimizes the formation of standing waves in the room.

Driver selection is simpler than for any other enclosure type, because the final system resonance and transient response are easier to determine. With open-baffle mounting, the driver free-air resonances change by less than 2Hz. For all practical purposes, the resonances of drivers in an open baffle and their free-air resonances are the same. Their transient response is also unchanged, which greatly simplifies driver selection.

THE FLIP SIDE

The predominant disadvantage of full-range open-baffle systems seems to be the widespread lack of understanding of their correct design principles. This became apparent when we unsuccessfully searched a dozen books for technical design information.

Driver suppliers aren't accustomed to open baffles, either. After we built the second prototype and verified that deep bass was possible using a 12" subwoofer, we decided to upgrade to a 15" unit. We telephoned several suppliers with some specific questions aimed at optimum driver selection. Most suppliers listened to our questions, then asked what we were doing. When we explained that we had already built a successful full-range open-baffle system and were planning to increase the subwoofer size, they usually told us politely that such a system couldn't possibly work at low frequencies; nevertheless, they provided the information we requested.

Open-baffle systems are not necessarily less efficient at low frequencies than other enclosures. Without the subwoofer, our frequency response is about the same as a comparable sealed box—both measured outdoors—having a system resonance of 50Hz, free-air driver resonance of about 25Hz, and a critically damped ($Q_S = 0.5$) response. Our passband efficiency above 100Hz is virtually identical to the sealed box. If no corrective equalization were applied to either system, both would be about 14dB down at 25Hz and they would have similar outputs for the same electrical input.

Unfortunately, when open baffles are indoors and close to a back wall, efficiency decreases by several decibels. Conversely, sealed box efficiency can increase when properly placed near rear and side walls. Efficiency isn't everything, however, and the open baffle's improved imaging and transparency more than compensate for this inadequacy.

The acoustics of conventional open baffles require a very large physical size in order to produce a flat frequency response down to the audibility limit. For example, a 10' × 10' open baffle would be 6dB down at about 27Hz. Rather than consider an unrealistically large baffle, we chose to attenuate the mid and high frequencies with a low-pass circuit to balance the smaller baffle's deep-bass self-cancellation.

Our low-frequency response is further improved by adding a subwoofer below 70Hz to increase the output. This doubled our power draw at very low frequencies, but increased our acoustical amplitude where we most needed it. We used the low-pass circuit to drive all cones, including the subwoofer, with a boost of 9dB at 25Hz. The extra design work was worth it, because our frequency response is nearly flat down to 25Hz; however, at these frequencies there is considerable cone excursion at the loudest levels.

RESPONSE MEASUREMENTS

The common misconception that you can't get deep bass from an open baffle is due to the front radiation's cancellation by the rear out-of-phase radiation. As you will see, this seemingly insurmountable problem can be solved. In fact, our system's frequency response will dispel any concerns you may have about its performance.

Photos 1 and *2* show the final prototype, front and rear; *Fig. 1* is the frequency response. The measurement was taken outdoors, 6' in front of our prototype perpendicular to the tweeter. This response includes the effects of our low-pass circuit, which is a fundamental design element, and our subwoofer resonant crossover (both of which we will describe later). We used no additional parametric equalization. The 1/3-octave warble tones on the second *Stereophile* test CD served as a sound source for all of our measurements.

PHOTO 2: The rear of the open baffle final prototype, with the wiring only partially connected.

In this geometry, the acoustical effects of the ground (a large flat patio) are included. This arrangement is more realistic and easier to standardize than a purely anechoic environment, hence it is more appropriate. Of course, no wall or ceiling effects exist with outdoor measurements, and we were sufficiently distant from any vertical structures to eliminate their effects.

As you can see from *Fig. 1*, the frequency response is generally within 3dB from 25Hz–20kHz. This extraordinary response would be noteworthy for any type of speaker system.

The rise near 150Hz is a cabinet-shape effect produced by the depth of the perpendicular side panels, which becomes noticeable when the average depth of the sides is greater than 8".[2] Our side panels are tapered, with a 4" minimum depth at the top and a maximum of 9" at the bottom. The average depth is 6.5". While eliminating the side panels would reduce this rise, we retained them to ensure the front panel's rigidity as well as to have self-standing units.

The 2dB drop at 4kHz appears in the tweeter manufacturer's published frequency response plots. We independently measured the same trend with our inexpensive approach, which was encouraging.

TOMBSTONE TERRITORY

The baffle's shape evolved from aesthetic considerations. After sketching a number of different possibilities, we experimented with the proportions of the most appealing ones. Since we couldn't consider anything close to "bookshelf" dimensions, we set the size limitations of 24" (width) and 54" (height). A high rectangular baffle with rounded top corners and sides sloping down to a wider base became our favorite.

The advantages of sloping sides became apparent as we added acoustical design considerations. Our goal was a self-supporting baffle, with a base and side panels deep enough to hide the drivers' frame when viewed from a profile. After experimenting with the first prototype, we settled on the final dimensions, which through careful listening and additional measurements proved themselves on the second prototype.

Listening experiments revealed that the subjective sound quality improved slightly when we tilted the top of the baffle back a few inches. After listening to the sound with several different configurations, we settled on a 6° rearward tilt.

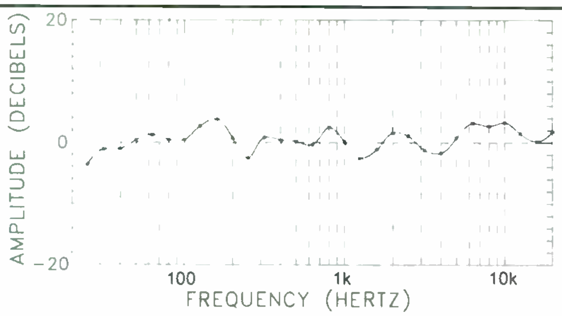

FIGURE 1: The system frequency response measured outdoors 6′ from the open baffle. The microphones were perpendicular to the tweeter. The sound sources were ⅓-octave warble tones.

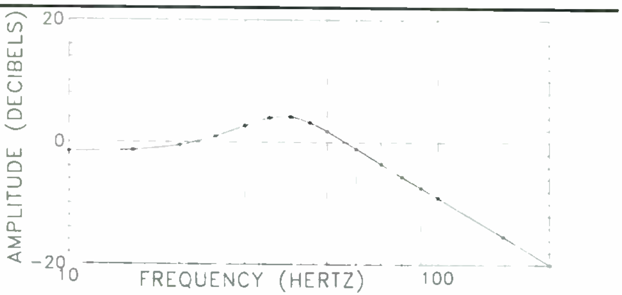

FIGURE 3: Frequency response of the low-pass resonant crossover connected to the subwoofer. The subwoofer is boosted 4dB at 35Hz by the resonance and merged with the midranges at about 70Hz to form a smooth transition.

A practical open baffle's shape must have diverse pathlengths (sound paths) from the rear to the front. This will eliminate the possibility that either a deep cancellation notch or a strong reinforcement peak will occur at any specific frequency. In order to provide the necessary distribution spread, the maximum pathlength should be at least twice that of the minimum.

Our maximum pathlength, from the subwoofer front to the baffle's top corner and back to the subwoofer rear, is 7.5′; the minimum pathlength is 2′. Since the former is more than twice the distance of the latter, there should be no problem with cancellation notches or reinforcement peaks. Our measurements bear this out.

Another benefit of this design is that each driver has a unique pathlength distribution, a result of the baffle's deliberately tapered shape. As an added advantage, our speakers have a distinctive appearance. Friends have taken to calling them the "tombstones."

DRIVER SELECTION

1. To design an open baffle, seven driver parameters are required: power handling, driver sensitivity in decibels/watts/meters, free-air resonance (f_O), maximum cone excursion (X_{MAX}), and the mechanical, electrical, and total resonant factors (Q_M, Q_E, and Q_T). Some of these are not T/S parameters, because open-baffle designs do not use this theory.

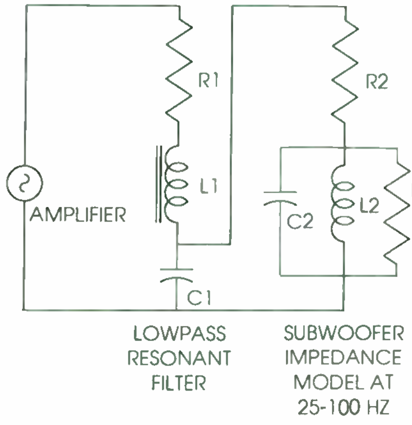

FIGURE 2: Subwoofer resonant crossover with Q = 1.6 and 4dB of electrical gain.

2. All drivers except the tweeter should have nearly identical f_Os, regardless of their diameters. This allowed us to use the same low-pass circuit for all the drivers.

3. Every driver must have a long throw capability (i.e., large X_{MAX}). A peak-to-center throw of at least 5 mm is needed.

4. The f_O should be about 25Hz, because it establishes the lowest frequency the design can reproduce. Below this resonance, the system response drops very rapidly.

5. The total system transient response Q_S is determined by the driver Q_T, which in turn is dominated by Q_E. Ideally, Q_S should be about 0.5; however, values ranging from 0.25–0.75 are acceptable and should not degrade the transient response to the point of audibility.

6. While the mechanical Q_M is not particularly important since Q_E dominates, it should be in the range of 3–6.

7. The driver electrical Q_E should be about 0.5, since it dominates the system damping.

8. An alternate design uses drivers with an inherently high Q_T of 4. The acoustical output would then increase steadily at progressively lower frequencies (approaching resonance) and cancel the open-baffle 6dB/octave rolloff.[3] This approach produces a flat response and would allow us to avoid using the low-pass circuit and subwoofer resonant crossover. Such drivers are unavailable to consumers, however, and our approach has better low-frequency transient response ($Q_T = 2$).

POSITION OF POWER

The midranges and tweeters are configured in a symmetrical D'Appolito design. Their relative positions produce a subjectively pleasing sound which is difficult to achieve with a single midrange. This configuration works superbly for both open-baffle and conventional enclosures.

The D'Appolito design has a much better vertical radiation pattern than the single midrange, with a superior (vertically stable) lobing pattern near the crossover frequency. Lobing makes the speaker system sound differently off the primary axis, while reduced lobing results in lower phase aberrations in the 1–4kHz frequency region. The ear is most sensitive in this frequency region, so all forms of distortion need to be minimized.

Small distortions in the upper midrange (1–4kHz) contribute more substantially to a degradation of overall sound quality than do similar distortions in the deep bass or high treble. To reduce phase distortions, our design has no electrical components (such as crossover components) other than connecting wire between the midranges and the amplifier.

The placement of our subwoofer—only 4″ from the floor—eliminates the severe suckout notch that can result from floor bounce effects in the 100–200Hz region. This effect is a major problem with most enclosures whose drivers are more than 2′ above the floor.[4–6] Notches of 10dB are common, but this problem is nonexistent for our design.

THE TWEETERS

For excellent tweeters at a reasonable price, a good friend recommended the Audax HD13D34H. Its features include low resonance (900Hz), nominal 8Ω impedance, 92.5dB/W/M sensitivity, unusually large magnet, soft-dome design, exceptionally smooth frequency response, and subjectively pleasing sound quality.

The tweeters' 900Hz resonance allows them to be crossed over at the high-frequency limit of our midranges, which are 6dB down at 2.5kHz. We experimented with 20, 15, 10, 4.7, and 2.2µF (polypropylene) as a first-order crossover, with the tweeters wired in parallel. The subjective sound quality for the three largest values seemed a bit harsh on massed violins; the best overall sound was achieved with the 4.7µF value.

To create a partial zobel effect and moderate the severe tweeter impedance rise at reso-

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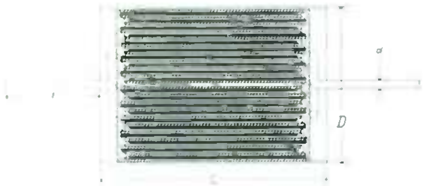

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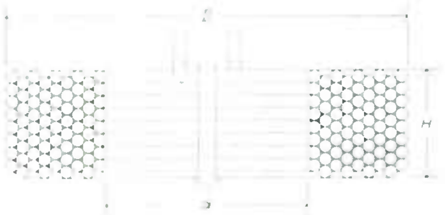

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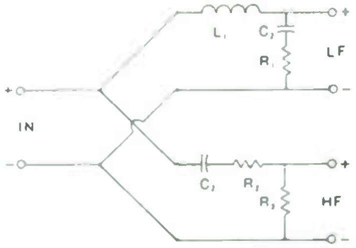

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nance, we placed a 1.2mH inductor directly across the tweeters. This produced a nearly constant load on the crossover down to 900Hz and successfully eliminated the resonance effects.

We chose the final crossover capacitor value by combining three methods. First, we strove for the flattest possible frequency response with the sound-level meter outdoors, using the warble tones to obtain as smooth a transition as possible at 2.5kHz. Second, with the baffles indoors, we again experimented with the capacitor values and the partial zobel. Finally, based on critical listening tests using three listeners, we settled on a 4.7μF capacitor with the 1.2mH zobel.

THE MIDRANGES

The midrange drivers should reproduce as wide a frequency range as possible with a very smooth response, as this maintains the purity of the critically important musical instrument fundamentals and their first few harmonics. We think the Madisound 1052DVC 10" driver best satisfies these criteria. It has excellent specifications, dual voice coils, polypropylene cones, and superb frequency response over the entire operating range from 25Hz–2.5kHz.

The manufacturer's values are: $f_O = 20Hz$, $Q_M = 3.68$, $Q_E = 0.28$, $Q_T = 0.26$, $R_E = 12.2\Omega$, $X_{MAX} = 6$ mm, sensitivity of 90dB/W/M, and power handling of 100W. We measured: $f_O = 25.2Hz$, $Q_M = 4.4$, $Q_E = 0.20$, $Q_T = 0.19$, $R_E = 11.4\Omega$, and sensitivity of 92dB/W/M. These represent averages for three drivers which were nearly identical in their measurements. We thought additional measurements were not needed for the remaining drivers.

In the midrange, polypropylene cones provide better sound quality than paper. With a higher stiffness-to-mass ratio, polypropylene enables better transient response where it is most important—across the extended midrange band. Cone material doesn't matter for subwoofers as long as it is rigid enough.

We wired the dual voice coils in the 16Ω configuration, with the two drivers wired in-parallel for a net midrange impedance of 8Ω. The net midrange acoustical output is equivalent to using one 8Ω midrange driver.

A midrange crossover is unnecessary, because the two 10" drivers are used over their full operating range. We measured their unequalized open-baffle acoustical output to be from 60Hz–2.5kHz at the 6dB down points, where the subwoofer and tweeters are brought on line.

MATCHING EFFICIENCIES

Our criteria for subwoofer selection included:

- High subwoofer efficiency to match the high midrange efficiency (and to avoid building a speaker that required high-cost amplifiers)
- A primary amplifier with only 60W/channel
- A 25Hz free-air resonance
- Long-throw drivers (subwoofers as well as midranges, because we must pump a lot of air at low frequencies to overcome acoustical self-cancellation)
- A driver Q_T near 0.5 (to provide optimum electromechanical transient response)
- Moderate power-handling capability

While no specific driver meets all these criteria, one comes close.

We selected the exceptional Polydax HD30P45TSM 12" driver as our subwoofer. It has the following manufacturers' values: $f_O = 25Hz$, $Q_M = 2.0$, $Q_E = 0.40$, $Q_T = 0.33$, $R_E = 5.2\Omega$, $X_{MAX} = 4.5$ mm, sensitivity of 95dB/W/M, and 90W power handling. Our measured values were: $f_O = 28Hz$, $Q_M = 2.75$, $Q_E = 0.17$, $Q_T = 0.16$, $R_E = 5.5\Omega$, and sensitivity of 98dB/W/M.

These values are averages for the two drivers; their individual measurements are quite similar. Our experience indicates that the measured resonance ($f_O = 28Hz$) is an initial value and will probably decrease several hertz after break-in. The same concept applies to the midranges.

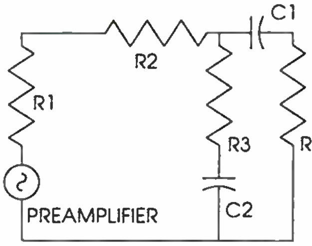

FIGURE 4: Passive low-pass circuit to partially correct for low-frequency self-cancellation.

Originally, we used a subwoofer which was identical to the midranges (both 10"). The bass was adequate but not dramatic—it failed the 1812 Overture cannon test. Since we like strong bass down to the audibility limit, we changed subwoofers.

The 12" Polydax driver produces extraordinary bass in our baffle. Even though we are pleased with it, we will probably use a 15" subwoofer in our next version and biamplify it. In that case, we can be less careful about matching the subwoofer and midrange efficiencies, because we will be able to control their levels independently.

RESONANT CROSSOVER

The subwoofer is brought on line only below 70Hz, where the 10" drivers' acoustical output has begun to roll off; it augments the midranges' deep bass output. Both the midranges and the subwoofer continue to operate down to 25Hz.

We did not use a traditional crossover; instead an innovative resonant crossover (Q = 1.6) drives the subwoofer and shapes its response. Due to the circuit resonance, our crossover provides 4dB of "free" electrical gain. It provides a steep high-frequency rolloff, and therefore also acts as a crossover.

The electrical values of the subwoofer resonant crossover are illustrated in *Fig. 2* (left). R1 is the connecting wire resistance plus the internal resistance of inductor L1, which combined equal 0.84Ω. L1 is 18mH. C1 is 330μF rated at 35V, although a higher voltage rating would provide some additional design margin for higher-powered amplifiers.

In our crossover calculations, we included the subwoofer's variable low-frequency impedance, as shown in *Fig. 2* (right). We did not assume the subwoofer to be purely resistive. The electrical values we used to model the subwoofer impedance near its resonance were: R2 (the DC resistance) is 5.2Ω; R3 is 28.3Ω. When combined at the subwoofer resonance, they become the total impedance at resonance of 33.5Ω. C2, the combined acoustical and electrical capacitance at resonance, is modeled as 1mF. L2 is the net acoustical and electrical inductance at resonance and is modeled as 40mH. The combined values of C2 and L2 must produce the subwoofer's 25Hz free-air resonance; they need not be physically precise to model the correct resonant value.

A constant resistance assumption would be seriously wrong for any subwoofer in the vicinity of its resonance. The variable subwoofer impedance actually improves the resonant crossover's performance, because it increases the resonant peak and slightly steepens the frequency drop. These desirable effects could not be obtained if the subwoofer had a constant resistance. The output of the resonant crossover driving the subwoofer impedance model is shown in *Fig. 3*.

TRANSITION FREQUENCY

Momentarily disregarding the effect of nearby walls, the idealized open baffle low-frequency response declines at an asymptotic rate of 6dB/octave.[7] The decline begins at the average baffle transition frequency (F_A), which is established by the baffle size and shape. Caused by rear- and frontwave cancellation, it can be worsened by close proximity to a back wall. Side walls don't have much effect.

The 6dB/octave decline continues to the driver resonance f_O. For an open baffle, this is nearly the same as the system resonance F_S. Below F_S the rolloff increases to 18dB/octave,

Continued on page 36

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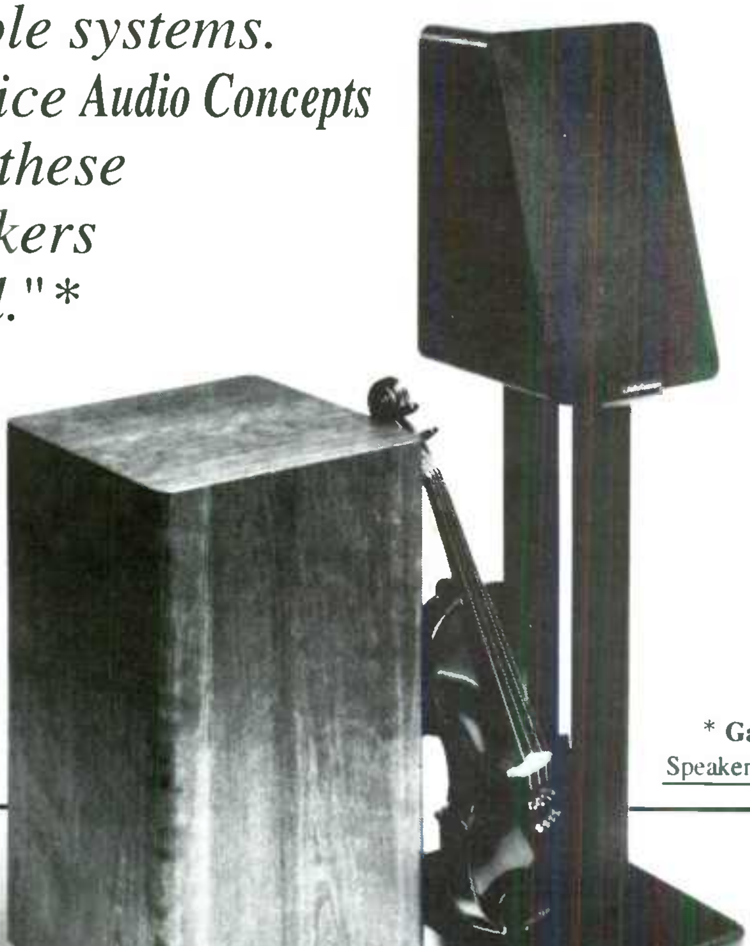

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ered, it can be as steep as 24dB/octave. This rolloff is the same as that exhibited by ducted-port (bass-reflex) enclosures below their low-frequency cutoff.

As with any speaker system, flat room response is obtained by using an additional parametric equalizer to selectively compensate for room effects. Open baffles are not unique in this regard, although ours requires less parametric equalization than many other systems due to its multidriver design.

As the average frequency below which cancellation begins, F_A is actually the same as the first reinforcement maximum. Reinforcement occurs when the wavelength of F_A equals twice the open baffle's mean front-to-rear acoustical pathlength P_A.[8]

To compute these pathlengths and frequencies, consider the speed of sound, which at standard temperature and pressure is 1,088′/second. Our baffle's minimum pathlength (P_M) is 2′, corresponding to an initial onset frequency (F_I) of 272Hz. While it is useful in an open-baffle design, F_I is not the most important factor. That distinction belongs to the average acoustical pathlength (P_A). Averaged over our tapered baffle for all three cone drivers, this is about 4′, which corresponds to an F_A of 136Hz.

Each driver's pathlength distribution can be measured by stretching a string from front to rear in uniform angular increments, such as 20°. Adding the individual lengths and dividing by their number results in P_A.

An alternate method of obtaining P_A is to take the square root of the baffle area.[9] In this calculation, use the front panel's full area. Divide the side panel area in half, since it is perpendicular to the front baffle and not as effective acoustically. This method also results in a P_A of 4′. F_I and F_A determine where the initial and average low-frequency rolloffs begin, not where they are 6dB down.

$$F_I = \frac{1{,}088}{2P_M} = \frac{544}{2} = 272\text{Hz}$$

$$F_A = \frac{544}{P_A} = \frac{544}{4} = 136\text{Hz}$$

Acoustical theory predicts that our 6dB down frequency F6 should be half of F_A.[10] The result is 68Hz, which is close to our measured value of 60Hz.

$$F6 = \frac{F_A}{2} = 68\text{Hz}$$

LOW-PASS CIRCUIT NAVIGATION

An important design aspect is shaping the three cone drivers' low-frequency response below 136Hz by placing a simple passive circuit (*Fig. 4*) between the preamp output and the main amplifier input. The circuit's frequency response is shown in *Fig. 5*. It has 2.8dB of insertion loss and 9.4dB of low-frequency boost between 15Hz, where it peaks, and 200Hz, where it flattens to within 1dB. We are calling it 9.4dB of low-frequency boost, although it is actually the equivalent amount of midfrequency attenuation.

The low-frequency rollover (3dB down point) is at 44Hz; the upper rollover (3dB up point) is at 100Hz. The 3.3dB/octave slope is less than desired, but the two rollover frequencies are so close that a full 6dB/octave slope cannot be achieved with this simple circuit. As you can see from the frequency response measurements in *Fig. 1*, however, the 3.3dB/octave slope is adequate. We don't need the full slope, because we bring in the subwoofer at the point where the unequalized midranges would begin to drop below 6dB. The subwoofer's additional contribution and the passive low-pass circuit's frequency shaping combine to produce a nearly flat low-frequency response.

The internal impedance characteristics of this circuit work well with our 1kΩ preamp output and 100kΩ amplifier input. These impedances are typical for most modern preamplifiers and amplifiers, so our circuit should work well in these systems, and is also suitable for use in a tape monitor loop. With a preamplifier having an output impedance of less than 1kΩ, the insertion loss would be slightly lower.

Due to the circuit's placement between the preamplifier output and the amplifier input, the resistors need be only ¼W and the capacitors rated at 10V. The latter should be Mylar® or polypropylene, not electrolytic. R1, which represents the preamplifier's internal resistance, is 1kΩ; R2 is 3.9kΩ; R3 is 1.5kΩ. R4 represents the amplifier's 100kΩ input impedance. C1 is 0.22μF; C2 is 1μF.

To provide sufficient isolation from induced hum, the circuit must be in a metal box. Our original version, which was in a plastic box, was sensitive to being near other electronic equipment. We used a passive rather than an active circuit because it is much easier to build and is distortion-free. In addition, it requires no power supply or integrated circuits.

Even though our passive low-pass circuit attenuates mid and high frequencies rather than boosts low ones, most systems do not need an active circuit. Provided sufficient power is available, passive equalization works because the frequency shape is the same despite a lower absolute amplitude. Even at extremely loud levels, volume controls are not usually turned up more than 30%. At least 15dB of unused gain capacity exists in the remaining volume-control rotation, and we use this excess to provide our boost. As a result, our volume control usually is turned up to about 60% rather than 30%. We never need to have the volume control at maximum—and we like our music quite loud.

DEEP BASS OUTPUT

One of our goals was to produce an open-baffle system with a frequency response extending down to 25Hz. We achieved this low-frequency output in several ways.

1. We evaluated midrange and subwoofer efficiencies before purchase, so as to have as efficient a subwoofer as possible compared to the midranges.
2. The drivers' free-air resonances are slightly above the lower limit of audibility.
3. To provide as much low-frequency augmentation as possible, we used the

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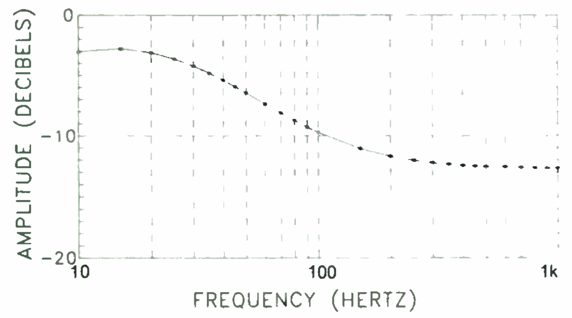

FIGURE 5: Passive low-pass circuit frequency response.

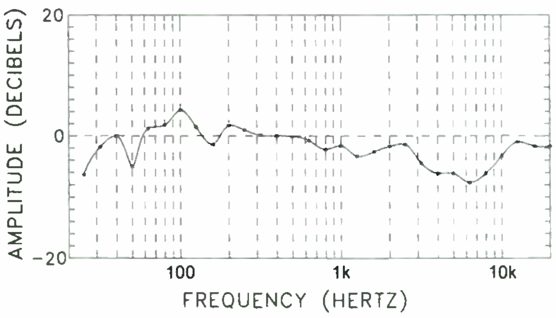

FIGURE 6: The system frequency response measured indoors at the listening location 10′ from both open baffles at tweeter height using ⅓ octave warble tones. No parametric equalization was used.

Continued from page 36

midranges over their full frequency range. Our measurements indicated they produce a nearly flat response from 2.5kHz–60Hz. Their response extends below 60Hz to 25Hz, where it augments the subwoofer output.

4. We used multiple drivers. Two 10″ drivers have an effective radiating area of 0.068 m^2, which is slightly larger than the 12″ subwoofer's 0.053 m^2. Operating the 10″ drivers below 60Hz is roughly equivalent to having a second 12″ subwoofer at that frequency. The phase shift induced in the subwoofer by the resonant crossover reduces this effectiveness slightly.

5. A resonant crossover (Q = 1.6) placed in series with the subwoofer provides 4dB of additional electrical boost at 30Hz (*Fig. 5*). Ignoring for a moment the acoustical self-cancellation effects and using the manufacturer's sensitivity results in a net difference between the subwoofer and the two midranges of:

95dB (woofer) – 90dB (midranges) + 4dB

= 9dB (net acoustical difference)

The 9dB excess in the subwoofer means that it is acoustically louder than the midranges. This net difference eases the power required from the amplifier at low frequencies and enables the system to perform with less low-frequency boost from the passive equalizer. Remember that the subwoofer begins to operate just below 70Hz, where significant self-cancellation is already occurring. This process balances the acoustical output of the subwoofer and the midranges.

INDOOR ACOUSTICAL MEASUREMENTS

To make all of our acoustical measurements, we used three Radio Shack sound-level meters simultaneously, mounted in a triangular pattern (*Photo 3*). Radio Shack's performance curves indicate that the meters do not have a flat frequency response, with increased sensitivity from 2–10kHz and decreased sensitivity above 10kHz. We made the necessary corrections to our results.

Figure 6 shows the frequency response measured 10′ from the drivers slightly above tweeter level at the listening location. Both speakers were in their standard placement 2′ from the back wall and were driven simultaneously. No parametric equalization was used.

By comparing *Figs. 1* and *6*, you can see the effects of room boundaries, which are dominated by the back wall. The most noticeable differences between the outdoor and indoor measurements are the –5dB notch at 50Hz and the room absorption dip between 4kHz and 9kHz. The notch corresponds to the listening room's 24′ depth.

As you might expect, the unequalized indoor frequency response is not as good as the outdoor measurements. Any speaker system suffers serious acoustic degradation when placed in a realistic environment. Nevertheless, our response is nearly flat—usually within 3dB—down to 25Hz when measured outdoors, and is reasonably flat below 3kHz indoors even without parametric equalization.

After the design work and construction were finished, and we knew the outdoor frequency response was excellent, we still had the degrading effects of room acoustics to deal with. We used the three sound-level meters in conjunction with a Sony ESD-1000 parametric equalizer to flatten the response at the listening location. *Figure 7* shows the results. Although the high frequencies measured relatively flat, they also sounded overly bright. By lowering the high-frequency equalization by 3dB, we achieved extraordinary subjective sound quality.

The Sony equalizer is very flexible. It allowed us to select any frequency and amplitude, and the Q of the shape. The specific settings listed in *Table 1* provided the measurements in *Fig. 7*.

In typical situations, the listening location is the only position that really counts. Near-field anechoic measurements are academically nice, but ordinary rooms are certainly not acoustically similar to anechoic chambers. Such completely absorbing chambers have virtually no wall, ceiling, or floor reflections.

Conventional near-field measurements made within a few inches of the drivers, or at the standard distance of one meter, are not meaningful for open baffles. They serve only to evaluate basic driver performance as a starting point for system design. For this reason, we have not included them in this article.

PHOTO 3: The sound-level meters used to measure frequency response. The average of three meters provides a more representative answer. Acoustical measurements at a single point can be very misleading, particularly when they are taken indoors.

Our frequency response measurements contain a 0.5dB meter reading error, caused primarily by difficulties in reading the warble tone sound source, plus the standard deviation of the RMS sum of the difference of each individual measurement from the three-meter average. These usually combined to be only about a decibel—a fairly small error—so our measurements should be reasonably accurate.

AMPLIFIER POWER

We used a medium-power, 60W/channel Adcom 535-II amplifier. Its overload indicator never flashed during any of our listening tests, even at very loud low-bass levels. This convinced us that even though our system was somewhat less efficient than other enclosures, it could still be driven to near-deafening levels by only 60W/channel.

To keep resistance to a practical minimum, we used 16-gauge, 0.004Ω/foot resistance wire between the amplifier and the speakers. A 12.5′ wire pair (25′ round trip) provided a 0.1Ω total resistance. This consideration is particularly important for an open-baffle system where the cone damping is primarily electrical.

While the impedance presented to the amplifier by the speakers is not constant, it should not be a difficult load. Higher-load impedances, and resistive rather than capacitive loads, are preferred. At high frequencies, the two tweeters, parallel inductor, and series capacitor combine to present a primarily resistive 4Ω load. Most amplifiers can easily drive 4Ω at high frequencies, where power requirements are usually modest. The first-order crossover increases the high-frequency impedance at 6dB/octave below the crossover frequency, so the tweeter load becomes about 50% capacitive. It exceeds 8Ω at that point but is in parallel with the midranges, which in this operating range are resistive and inductive. This combination should produce an easy amplifier load.

The midranges' parallel impedance is 8Ω at 100Hz, increasing to 25Ω at 20Hz and 2kHz. The subwoofer's actual impedance in its operating frequency range below 70Hz is always greater than 10Ω. Below 35Hz, the subwoofer resonant crossover imparts a partially capacitive character to the amplifier load, but it is offset by the 10–25Ω varying resistive and inductive load of the paralleled midranges.

Impedance-matching networks (zobels) are unnecessary for our application, because we use no conventional low- or midfrequency crossover networks which require performance stabilization. Impedance matching is necessary only to obtain proper performance from a crossover network by maintaining a constant resistance load. Since our midranges

Continued on page 40

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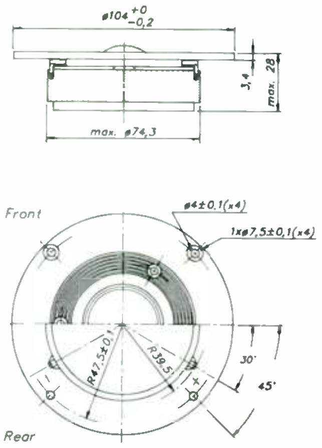

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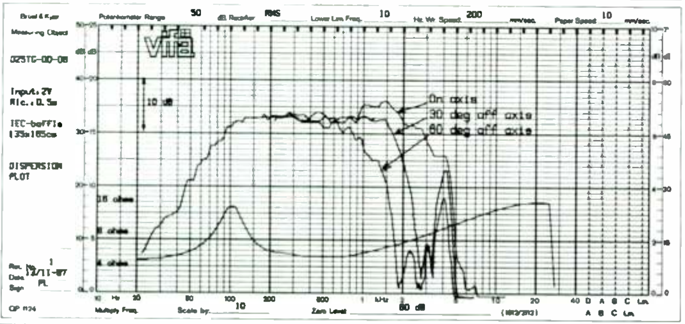

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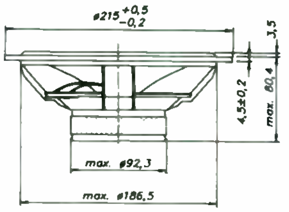

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VOICE COIL RESISTANCE	5,8	Ω
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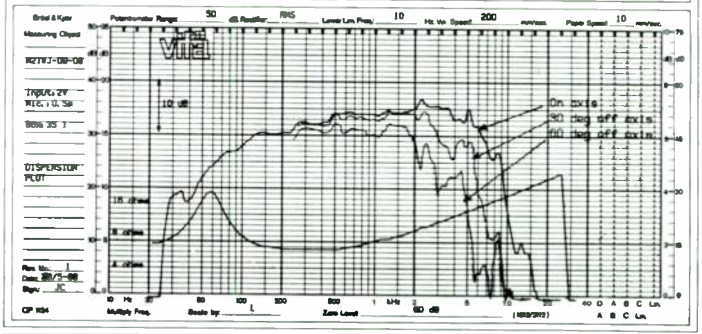
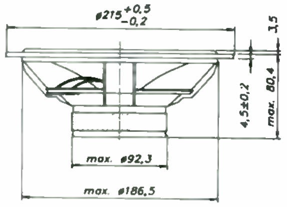
Vl			5.1"	3.7"



8" WOOFER M21WJ-49-08

SPECIAL FEATURES:

- MAGNESIUM BASKET
- EXCELLENT LARGE SIGNAL BEHAVIOUR
- 4–LAYER VOICE COIL
- IDEAL FOR BASS REFLEX
- BLACK COATED PAPER CONE
- HIGH DAMPING RUBBER SURROUND

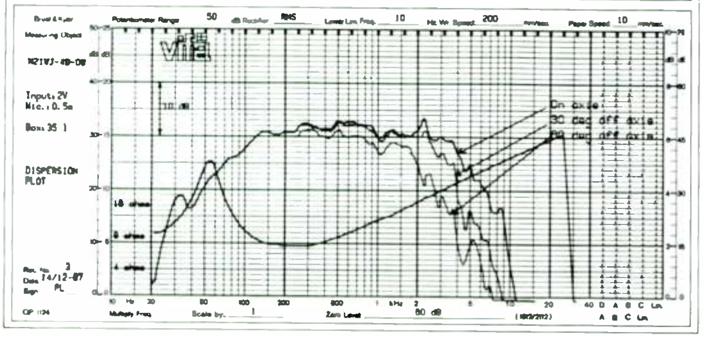


NOMINAL IMPEDANCE	8	Ω
NOMINAL POWER (IEC 268–5)	80	W
MUSIC POWER (DIN 45 500)	120	W
FREQUENCY RANGE	25–3000	Hz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE CONE AREA	239	cm²
VOICE COIL RESISTANCE	5,8	Ω
VOICE COIL INDUCTANCE	1,7	mH
OPERATING POWER	4	W
VOICE COIL DIAMETER	32	mm
VOICE COIL HEIGHT	13	mm
AIR GAP HEIGHT	6	mm
FREE AIR RESONANCE	25	Hz
MOVING MASS (incl. air)	23,5	g
FORCE FACTOR, B x l	8,9	Txm
MAGNET WEIGHT (14,6 oz)	415	g
Qms	2,2	
Qes	0,27	
Qts	0,24	
Vas	140	ltr

Price Each $20.00

M21WJ-49-08

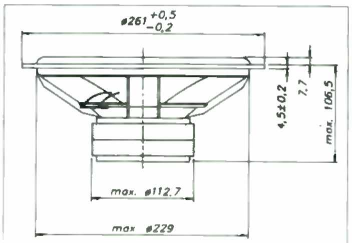
Vb Liter	14	20	28	34
Fb Hz		41	41	43
F3 Hz	75	58	50	46
Qtc	8			
Ql	Sealed	7	7	7
Vd		2 0"	2 0"	2 0"
Vl		5 8"	3 8"	2 4"



10" WOOFER M25W0-35-06

SPECIAL FEATURES:

- MAGNESIUM BASKET/COATED PAPER CONE
- DOUBLE MAGNET SYSTEM
- LONG THROW/LOW DISTORTION MOTOR
- SMOOTH FREQUENCY RESPONSE
- IDEAL FOR THE ULTIMATE EXTENDED BASS REFLEX BOX

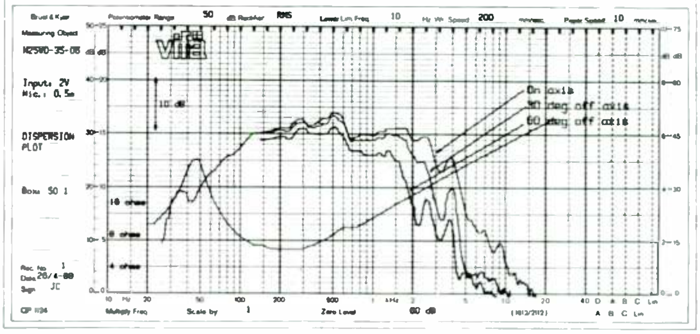


NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268–5)	100	W
MUSIC POWER (DIN 45 500)	130	W
FREQUENCY RANGE	24–1500	Hz
SENSITIVITY (1W, 1m)	90	dB
EFFECTIVE CONE AREA	346	cm²
VOICE COIL RESISTANCE	5,1	Ω
VOICE COIL INDUCTANCE	2,3	mH
OPERATING POWER	4	W
VOICE COIL DIAMETER	40	mm
VOICE COIL HEIGHT	20	mm
AIR GAP HEIGHT	6	mm
FREE AIR RESONANCE	24	Hz
MOVING MASS (incl. air)	51	g
FORCE FACTOR, B x l	12,3	Txm
MAGNET WEIGHT (49,2 oz)	1396	g
Qms	3,9	
Qes	0,27	
Qts	0,25	
Vas	141	ltr

Price Each $36.00

M25W0-35-06

Vb Liter	21	40	35	31
Fb Hz		39	39	38
F3 Hz	67	42	44	46
Qtc	7			
Ql	Sealed	7	7	7
Vd		3 0"	3 0"	2 5"
Vl		6 7"	8 1"	6 8"



Reader Service #1

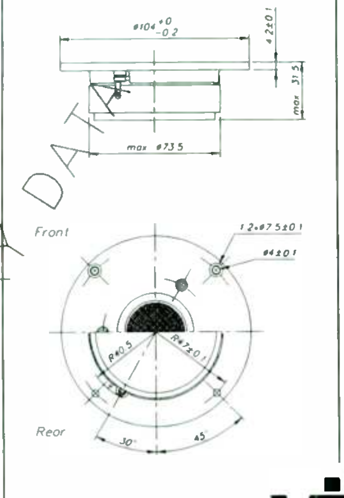
Vifa Madisound Vifa Madisound Vifa Madisound Vifa

1" DOME TWEETER D27TG-05-06

Features
- * SILK & FABRIC DIAPHRAGM WITH A SPECIAL COATING.
- * LINEAR RESPONSE FRONTPLATE.
- * LOW DAMPING FERROFLUID W. HIGH COLLOID STABILITY.
- * DAMPED CAVITY IN POLEPIECE.
- * REPLACEABLE VOICE COIL.

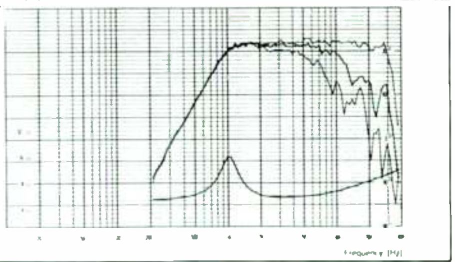
Nominal Impedance [ohm]	6
Nominal Power (IEC 268-5) [W]	100
Frequency Range [kHz]	2.5-30
Sensitivity (1W,1m)/(2,83V,1m) [dB]	91/92
Effective diaphragm Area [sq.cm]	7.1
Voice Coil Resistance [ohm]	4.60
Operating Power [W]	3.2
Voice Coil Diameter [mm]	26
Voice Coil Height [mm]	1.6
Air Gap Height [mm]	2
Moving Mass (incl. Air) [g]	0.29
Free Air Resonance [Hz]	1000
Force Factor, B*l [T*m]	2.6
Magnet Weight [mm]	240
Total Q-value [Qt]	0.96

Price Each $15.00



Neutral front plate

These new tweeters from Vifa are in keeping with the engineering excellence and competitive pricing that has made Vifa a leader in the audio industry. The tweeters have very flat frequency responses and fast rise times. These improvements have come from redesigned face plates with "tube" loading instead of "horn" loading. This difference results in a flat response and smooth roll off, both on and off axis. The new dome materials help damping and virtually eliminate ringing.

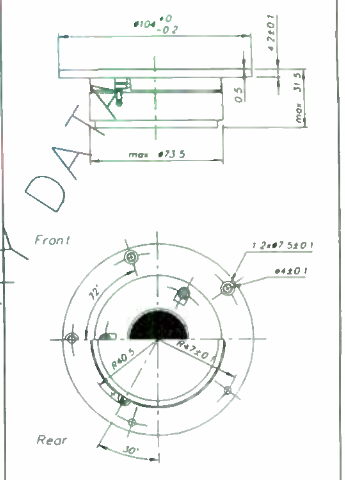


1" DOME TWEETER D27TG-15-06

Features
- * SILK & FABRIC DIAPHRAGM WITH A SPECIAL COATING.
- * LINEAR RESPONSE FRONTPLATE WITH IMPROVED DISPERSION.
- * LOW DAMPING FERROFLUID W. HIGH COLLOID STABILITY.
- * DAMPED CAVITY IN POLEPIECE.
- * REPLACEABLE VOICE COIL.

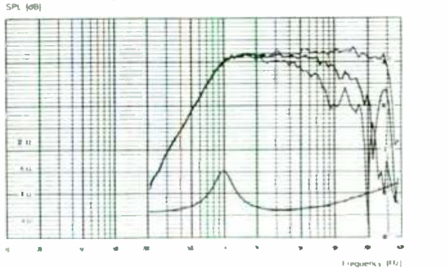
Nominal Impedance [ohm]	6
Nominal Power (IEC 268-5) [W]	100
Frequency Range [kHz]	2.5-30
Sensitivity (1W,1m)/(2,83V,1m) [dB]	91/92
Effective diaphragm Area [sq.cm]	7.1
Voice Coil Resistance [ohm]	4.60
Operating Power [W]	3.2
Voice Coil Diameter [mm]	26
Voice Coil Height [mm]	1.6
Air Gap Height [mm]	2
Moving Mass (incl. Air) [g]	0.29
Free Air Resonance [Hz]	1000
Force Factor, B*l [T*m]	2.6
Magnet Weight [mm]	240
Total Q-value [Qt]	0.96

Price Each $15.00



Bulged front plate

The D27TG-05 and -35 tweeters have a flat flange with an exposed silk dome. The D27TG-15 and -45 silk domes utilize a very small bell or bulge to help focus the sound toward the front. The -35 and -45 units use a rear chamber to lower resonance. We recommend the -35 and -45 for use in two way systems. Vifa has described the sound difference between the flanges, suggesting the flat flange as being neutral, and the bulged flange as reproducing aural perspective with accuracy.

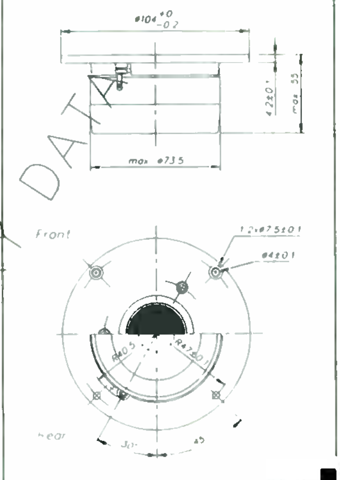


1" DOME TWEETER D27TG-35-06

Features:
- * SILK & FABRIC DIAPHRAGM WITH A SPECIAL COATING.
- * LINEAR RESPONSE FRONTPLATE
- * LOW DAMPING FERROFLUID W. HIGH COLLOID STABILITY.
- * BRAIDS AND REAR CHAMBER.
- * REPLACEABLE VOICE COIL.

Nominal Impedance [ohm]	6
Nominal Power (IEC 268-5) [W]	100
Frequency Range [kHz]	1.5-30
Sensitivity (1W,1m)/(2,83V,1m) [dB]	90/91
Effective diaphragm Area [sq.cm]	7.1
Voice Coil Resistance [ohm]	4.60
Operating Power [W]	3.9
Voice Coil Diameter [mm]	26
Voice Coil Height [mm]	1.6
Air Gap Height [mm]	2
Moving Mass (incl. Air) [g]	0.29
Free Air Resonance [Hz]	700
Force Factor, B*l [T*m]	2.5
Magnet Weight [g]	240
Total Q-value [Qt]	0.38

Price Each $19.50



Neutral front plate

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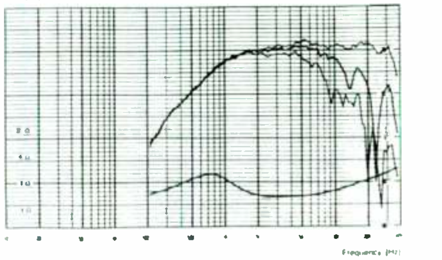
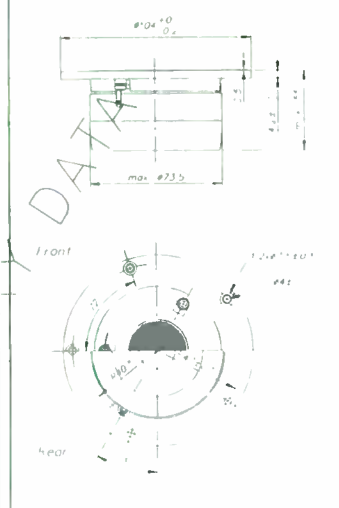


1" DOME TWEETER D27TG-45-06

Features:
- * SILK & FABRIC DIAPHRAGM WITH A SPECIAL COATING.
- * LINEAR RESPONSE FRONTPLATE WITH IMPROVED DISPERSION.
- * LOW DAMPING FERROFLUID W. HIGH COLLOID STABILITY.
- * BRAIDS AND REAR CHAMBER.
- * REPLACEABLE VOICE COIL.

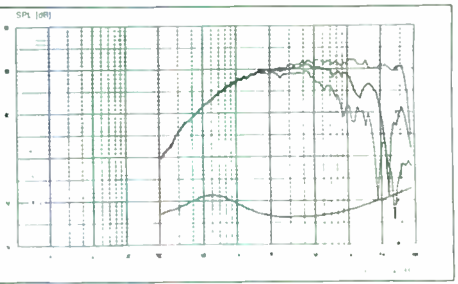
Nominal Impedance [ohm]	6
Nominal Power (IEC 268-5) [W]	100
Frequency Range [kHz]	1.5-30
Sensitivity (1W,1m)/(2,83V,1m) [dB]	90/91
Effective diaphragm Area [sq.cm]	7.1
Voice Coil Resistance [ohm]	4.60
Operating Power [W]	3.9
Voice Coil Diameter [mm]	26
Voice Coil Height [mm]	1.6
Air Gap Height [mm]	2
Moving Mass (incl. Air) [g]	0.29
Free Air Resonance [Hz]	700
Force Factor, B*l [T*m]	2.5
Magnet Weight [g]	240
Total Q-value [Qt]	0.38

Price Each $19.50



Bulged front plate

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



Continued from page 38

have no crossovers, they don't need impedance matching. Conventional first-order crossovers have been carefully matched with a partial zobel to the tweeters' varying impedance in the vicinity of their resonant frequency. The woofer uses a resonant crossover with a different performance and function from that of a conventional crossover. The varying subwoofer load on this network is desirable and aids rather than hinders its performance.

A multidriver speaker system does not need a constant impedance load in order to work well. The additional complexity required to achieve it would degrade other parameters (i.e., raise costs, increase complexity, and induce unneeded phase shifts). The most important aspect of impedance is that it be thoroughly considered in the design; it need not be constant.

TRANSIENT RESPONSES

The overall system Q_S determines the transient response. Quite unlike conventional enclosures designed using T/S theory, our open-baffle Q_S is found from the reciprocal sum of the parallel Q values plus a straightforward addition of the series Q values. While it would be preferable to have $Q_S = 0.5$ for the subwoofer, midranges, and tweeters in order to achieve optimum transient response, we did not quite achieve this goal.

An underdamped cone ($Q_S > 0.5$) returns to its neutral position and then overshoots. It oscillates a few times about the neutral position before stopping. In this condition, the cone is inadequately controlled, hence underdamped. Neither modest underdamping nor modest overdamping is audible; however, either extreme is quite audible. In our opinion, if critical damping cannot be achieved, then the latter is preferable because the mushy sound resulting from uncontrolled cone motion is particularly unsatisfactory.

An overdamped system ($Q_S < 0.5$) provides no benefits. The cones do not return to their neutral position rapidly enough, but at least there is no oscillatory motion.

The subwoofer Q is obtained from the free-air driver parameters provided by the manufacturer or measured by the speaker builder. These parameters are modified by external wire resistances and combined with the Q of the low-pass resonant network that is in series with the subwoofer.

The low-pass network reactance (R_I) was 0.74Ω at resonance, and was due to the series inductor's DC resistance. A minor contribution from the connecting wire (R_W) was 0.1Ω. The amplifier internal resistance (R_G) was a negligible 0.05W. The low-pass resonant crossover had Q of 1.6. The specific calculation for the subwoofer transient response using the manufacturer's values is:

$$1/Q_{SUBWOOFER} \approx 1/Q_M + R_E/[Q_E(R_G + R_I + R_W + R_E)]$$

$$\approx ½ + 5.2/[0.40(2.0 + 0.10 + 5.2)]$$

$$= 0.50 + 1.71 = 2.21$$

$$Q_{SUBWOOFER} \approx 0.45$$

The effect of the series resonant crossover must be added to that of the subwoofer:

$$Q_{SUBWOOFER\ SYSTEM} =$$

$$Q_{SUBWOOFER} + Q_{RESONANT\ CIRCUIT}$$

$$= 0.45 + 1.6 = 2 \text{ (approximately)}$$

When an electrical circuit drives an electromechanical device in series with it (as opposed to paralleled), the Q values are added. Our subwoofer Q of 0.45 is close to the theoretically ideal value of 0.5 for critical damping of the cone motion, so we have excellent electromechanical performance. When combined with the resonant crossover's Q of 1.6, it increases to about 2: our electrical/acoustical transient response is underdamped. This transient response is much better than that of most ducted-port, but not as good as most sealed-box, enclosures. Subjectively, we hear tight, superb deep bass.

The midrange Q is easy to determine, because there is no circuitry between the midranges and the amplifier other than wire. As before, R_G is negligible.

$$1/Q_{MIDRANGE} = 1/Q_M + R_E/[Q_E(R_G + R_W + R_E)]$$

$$= 1/3.68 + 12.2/[0.28(0.10 + 12.2)]$$

$$= 0.271 + 3.54 = 3.81$$

$$Q_{MIDRANGE} = 0.26$$

Our midrange transient response is somewhat overdamped, so we attempted to move the Q closer to the ideal value of 0.5 in order to achieve exact critical damping. We inserted a series resistor in the wire path to the amplifier, which both lowered the midrange amplitude relative to the subwoofer by a few decibels and increased the Q. These results should have been beneficial but were not.

We inserted a 5.6Ω resistor (for $Q_{MIDRANGE} = 0.37$), and then a 3.3Ω resistor (for $Q_{MIDRANGE} = 0.33$). Three of us listened carefully to the results, and we all heard a slight "veiling" of the details and nuances in the sound in each case. While the results are puzzling, we trusted our ears and removed the resistors. Perhaps, for some reason, slight overdamping is subjectively more pleasing at midrange frequencies.

The tweeter Q should be given by the manufacturer's values, as modified by the wire resistance R_W. Unfortunately, the manufacturer does not provide the information.

$$1/Q_{TWEETER} = 1/Q_M + R_E/[R_G + R_W + R_E)]$$

At lower frequencies, the calculation would be slightly different because it is affected by the capacitor's series reactance R_C. At the tweeter-crossover frequency, R_C and R_E are nearly equal if R_G and R_W are small. The equation would be:

$$1/Q_{TWEETER} = 1/Q_M + R_E/[Q_E(R_G + R_W + R_E + R_C)]$$

The open-baffle frequency response is not related to the system Q_S in the same manner as in conventional enclosures. We obtained excellent frequency response by varying the baffle shape, using multiple drivers, applying the D'Appolito configuration, balancing the subwoofer and midrange acoustical efficiencies, using the subwoofer resonant crossover, incorporating the passive low-pass circuit,

TABLE 1

PARAMETRIC EQUALIZER SETTINGS*

FREQUENCY (HZ)	Q-VALUE	GAIN
26	3.7	+8.0
91	3.7	−3.5
10k	0.7	+6.0

*Used for *Fig. 7*.

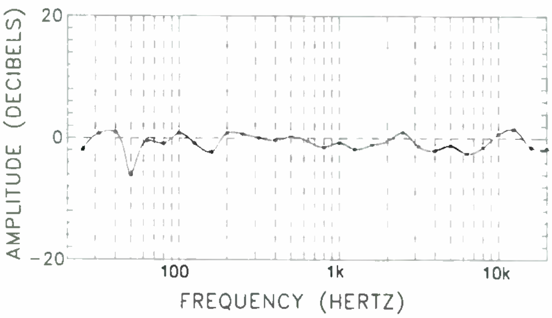


FIGURE 7: The same as *Fig. 6*, except that the parametric equalizer was used to compensate for the room boundaries and smooth the frequency response.

and placing the baffle at least a foot from the back wall.

To achieve good transient response, strive for a system Q_S that is as close as possible to 0.5. The system Q_S is a combination of the driver mechanical Q_M and electrical Q_E, as modified by the presence of series resistances and other Q values in the system.

TWO-WAY DESIGN

Despite the fact that each baffle uses five drivers—one subwoofer, two midranges, and two tweeters—our design is essentially a full-range two-way plus a subwoofer. This configuration produces a balanced dipole sound pattern at all frequencies. One important aspect of good dipole design is to maintain a full-range frequency response in both the rear and front directions.

- The two 10″ drivers act as a single driver acoustically.
- The rear tweeter is out-of-phase with the front one. Except for their physical separation, they radiate as if they were a single dipole tweeter.
- The 12″ subwoofer near the floor is also a dipole. It operates only below 70Hz, where the wavelengths are very long.

MIDRANGE DRIVER SIZE

Midrange size is more flexible with the open-baffle design than with conventional enclosures. In single-driver designs, the diameter must be small enough to preclude severe forward beaming (narrowing radiation pattern) at high frequencies. A larger midrange must be crossed over to the tweeter at lower frequencies. In the crossover region, a single 10″ driver would not ordinarily blend well with the tweeter, which is radiating broadly over nearly the full frontal hemisphere.

Our design's narrower forward radiation pattern improves the imaging performance. In addition, after the rear-firing radiation reflects from the back wall, it substantially broadens the soundfield throughout the listening room. The speakers sound transparent and have a very wide sweet spot, due to minimized lateral wall reflections. Slight midrange beaming near the tweeter crossover frequency is thus less important than in conventional designs, so we can do quite nicely with 10″ midrange drivers.

An enclosed cabinet with a single 10″ driver and a tweeter would be a poor choice, because the front radiation pattern near 2.5kHz becomes vertically unstable (i.e., moves up and down with frequency). A higher-order tweeter crossover could reduce these effects but would not improve the midrange's angular beaming. Our open baffle combined with a D'Appolito configuration and a subwoofer near the floor virtually eliminates these problems.

ACKNOWLEDGMENTS

The authors would like to express their appreciation to Diane Janni for providing outstanding computer expertise. We also appreciate the excellent advice on tweeter selection provided by Darrell Spreen.

REFERENCES

1. Salmi, Jorma, "Dipole Source Placement in a Room," AES Preprint 3327, 3/92 (92nd Vienna Convention).

2. Olson, Harry, *Acoustical Engineering*, Philadelphia: Professional Audio Journals Inc. (1991): 155.

3. Muxlow, Peter, "Subwoofers Part II," *SB* (1/89): 52.

4. Moran, David, "We Get Mail," *SB* (4/92): 80.

5. Moran, David, "Baffles with Bowers & Wilkins," *SB* (5/92): 74.

6. Allison, Roy, "The Influence of Room Boundaries on Loudspeaker Output," 2nd Ed. *Audio Engineering Society Anthology* (10/80): 353.

7. Olson, *Acoustical Engineering*, p. 152.

8. Woram, John, *Sound Recording Handbook*, Indianapolis: Howard W. Sams & Co. (1989): 173.

9. Lin, James, "A Dipole Subwoofer for Quads," *SB* (6/89): 10.

10. Linkwitz, Siegfried, "Development of a Compact Dipole Loudspeaker," AES Preprint 3431, 10/92 (93rd San Francisco Convention).

To summarize, the D'Appolito configuration minimizes the development of angular lobes in the vertical plane and stabilizes the horizontal sound direction as a function of frequency. The dipole design also produces a radiation dispersion pattern in the horizontal plane that is narrower at lower frequencies than for conventional designs, producing a better angular balance with frequency and better imaging. In addition, the narrow angular width in the horizontal plane is moderated by the dispersion from the back wall.

BAFFLE CONSTRUCTION

For the prototypes, we used medium-density particleboard. It is much easier to work with and more structurally solid than the low-density variety, but is priced reasonably enough for this purpose.

We constructed the final version from solid red oak. The price of this hardwood is not particularly prohibitive, and working with solid wood offers construction and aesthetic advantages. Besides, the first two homes to receive "tombstones" were furnished with natural finished oak.

The edges of the actual baffle or front plane are mitered or beveled at a 45° angle to meet the mitered front edges of the side and top panels. This joint will receive a nice round-over (3/8" radius) and be virtually invisible once stained and lacquered. Biscuit joinery and common gluing techniques are used for assembly.

The drivers are mounted flush on the baffle with a rabbet or recess receiving the lip around the driver frame to reduce diffraction effects. High frequencies traveling over an abrupt edge are modified by diffraction in undesirable ways while rounded edges reduce acoustic diffraction effects, particularly at the higher frequencies where such effects are more pronounced.

DESIGN SUMMARY

We selected a subwoofer that was 5dB more efficient than the midranges and mounted it near the floor. Using a low-pass resonant crossover (Q = 1.6) having a boost of 4dB increased its output even more. A simple passive circuit corrected most of the self-cancellation rolloff. Multiple drivers mounted on a variable-shape baffle minimized lobing and prevented midrange suckouts. This approach also has the advantage of minimizing side wall effects.

An open baffle is inefficient at low frequencies because of the self- cancellation that occurs when the acoustical path length from the rear to the front of the baffle exceeds a wavelength. As we have shown, this cancellation does not prevent excellent bass performance. We achieved a smooth frequency response that extends down to 25Hz, and the transient response is excellent. Moreover, construction is simpler than for any other type of enclosure. The optimum listener location is not particularly sensitive to room location. The astonishing subjective sound quality and extraordinary performance are achieved with simple components that are reasonably priced and easily obtained. ➥

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AN EVOLVING MAGNEPAN MG-1

By Mark D. Seymour

One of Jim Winey's noteworthy contributions to audio is a relatively low-distortion planar speaker, which generates sound waves by employing magnetic (versus electrostatic) attraction and repulsion between a fixed air-permeable grid and a vibrating planar diaphragm. This simple speaker design has enabled a glimpse of midrange detail and imaging available only in exotic, less affordable designs.

I have owned a pair of MG-1s—a two-way design popular in the '70s—for 12 years. While there have been many advances in the art of speaker design since then, including a number by Winey himself, I remain attached to my "Maggies."

In order that these old friends conform to my evolving estimation of "good sound," I've had to rejuvenate them. I know a number of you are Magneplanar® speaker fans, and, since used MG-1s are showing up for sale in the classifieds, I thought I would share some tricks aimed to keep them close to the state-of-the-art. While many techniques for upgrading speaker performance are available, I'll stick to those modifications which worked for the MG-1s per se. Even so, many of these improvements are applicable to both electrostatic and cone-driver designs.

I have successfully improved detail and imaging, frequency response, and bass solidity. Moreover, you can make modifications in stages of increasing complexity and each will yield marked improvement.

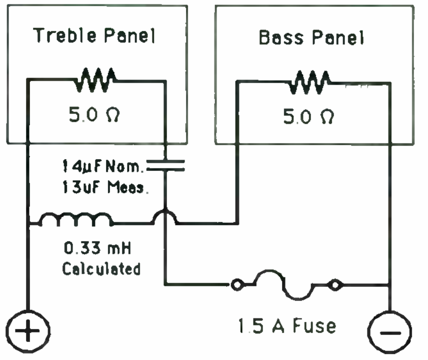

FIGURE 1: Schematic of MG-1 as purchased.

DETAIL AND IMAGING

Upgrading "passive" circuit components to achieve sonic improvement has long been a pursuit of J. Peter Moncrieff, audiophile and publisher. His advocacy of Wonder Caps®, Wonder Wire™TM, and Wonder Solder™TM has resulted in some far-reaching changes in the construction and performance of high-end audio.$^{1, 2}$ Amateurs have "recapped" amps, preamps, and crossovers to literally transform their systems' sonic clarity and neutrality.

I realized the best value—fidelity and dollar—with my Magneplanars by replacing the stock crossover capacitors with Ultra Kap™ high-quality film devices. My speakers came with bipolar-polar electrolytic capacitors labeled 14µF; they actually measured 12.8µF and 13.1µF, respectively (*Fig. 1*). These crossover caps are attached with a gob of adhesive to the metal panel just beneath the back grille cloth near the input terminals. To replace them, you must disassemble the speaker frame and carefully remove the cloth

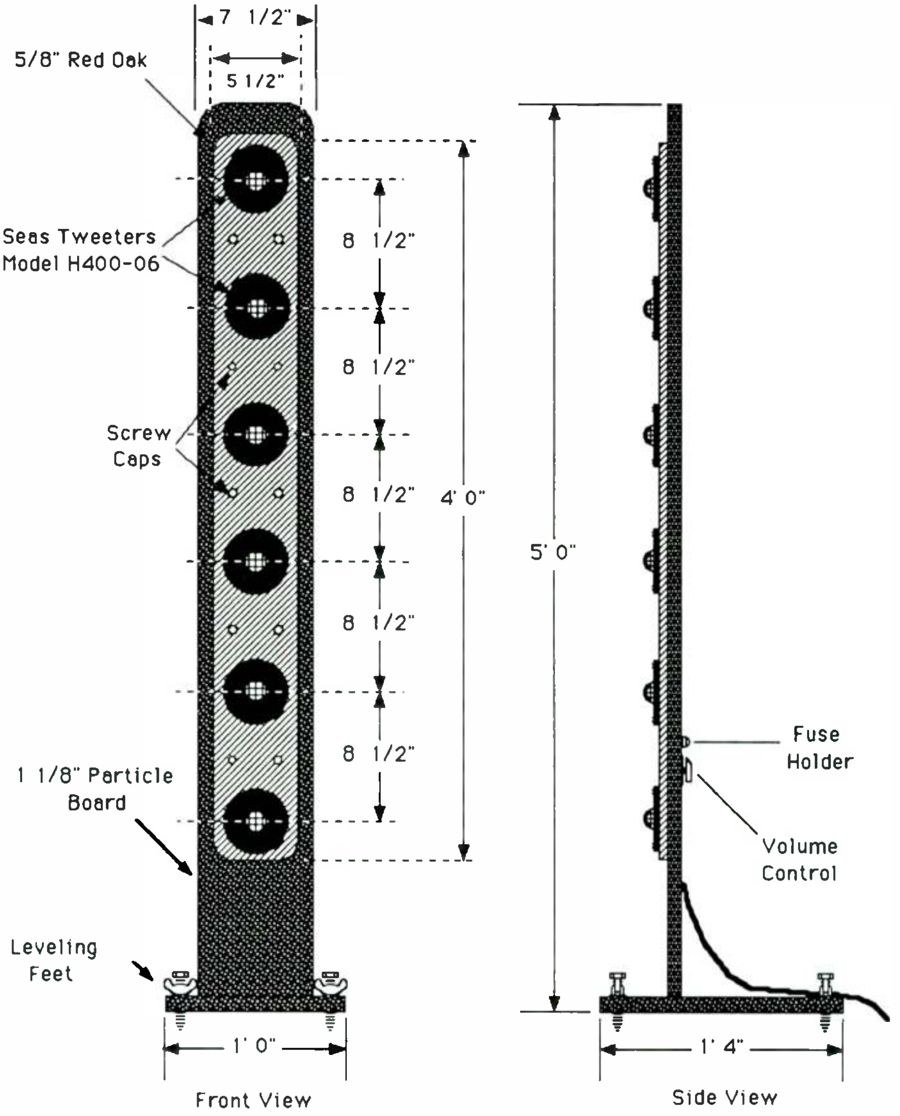

FIGURE 2: Tweeter tower design.

ABOUT THE AUTHOR

Mark Seymour has experimented with things electrical and mechanical since age four. He received his BS in chemistry from LeMoyne College and a PhD from Iowa State. Mark has built or modified numerous audio components and has just finished his third set of speakers and a neutral preamp of his own design.

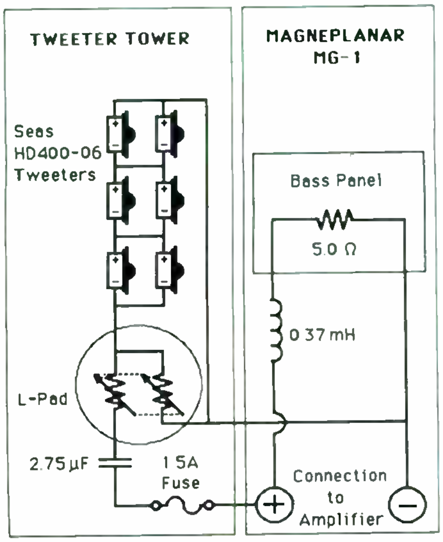

FIGURE 3: Wiring schematic with outboard tweeter array.

sleeve that surrounds the panel, front and back. Be careful not to damage either the internal diaphragm or the supporting metal frame. (Two sets of hands are better than one for this operation.) Unsolder, pry loose, and replace each stock capacitor with 6µF and 8µF (5.4µF and 7.1µF, measured) Ultra Kaps wired in parallel.

To further reduce veiling, remove the grille cloth; these items are notorious for sucking up high-frequency information, and for generally fuzzing up the performance of otherwise good speaker designs. The MG-1 is no exception.

To improve the appearance of the bare panels, first remove all the oak trim from the speaker. Using a virtually dry sponge brush (i.e., most of the paint squeezed out), carefully apply good-quality flat-black paint to the front panel. Include the perforated plate that supports the stationary magnets, but do not let paint run into the holes or reach the diaphragm. Finally, replace the trim, side and top rails, and base.

A word of caution: MG-1s are sensitive to sunlight, which degrades the adhesive used on the diaphragm. If your speakers back up to glass, be it window or door, do not remove the grille cloth.

FREQUENCY RESPONSE

The original frequency specification for my MG-1s was 50Hz–16kHz 4dB. With time, however, they became dull sounding: the sense of air and crispness associated with good treble response simply faded away. Subsequent Magnepan models offer extended and more durable high-frequency response by means of a ribbon tweeter rather than the film-and-wire diaphragm used in earlier, less expensive models.

With a little work and about $350 in materials, I was able to obtain markedly improved high-frequency performance. The basic plan calls for tweeter "towers," made by disconnecting the existing tweeter panels and replacing them with separate, free-standing units supporting a vertical array of high-quality dome tweeters (*Fig. 2*).

I decided on an array versus a single-tweeter design for several reasons. The first was to maintain a "line" (as opposed to a "point") sound source to match the remaining bass panel's radiation pattern consistent with Winey's original design philosophy. Multiple tweeters also enable power distribution minimizing distortion. In addition, a relatively rugged design would easily handle the transients reproduced by state-of-the-art audio source/amplifier combinations.

Three paralleled pairs of tweeters are wired in series, as illustrated in *Fig. 3*. The effective impedance of each tweeter in the frequency range of interest is 6Ω; therefore, the tweeter network's effective impedance is 9Ω (*Fig. 4*).

The crossover design is a combination of trial and computer simulation. For simplicity, I chose to use the original 6dB/octave design and the original choke, which is firmly affixed to the panel frame. The challenge was then to select the crossover point and component values which would yield a flat frequency response without unworkable phase anomalies.

I determined the choke inductance by calculation and experimentation. Given the MG-1's original specified crossover frequency (2.4kHz) and the impedance of the panels (5Π), I calculated the optimum capacitance and inductance to be 13.3µF (13µF measured) and 0.33mH, respectively.[3]

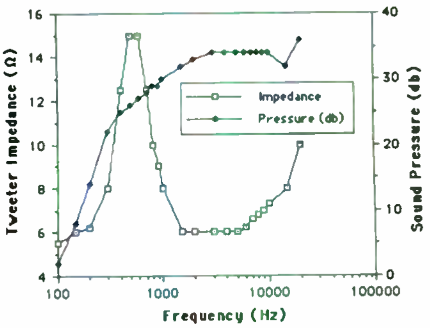

FIGURE 4: Seas H400 tweeter characteristics.

In-circuit measurement of the inductance resulted in an average value of 0.36mH. In the crossover, the bass panel (said to represent 5Ω "purely resistive impedance") is series-connected to the choke (*Fig. 1*). The power amp impresses a voltage across this series circuit. Since

$$E_R = I \times R \text{ and } E_L = I \times X_L$$

where:

E_R = Voltage drop across the resistor (bass panel)
E_L = Voltage drop across the choke
I = Current
R = Resistance of the bass panel
X_L = Reactance of the inductor

and

$$X_L = 2 \times \Pi \times F \times L$$

where:

F = Frequency in hertz
L = Inductance in henries

therefore, combining and rearranging, we obtain:

$$(E_L \times R)(E_R \times 2 \times \Pi \times F)$$

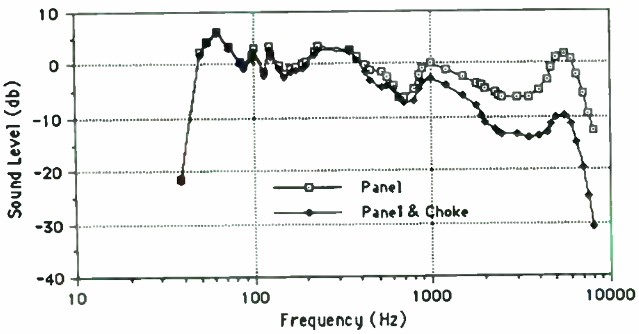

FIGURE 5: Bass panel response at 10 cm.

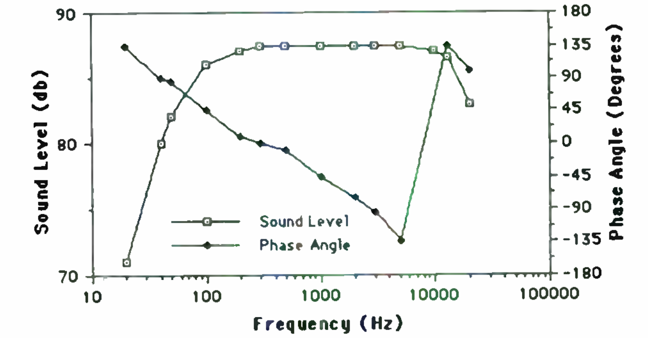

FIGURE 6: Response predicted by LMP software.

TABLE 1

INPUT VALUES FOR LMP PROGRAM

	TWEETER ARRAY	BASS PANEL
Low-frequency corner	1kHz	50Hz
High-frequency corner	20kHz	6.5kHz
High-frequency rolloff order	2	2
Sensitivity	90dB	88dB
Low-frequency damping ratio	1	1
High-frequency damping ratio	1	1
Polarity inversion	Yes	No
Depth offset*	0"	0"
Step frequency	None	None
Step height	0dB	0dB
Tweeter (network) impedance	9.5Ω	5.0Ω
Crossover	6dB/octave	6dB/octave
Crossover capacitance	2.57μF	
Crossover inductance		0.36mH

*If the distance from each driver to the listener is the same (depth offset = 0)

Using this equation, an AC voltmeter, and a function generator to drive the amplifier, I obtained values of 0.30mH at 10.0kHz, 0.36mH at 5.00kHz, 0.39mH at 1.00kHz, and 0.40mH at 500Hz.

I next measured the frequency response of the bass panel, using a Beyer dynamic M101N(C) microphone placed 10 cm away. The short distance diminishes secondary reflections and increases the signal/noise level for more accurate measurements. The output was amplified, digitally smoothed, and corrected for microphone sensitivity as a function of frequency. *Figure 5* shows the bass response both with and without the choke in series, which ranges within 7dB from 50Hz–6.5kHz. Much of the fluctuation is undoubtedly due to room effects. (An anechoic chamber and a warble-tone generator would have produced a smoother response curve.)

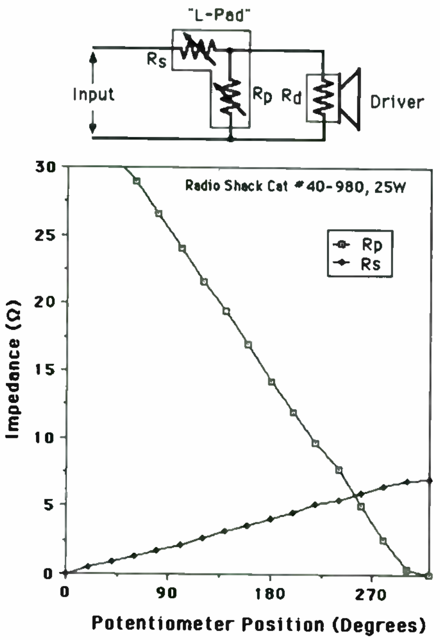

FIGURE 7: L-Pad tweeter attenuator.

My choice of Seas H400-06 1" dome tweeters was based on several factors:

- Adequate frequency range (700Hz–18kHz 5dB)
- Reproducible driver characteristics
- Ready availability from a variety of vendors
- Cost ($16.90 each for 12 in a lot of 14; total with freight = $215)

ATTENUATING CIRCUMSTANCES

To reduce the number of iterations required to arrive at the correct tweeter array/bass panel balance, I included an L-pad attenuator (Radio Shack #40-980). Based on experiments with alternate tweeter connection schemes, I guessed that the tweeter output would need to be attenuated slightly for flat response in a "hard" room, but that little or no attenuation would be needed in a "softer" acoustic environment.

Knowing some of the tweeter/bass panel characteristics and guessing at others, I used Ralph Gonzalez's "LMP Professional" software to select the correct driver polarity, crossover capacitance, and the relative distance of the tweeter array bass panel from the listener.[4] This software is of great practical utility in designing speakers: it integrates driver properties, crossover design, phase relationships, and relative driver displacement to derive frequency response curves and estimate waveforms. I used the values in *Table 1* (the relative treble/bass sound pressure can be adjusted via the L-pad) in the final simulation. [*Note: LMP Professional software is available from Old Colony Sound Lab.—Ed.*]

When the phase of the tweeter array panel is inverted relative to the bass panel, and the crossover capacitance is 2.57μF, the predicted result is as shown in *Fig. 6*. This response is consistent with a crossover frequency of 2.4kHz. The 2.57μF crossover capacitance is the measured value of a pair of 3μF Ultra Kaps, although other values from 2.25–2.75μF would work as well.

I used a value of 9.5Ω in the LMP program to represent the combined L-pad/tweeter array impedance. This nominal value is determined by characterizing the L-pad and then calculating the combined impedances (*Table 2*).

An L-pad attenuator is a pair of ganged variable resistors which change simultaneously as the stem is rotated, as illustrated in *Figs. 3* and *7*. One is wired in series with the driver, the other in parallel. When used with an 8Ω driver, the design intent is to hold the effective impedance of the combined driver/attenuator constant at 8Ω, regardless of the actual current passing through the driver. A real L-pad and a 9Ω driver impedance (tweeter array) result in a network resistance that varies slightly with attenuation (*Table 2*).

Attenuation can be calculated with the following expression:

$$\text{Attenuation (dB)} =$$

$$20 \times \mathrm{LOG}[(1 + R_D/R_P) \times (R_S + 1/(1/R_P + 1/R_D))]/(R_{SO} + R_D)]$$

where:

R_D = Impedance of the driver
R_P = Impedance of the L-pad parallel resistance element
R_S = Impedance of the L-pad series resistance element
R_{SO} = Value of R_S at the lowest attenuator setting (dB = 0)

Typically, $R_{SO} < R_D$ and can be ignored.

Continued on page 49

TABLE 2

ATTENUATOR CIRCUIT CHARACTERISTICS

L-Pad Dial Position Degrees	Var. Parallel (Rp) Impedance Measured (Ω)	Var. Series (Rls) Impedance Measured (Ω)	Total Network Impedance Calculated (Ω)	Attenuation Calculated (dB)
0	Very Large	0.0	9.0	0.0
20	Large	0.5	9.5	0.5
40	30.8	0.9	7.9	1.0
60	29.0	1.3	8.2	1.5
80	26.6	1.7	8.4	2.0
100	24.0	2.2	8.7	2.5
120	21.6	2.7	9.1	3.1
140	19.4	3.2	9.3	3.6
160	17.0	3.6	9.5	4.1
180	14.2	4.1	9.6	4.8
200	11.9	4.5	9.6	5.5
220	9.7	5.1	9.8	6.4
240	7.7	5.4	9.5	7.2
260	5.0	6.0	9.2	9.1
280	2.6	6.5	8.5	12.5
300	0.4	6.9	7.3	25.6
320	0.0	7.0	7.0	57.0

NEW CALSOD 3.0 PROFESSIONAL SOFTWARE

This great package from Australia is the brand new upgrade to the CALSOD 2.50 Professional software from which smaller, CALSOD Standard versions such as 1.20 and 1.30 were originally extracted. Although CALSOD Standard 1.30 is an excellent and popular package which more than does the job for most people, Professional CALSOD 3.0 differs in that it is more extensive and aimed (as CALSOD was originally intended) at professional engineers, as shown below. By Witold Waldman/Audiosoft.

FEATURE DESCRIPTION	CALSOD 1.30	3.0
Modeling of sound pressure and impedance responses of vented-box closed-box, passive-radiator, and filter-assisted systems	Y	Y
Closed rear chamber and vented front chamber bandpass enclosure models for sound pressure and impedance response calculations	Y	Y
Driver impedance models with frequency dependent voice-coil inductance	Y	Y
Calculations incorporate effects of geometric placement of drivers on the loudspeaker baffle to account for time offsets	Y	Y
Simple manual curve fitting procedure for creating accurate driver sound pressure and impedance models	Y	Y
All calculations embody both the magnitude and phase response, and the driver models include minimum-phase equalizers for introducing peaks and dips in the response to accurately model actual driver behavior	Y	Y
Fast passive crossover network simulation and optimization using up to 60 components, with arbitrary component and loudspeaker connections	Y	Y
Standard filter target functions for Linkwitz-Riley, Butterworth, Bessel, constant voltage, and user-defined filter functions	Y	Y
Extensive graphics, including loudspeaker impedance and sound pressure response, crossover network filter transfer function, system input impedance, SPL on-axis and off-axis response, and SPL power response	Y	Y
Utility program for designing air-cored inductors	Y	Y
Simulation of baffle diffraction step	Y	Y
Simulation of effects of floor reflections on low-frequency response	Y	Y
Ability to provide simple analytical simulations for design studies	Y	Y
Support for HP LaserJet and DeskJet printers, as well as popular 9-pin and 24-pin dot matrix printers from IBM, Epson, NEC, and Toshiba	Y	Y
Screen plots can be saved as PCX files for importing into desktop publishing and word processing software	Y	Y
Full user's manual supplied on disk, complete with tutorial examples and formulas for initial filter network design	Y	Y
Double ported bandpass enclosure models for sound pressure and impedance response calculations	N	Y
Vented-box, closed-box, and passive-radiator simulations include the effects of enclosure leakage, absorption, and port/passive-radiator losses	N	Y
Active crossover network simulation and optimization	N	Y
Impedance optimizer for designing zobel and conjugate load matching networks to meet a desired target impedance function	N	Y
SPL optimization using up to 5 on-axis and off-axis observation points	N	Y
Powerful curve-fitting optimizer for determining sound pressure and impedance models of loudspeaker drivers	N	Y
Importing of SPL and impedance data from MLSSA, SYSid, System One, LMS, IMP loudspeaker test systems, incl. SPL optimization using up to 5 observation points to include off-axis radiation behavior	N	Y
Optimizing Thiele/Small parameter estimator utilizing impedance data from added mass, differential box volume, and vented-box impedance measurements, including frequency-dependent voice coil inductance	N	Y
Piston approximation for driver models to simulate driver off-axis radiation characteristics	N	Y
Specify orientation of principal radiation axis of each individual driver model used in the loudspeaker system	N	Y
Simulation of the effects of room gain on low-frequency response to help the tuning of low-frequency alignments	N	Y

Purchasing options available:

SOF-CAL3B3G	CALSOD 3.0 PROFESSIONAL Software, 3½" IBM	**$269.00**
SOF-CAL3B5G	CALSOD 3.0 PROFESSIONAL Software, 5¼" IBM	**269.00**
SOF-CALSODUP3	Upgrade from CALSOD 1.20 or 1.30 STANDARD to CALSOD 3.0 PROFESSIONAL. Please give CALSOD 1.20 or 1.30 registration number and specify new disk size.	**199.00**

ROOM DESIGN POWERSHEET SOFTWARE

Marc Bacon

NEW THIS ISSUE

A companion to the famous LOUDSPEAKER DESIGN POWERSHEET, this program covers a wide range of knowledge in easy-to-use spreadsheet format. Working from a main menu, the user can access programs dealing with room resonances, reverberation, boundary augmentation, wall diffuser design, resonance traps, and so forth. Since architectural acoustics can have more effect on final sound than anything else in the audio chain except the speakers themselves, the ease of being able to work through countless options without spending a dime on remodeling or moving furniture is very appealing. As with LDP, the RDP is priced extremely low in relation to the amount of programming effort, in order to make computer-aided architectural design available to everyone.

An unprotected source code allows the user to customize and build upon individual spreadsheets for his own use. Context-sensitive HELP and an introductory README.1ST file are also included. Requires an IBM PC or compatible with 640K of memory, hard disk, and Lotus 1-2-3, Quattro-Pro, Excel, or any other spreadsheet which can use Lotus *.WK1 files. PLEASE NOTE THAT SPREADSHEET SOFTWARE IS NOT INCLUDED.

ROOM DESIGN POWERSHEET MENU TREE

Menu Level 1	2	3
Main menu		
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	Room resonances below 250Hz	
		Help
	Room reverberation above 250Hz	
	Loudspeaker room boundary effects	
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STEREO DESIGN FOR THE HOME BKAU1 $7.95

Ray K. Dehn

Still in its original dust jacket, this is the first book ever published about the design of home stereo systems. Actual installations are shown in great detail through the use of 108 illustrations and photographs. Topics covered include how to save money and floor space; choosing your stereo components; the qualified dealer; how much you should spend; the effects of room environment and what to do about it; FM multiplex broadcasting; proper volume level; how much amplifier power you really need; operating your equipment; and (!) future developments and obsolescence. Chapters are: The Exciting Stereo Idea; Drapery—An Ideal Stage; Stereo Systems in Bookshelves; Speaker Staging with Cabinets; Speaker Systems—Their Design and Selection; Selection of Stereo Components; Operating Your Equipment; and You and Your Dealer. A collector's classic that belongs on every audiophile's bookshelf! 1962, 203pp., 6¼" x 9¼", hardbound.

THE THEORY AND DESIGN OF LOUDSPEAKER ENCLOSURES BKSA1 $24.95

J. Ernest Benson

This recent reprinting highlights the work of the man who was Richard Small's mentor and indeed has been called the Australian (Harry F.) Olson. The work was originally published in three parts in the *Amalgamated Wireless Australia Technical Review* in 1968 ("Electro-Acoustical Relations and Generalized Analysis"), 1971 ("Response Relationships for Infinite-Baffle and Closed-Box Systems"), and 1972 ("Introduction to Synthesis of Vented Systems"). Don Keele has said of this book: "It is a classic, and even more comprehensive and detailed than Thiele and Small's loudspeaker papers as published in the *AES Journal* (if you can believe that!). He goes into an exhaustive analysis of the infinite-baffle, closed-box, damped vented-box, passive-radiator vented-box, and acoustic-resistance controlled systems. The papers are very instructive and a must-read for anyone seriously interested in low-frequency cabinet design. It contains the only complete mathematical model and formula that includes all the previously mentioned systems that I know of... It's very mathematical, but written in an easy-to-understand manner. I highly recommend it!" We recommend it as well, with the understanding that it is indeed very technical and is certainly not for beginners. 1993, 244pp., 6" x 9", softbound.

AUDIO TEST & MEASUREMENT: CONFERENCE PROCEEDINGS BKAS9 $34.95

These thirty-two papers from the Portland, Oregon, Eleventh International Audio Engineering Society Conference of 29-31 May 1992 cover both the engineering and production aspects of testing, including state-of- the-art techniques. Authors examine electronic, digital, and acoustical measurements, bridging the gap between subjective and objective measurement to advance the science of audio testing. 1992, 359pp., 8¼" x 11¼", softbound.

305 CIRCUITS BKAA28 $27.70

Elektor Electronics Magazine

This recent addition to the famous "Circuits" collection demonstrates the practical aspects of electronics through projects that can be built at home, in a small workshop, or in the physics or science departments of schools and colleges. The volume includes projects for everyone, covering audio and hi-fi; computers and microprocessors; music and electrophonics; radio, television, and communications; and test and measurement equipment. United Kingdom, 1993.

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PLEASE NOTE THAT OLD COLONY NO LONGER CARRIES SOF-CAP1B5G, THE SHAREWARE VERSION OF PC-ECAP WHICH SOLD FOR $9.95.

DAT TECHNICAL SERVICE HANDBOOK BKVN5 $49.95

Richard Maddox

DAT recorders are already in use in 75% of the recording studios in the US, but there is a critical lack of reliable information on service and repair. This reference fills that need, supplying a wealth of technical detail and practical advice for engineers and technicians. Included are step-by-step instructions for regular maintenance and repair; detailed techniques for maintaining and adjusting tape transport and head alignment; complete how-to instructions for replacing the record/play head drum; a complete engineer's checklist for DAT troubleshooting; and specific service notes, by make and model. 1993, 256pp., 6" x 9".

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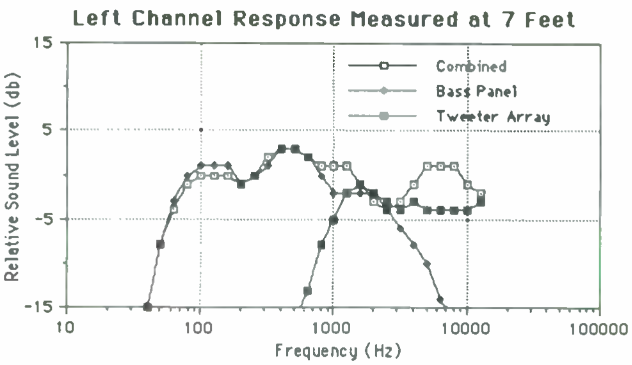

FIGURE 8: Actual system response (left-channel response measured at 7′).

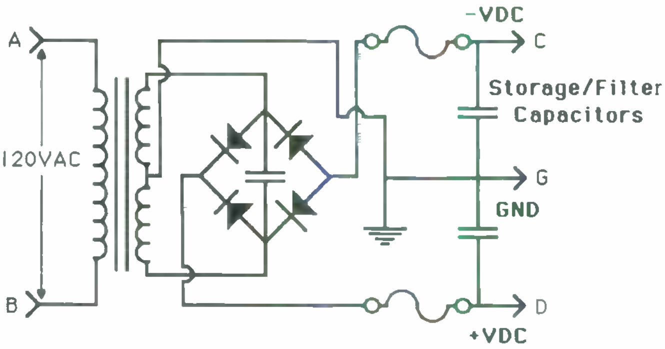

FIGURE 9: Typical amplifier power supply.

Continued from page 46

Fixed resistors may be chosen instead of the L-pad, or the attenuation circuit may be eliminated altogether. Listening areas vary in tonal balance; however, including the L-pad virtually guarantees a setting that will meet your needs.

BUILD A DAGWOOD

The "sandwich" design of the tweeter towers hides the wiring and crossover components within the unit, so assembly requires some experience with power tools (*Photo 1*). To place the capacitors, hook-up wires, and L-pad pot between the oak and particleboard layers, you must create compartments or channels using a router. I chose 9/8″ particleboard for its mass and thickness, and because it allows use of fairly large-diameter capacitors. The volume control, bayonet-type fuse holder, and input cable are all mounted in the particleboard.

The tweeter connecting wires are channeled through a groove that runs from top to bottom of the oak faceplate tangential to the tweeter mounting holes. All connections are made with Wonder Solder; the tweeters are connected with 20-gauge multistrand wire. The input lead, constructed from RG-8 Mini-Foam cable (Radio Shack #278-1323), is permanently anchored inside the tower with wire staples. The other end is connected to the bass panel via a high-quality, gold-plated RCA plug. A female receptacle mounted on the bass panel's terminal plate makes it possible to disconnect the tweeter tower.

After all the holes were cut, the routing completed, and edges rounded, I assembled the particleboard frame and base. I filled the surface with a wood-filling compound and sanded prior to adding three coats of flat-black lacquer, then attached components using solder and RTV (GE silicone). The oak faceplate was similarly prepared by cutting holes for the tweeters, rounding the edges, and

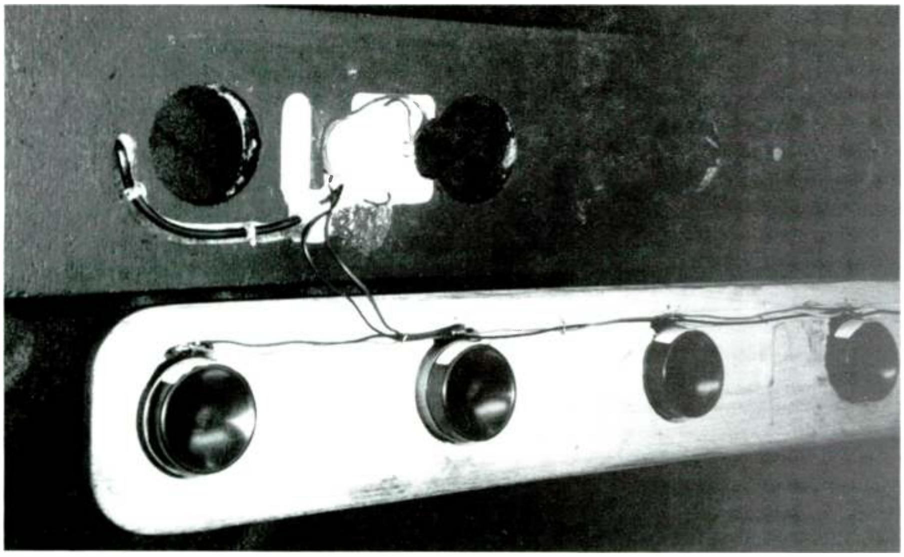

PHOTO 1: Tweeter tower assembly.

drilling and countersinking screw holes prior to sanding and sealing with "Red Oak" hand-rubbed stain and oil-resin sealer. Brass screws mount the tweeters to the plate; the oak faceplate is mounted to the particleboard frame with ¾" wood screws behind stained and sealed caps. The tweeter towers are placed outboard of their respective bass panels (the old MG-1s with disconnected tweeter panel). They should be aimed at the preferred listening spot and located at a distance from the listener which is roughly equal to that of the bass panel. You can adjust the tilt angle to match the MG-1s by making adjustment screws from carriage bolts, wing nuts, and washers, mounted through a threaded insert (T-nut) and applied from the bottom of the tower base.

After several days of break-in, the assembled system had a reasonably flat frequency response (*Fig. 8*). Please be patient: the Seas tweeters increase in output level and smoothness over the first 40–50 hours of operation.

Take some care in positioning the panels relative to your listening area. The separate bass and treble panels offer numerous possibilities, but I think you will like the results. I found the improvement in sound quality dramatic, especially in detail, imaging, and depth of stage. The air and presence now match the clarity of the midrange/bass.

SUBWOOFER ADDITION

At the other end of the audio spectrum, you can gain noticeable improvement with a subwoofer. Although the quality of bass from the MG-1 is very good above 60Hz (provided you have an amplifier equivalent to the task), for earth-rumbling reproduction of all sonic information encoded on CDs—both heard and felt—some assistance is required from 20–60Hz.

Many fine subwoofers of compact design are available, so I'll not go into great detail about mine.[5] Let's just say I went the time-honored route of building a monstrous enclosure to accommodate a monstrous driver—the Hartley 224HS. This driver employs a magnetic suspension design; the cloth cone is impregnated with a polymerized resin for increased durability and reduced sensitivity to humidity. The effective

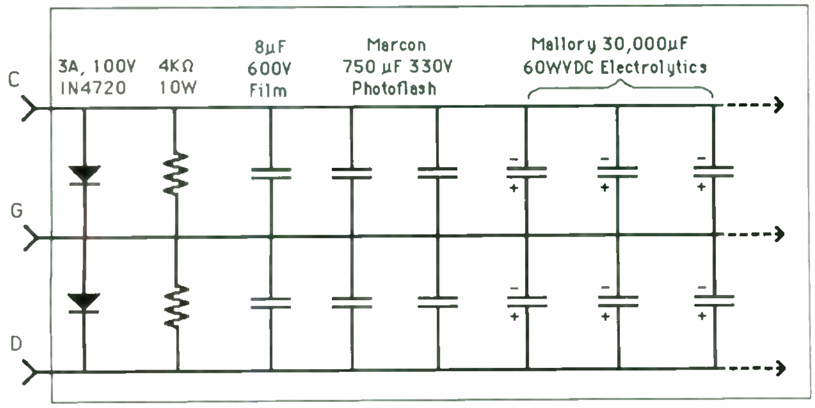

FIGURE 10: Outboard capacitor bank.

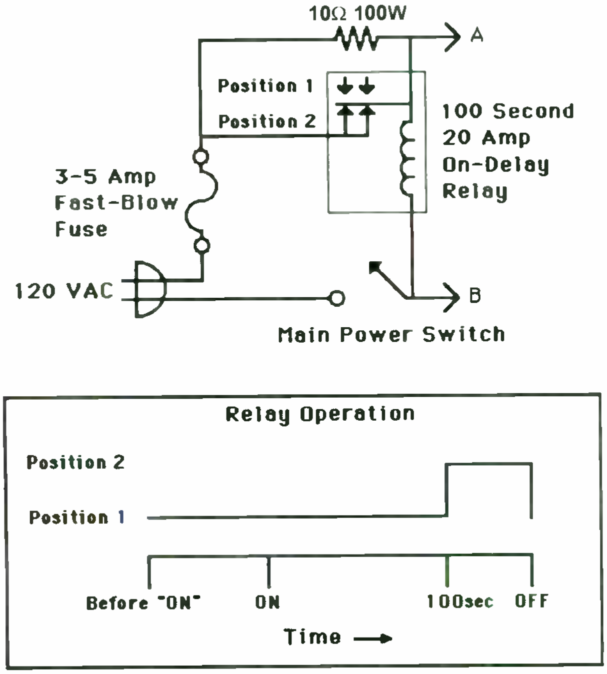

FIGURE 11: Soft-start protection circuit.

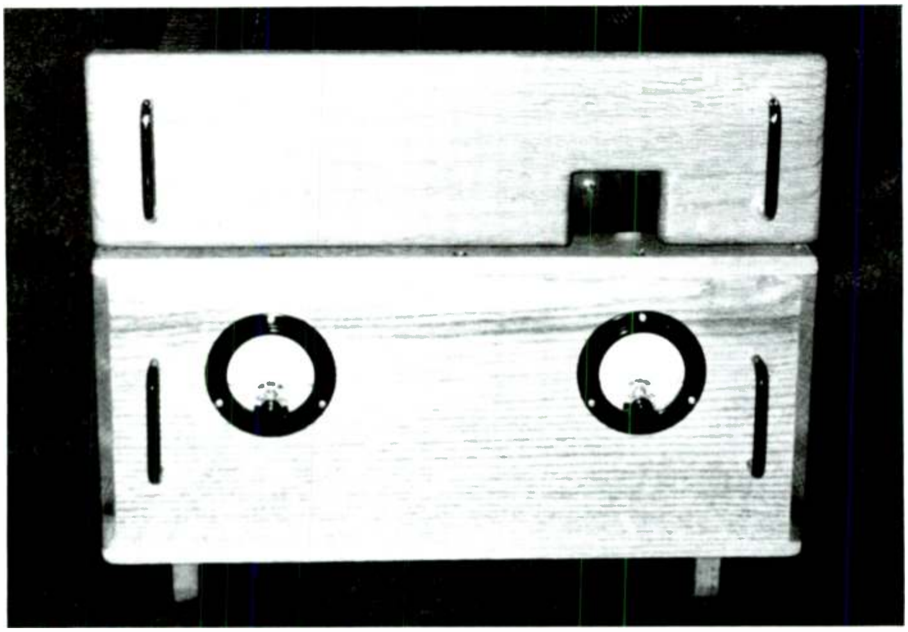

PHOTO 2: Power amp with capacitor bank.

piston diameter is 21.5", with a nominal impedance of 5Ω.

The driver is centrally mounted in the source-end wall of a specially constructed 11′ × 22′ listening room. Its center is 30" off the floor: half the height of the bass and tweeter panels, which are positioned on either side. The wall's reinforced frame is made from 2" × 6" lumber. All joints are fastened with adhesive and screws, as are the successive laminations of high-density particleboard and tongue-and-groove cedar paneling that comprise the vibration-resistant surface. The enclosure volume is 150 ft.^{3}; 100 ft.^{3} is the manufacturer's suggested minimum volume for maximum performance.

The 224HS is powered by a Rotel RB-850 power amp switched to the "bridged" mode, where it is rated at 150W (monaural) into 8Ω. To prevent disasters, I connected a 100W 5Ω resistor in series with the driver: the Rotel can clip when driving lower impedance loads.

I use Audio Control's Richter Scale Series III as both the crossover (24dB/octave, 90Hz) and bass equalizer.[6] This device easily extracts and controls a summed low-bass channel from the output of a stereo preamp, with minimum distortion of high- and low-frequency information. In addition, bass output can be equalized from 22.5–250Hz by means of a built-in warble-tone generator, matched mike, six-band equalizer, and subwoofer volume control. In this specific application, no further equalization is needed once the subwoofer output level has been properly set. Output within 4dB from 22.5–250Hz (center of the room; 4′ off the floor; 10′ from the subwoofer) is attained with the equalizer circuit disengaged.

Correctly set, the woofer amplifier gain and speaker phase result in gratifying sonics, notwithstanding arguments about loss of low-bass imaging when using a single subwoofer.[7] With reasonably flat response to 20Hz, music becomes more natural and everything is audible (partly due, no doubt, to biamping in this power-hungry portion of the audio spectrum). If you don't like hearing the pedal mechanism of a harpsichord or the tapping of a musician's foot as he keeps time, however, you may not wish to make this last addition to your system.

BASS SOLIDITY

I considered the power amp used in my original setup—the Hafler DH-200, conservatively rated at 100W/channel into an 8Ω load—more than adequate to tame a wide range of speakers. By using a long-established "soup-up" technique, however, I found I could greatly improve the MG-1's mid-bass solidity.[8, 9] The basic idea is that tight bass requires instantaneous demand current. One way to achieve this is to increase the power

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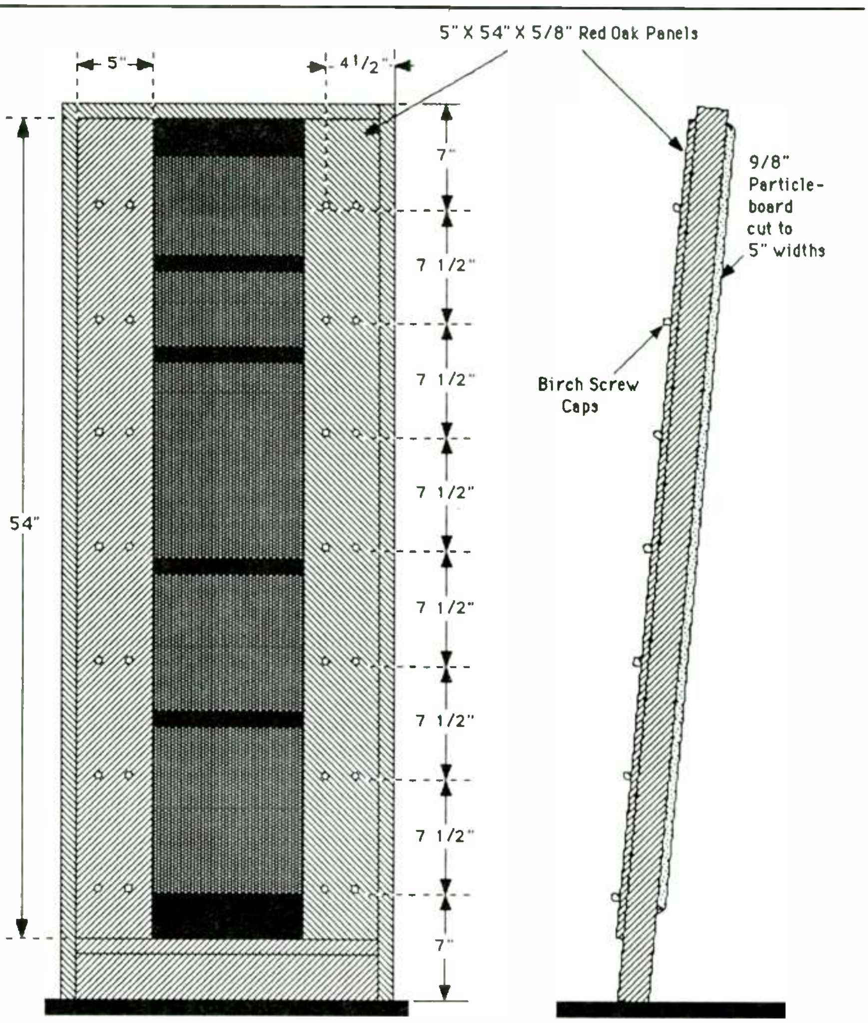

FIGURE 12: Reinforced bass panel.

supply's current reserve and reduce its effective impedance by adding capacitors in parallel to the filter caps (*Photo 2*). A typical power supply is shown in *Fig. 9*. The transformer "transforms" the voltage applied to the diode bridge, a rectifier emitting pulsed-direct current which is rendered "ripple-free" by large electrolytic capacitors. "C" is the negative rail; "D" is the positive rail; and "G" is common ground. Regulators may be added if various amplification stages require further voltage control. *Figure 10* shows an outboard capacitor bank configuration with an extra 90,508µF per side. With extreme caution, this can be extended up to a total of 250,000µF per side (my current setup). Sonic improvements are audible, albeit diminishing, with each additional 30,000µF increment. Handling electrical charges of this magnitude at 50, 60, 70, or 80V DC is risky business, even if you know exactly what you are doing! An accidental connection can lead to equipment damage or serious personal injury. For a detailed and scholarly discussion of this technique, please refer to Walt Jung's article in *Audio.*[10] Some key points are worth mentioning here. Choose capacitors that are conservatively rated for the desired operating voltage. Polar electrolytic capacitors *must* be connected according to the polarity indicated on the terminals. Since most electrolytics need to be "formed" before use, an external bench power supply will come in handy. Wir-

Resource for This Article

Loudspeaker Modeling Program (LMP) Professional (also known as "Souped Up") is available for IBM ($49.95) or Macintosh ($39.95) from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458; (603) 924-6526, FAX (603) 924-9467. Please include $3 s/h.

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ing must be heavy gauge to minimize impedance and great caution taken to avoid shorting to ground (or the amplifier chassis).

I've learned from experimentation that adding photoflash and high-quality film capacitors in parallel with the big electrolytics improves detail and removes sonic "grunge." Resistors added in parallel help the capacitor bank drain when the amp is switched off. Even so, several seconds may elapse before the bank voltage drains to zero. I suggest hooking a voltmeter from each of the "hot" rails to ground, so you can monitor the charge/discharge of the capacitor bank. Knowing when the amp is effectively off is helpful. Finally, you must add a circuit to damp the current surge created by initial connection of your rectifier bridge and transformer to this large capacitor bank. When the amplifier is switched on, the transformer and rectifier circuits are connected to a virtual dead short. A "soft-start" circuit will protect your amplifier and provide years of good service (*Fig. 11*). Without such protection, the initial current surge will blow fuses, burn up the rectifier bridge, and degrade the transformer. After using this setup for the greater part of a decade, I've experienced no electrical problems with these modifications; however, there may be practical limitations to the size of the capacitor bank feasible for your amp.[11]

To limit this current surge, momentarily insert a 10Ω 100W resistor between the 120V AC line and the power transformer. A time-delay relay with a built-in, solid-state adjustable timer (Syracuse TIR115A1002) set to bypass the resistor 100 seconds after initial power-on works nicely.

Allow sufficient time for your amp to adjust to the new capacitors. The longer it is on,

REFERENCES

1. Moncrieff, J.P., "The Sonic Importance of Passive Parts," *International Audio Review Hotline* 13 (September 1981): 1–13.

2. Moncrieff, J.P., "The Sonic Importance of Passive Parts," *International Audio Review Hotline* 14 (October 1981): 1–13.

3. Benson, K.B., "Dividing Network Systems," *Audio Engineering Handbook*, 7.42. McGraw Hill, 1988.

4. Gonzalez, R., "An Introduction to Frequency Response and LMP," *Speaker Builder* (1, 2/87): 18, 42.

5. Cushing, C., "A More Compact Transmission Line Subwoofer," *Speaker Builder* (1/87): 9.

6. Galo, G., "Audio Control's Richter Scale," *Speaker Builder* (2/87): 50.

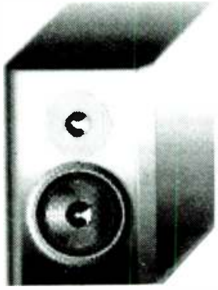
7. Frane, J., "Matching a Subwoofer (or Two) to Your System," *Speaker Builder* (3/88): 45.

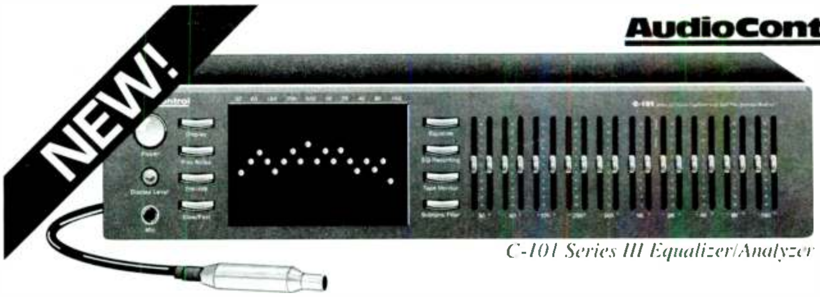
8. Jung, W., "Building an Energy Storage Bank," *Audio* (8/80): 38.

9. Jung, W. and Dick Marsh, "POOGE-2: A Mod Symphony for your Hafler DH200 or Other Power Amplifiers," *TAA* (4/81): 7.

10. Jung, W., *Audio* (8/80): 38.

11. Staggs, D.E., "Megafarad Power Supplies," *TAA* (4/91): 44.





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and the more frequently it is used, the better it will sound. The capacitors take time to form after long idle periods, but the improvement in sound is definitely worth it. This process will tighten and define the spatial location of the bass: the directionless "boom" that was once a pillow-stuffed bass drum will become a precisely located "thud."

A final modification that audibly improves lower midrange and bass response also improves speaker appearance. When driving the units with 350–125Hz warble tones, I could feel vibration in the metal panel that supports the Mylar® (bass) diaphragm. After all, the sound-emitting element is a tightly stretched 9½" × 45" plastic film. This should give rise to harmonic frequencies at 151Hz, 301Hz, 452Hz, 603Hz, 714Hz, 753Hz, and so on.

The metal support panel is not particularly rigid, but it is flat and warp free—properties important to reliable performance. Adding front and back braces that clamp to the frame top-to-bottom will attenuate sympathetic vibrations, better defining the diaphragm location in time and space. The result is enhanced sound clarity and sonic image solidity.

For structural reinforcement, I fabricated pairs of panels, right/left and front/back, and mounted them with screws to the metal support panel (*Fig. 12*). The front panels are ⅝" red oak, 5" wide × 54" long, two per speaker panel; the back panels are made from the same particleboard as the tweeter tower, 1⅛" thick × 5" wide × 45" long. I chose the materials based on warp resistance, weight, and aesthetics. Fourteen holes per panel accommodate anodized wallboard screws that pass through the oak panel and the steel support panel, and anchor in the particleboard.

Countersinking the screws enables the use of wood plugs (screw caps), which hide the heads for a professional-looking finish. The rear particleboard braces are routed to fit flush to the diaphragm frame, over the six pop-riveted clamps that hold the frame to the metal support panel. I applied red oak stain to the

SOURCES

TRT
Box 4271
Berkeley, CA 94704
(Ultra Kap capacitors)

Old Colony Sound Lab
PO Box 494
Peterborough, NH 03458-0494
(603) 924-6371
FAX (603) 924-9467
(LMP software)

California Audio Labs
7281 Garden Grove Blvd.
Suite E
Garden Grove, CA 92641
(714) 894-9747

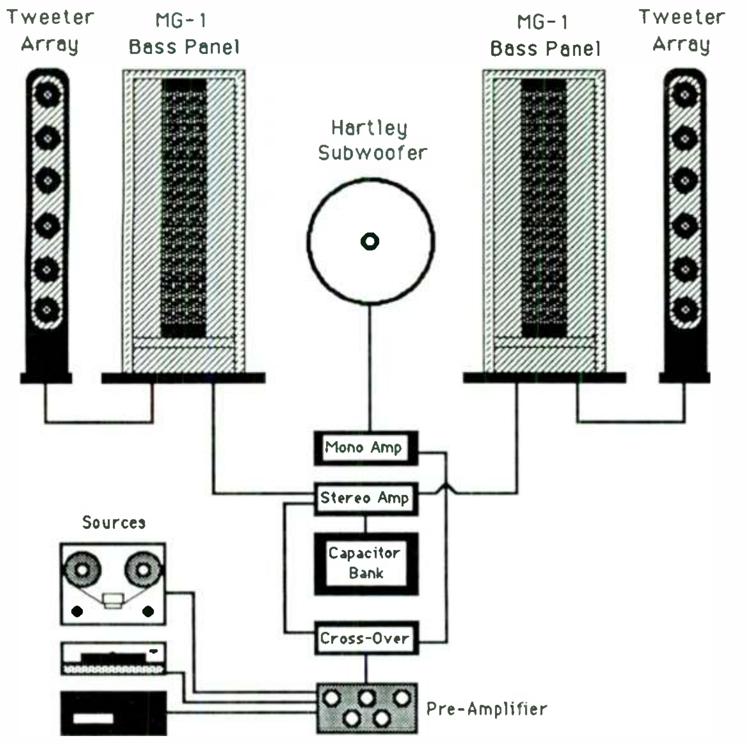


FIGURE 13: Current state of evolution.

properly sanded panel and allowed it to dry before applying a sealer.

CONCLUSION

Figure 13 shows the overall hook-up diagram for the evolved system. My goal was accurate, credible reproduction of a live performance rather than dramatic or exciting transformation of a recording. The changes described here are the most gratifying of many attempted modifications.

Sources for listening tests included a California Audio Labs Aria Mark III(b) CD player, a Simiko Blue Point cartridge, Grace 707 tonearm, Kenwood KD-500 turntable, and Platter Matter™ damping pad. Amplification was in two discrete stages: a conrad-johnson preamp (modified PV2A), and power amplifiers from Hafler (extensively modified DH-200) and Mark Levinson (unmodified #27).

Auditioning my favorite recordings, I find the music detailed and effortless, with well-defined sound source location and depth of field. The musical frequency range is easily spanned: bass drum and tympani are spectacular; massed strings are melodic; and the attack and resonance of plucked strings or brass percussion are outstanding. Overall performance compares favorably to reference systems employing Theil 3.5s or Magnepan MG-IIIs.

My efforts have been well worthwhile, judging from the quality of the finished product and the feeling of accomplishment as I put on my favorite recording, sink into my listening chair, and am transported to a higher realm. Which changes were most significant? Well, this experiment spanned a decade, but I'd rank the changes in the following order (from most to least significant): tweeter array; capacitor bank addition; crossover-capacitor upgrade; subwoofer addition; bass panel reinforcement; grille-cloth removal.

In light of the effort and dollars spent to obtain earth-rending bass, some people might say I received a relatively modest return on my investment. Personally, I'd do it all over again. I hope my experiences will encourage you to take soldering iron, saw, and screwdriver in hand, and experiment on your own. Happy sonic adventures!

ACKNOWLEDGMENT

I'd like to acknowledge the collaboration of Kenneth B. Buell, audiophile, mechanical engineer, and inventor. Our ongoing debate on the best routes to sonic truth, along with his keen insight and intuition, have been a source of inspiration.

LOW-FREQUENCY AC-TO-DC CONVERTER

By Homero Sette Silva

When evaluating T/S parameters, such as Q_{TS} or V_{AS}, we sometimes face the problem of measuring AC signals with frequencies lower than 20Hz. Unfortunately, most digital multimeters (or analogics) can't do that because of excessive drop in the coupling capacitor.

To remedy this situation, I have developed a full-wave precision rectifier (*Fig. 1*) that converts a low-frequency AC signal to a DC value which corresponds to the average value (not the peak value, as in common capacitor-filtered rectifiers) of the input signal modulus.

VALUE JUDGMENT

I achieved this with IC2A and associated passive components. Four diodes form the bridge rectifier, which, when placed inside the operational amplifier feedback loop, behave like ideal ones even for signals much lower than 0.7V (diode drop). Filter capacitors are used to remove the AC components, preserving only the DC mean value.

Large capacitors provide the best filter action but take a longer averaging time. You have a choice of two capacitor values for fast and slow operation. Switch to slow for stable voltage readings at the low-frequency end. If you aren't satisfied, try other values.

You must provide a discharge path (10k resistor) for the capacitor; otherwise, it will be unable to successively average the input signal. Increasing the resistor value allows you to reduce capacitor size proportionally, but temperature effect in the diode reverse current will be of concern.

IC1A and IC1D are differential amplifiers with unity gain. While not essential to circuit operation, they are rather convenient. The former, due to its high input impedance, ensures negligible voltage drops even with small-diameter wires between workbench and suspending loudspeaker. As a bonus, picked-up noise is greatly reduced by differential cancellation.

IC1D enables you to use voltmeters with almost any input impedance without disturbing the averaging circuit. High-input-impedance voltmeters (1MΩ or greater)

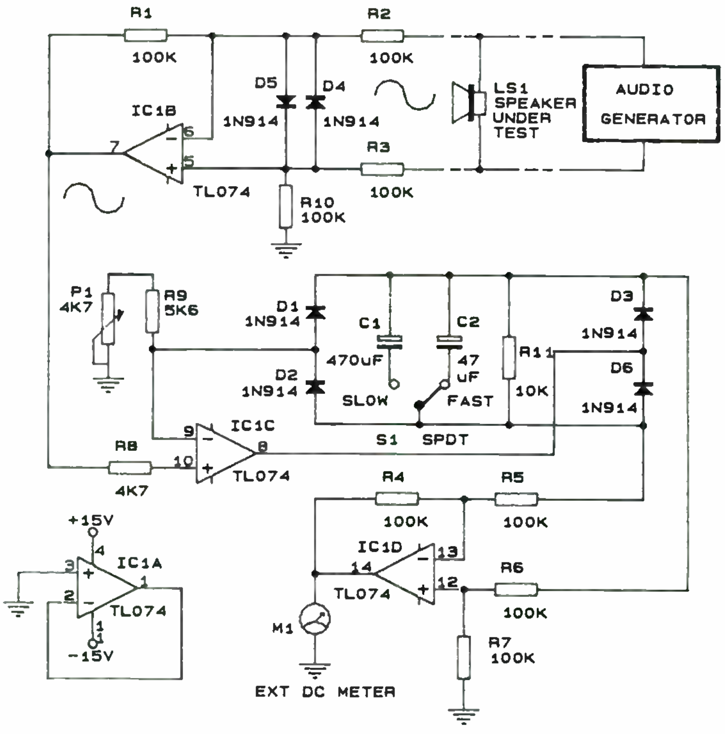


FIGURE 1: AC-to-DC converter, short version.

TABLE 1

AC-TO-DC CONVERTER PARTS LIST

QTY.	PART	DESCRIPTION
1	C1	470μF
1	C2	47μF
6	D1–6	1N914
1	IC1A	TL074
1	LS1	Speaker under test
1	M1	Ext DC meter
2	R8, P1	4k7
8	R1–7, 10	100k
1	R9	5k6
1	R11	10k
1	S1	SPDT

ABOUT THE AUTHOR

Homero Sette Silva holds a BS in electrical engineering and has taught electronics at several universities in Rio de Janeiro and at a technical high school. His main interest in audio is in system modeling and performance evaluation, and for the past two years he has trained technical personnel in loudspeaker design and Thiele/Small theory. He has designed a Thiele/Small parameter analyzer.

CURRENT DRIVE SOURCE

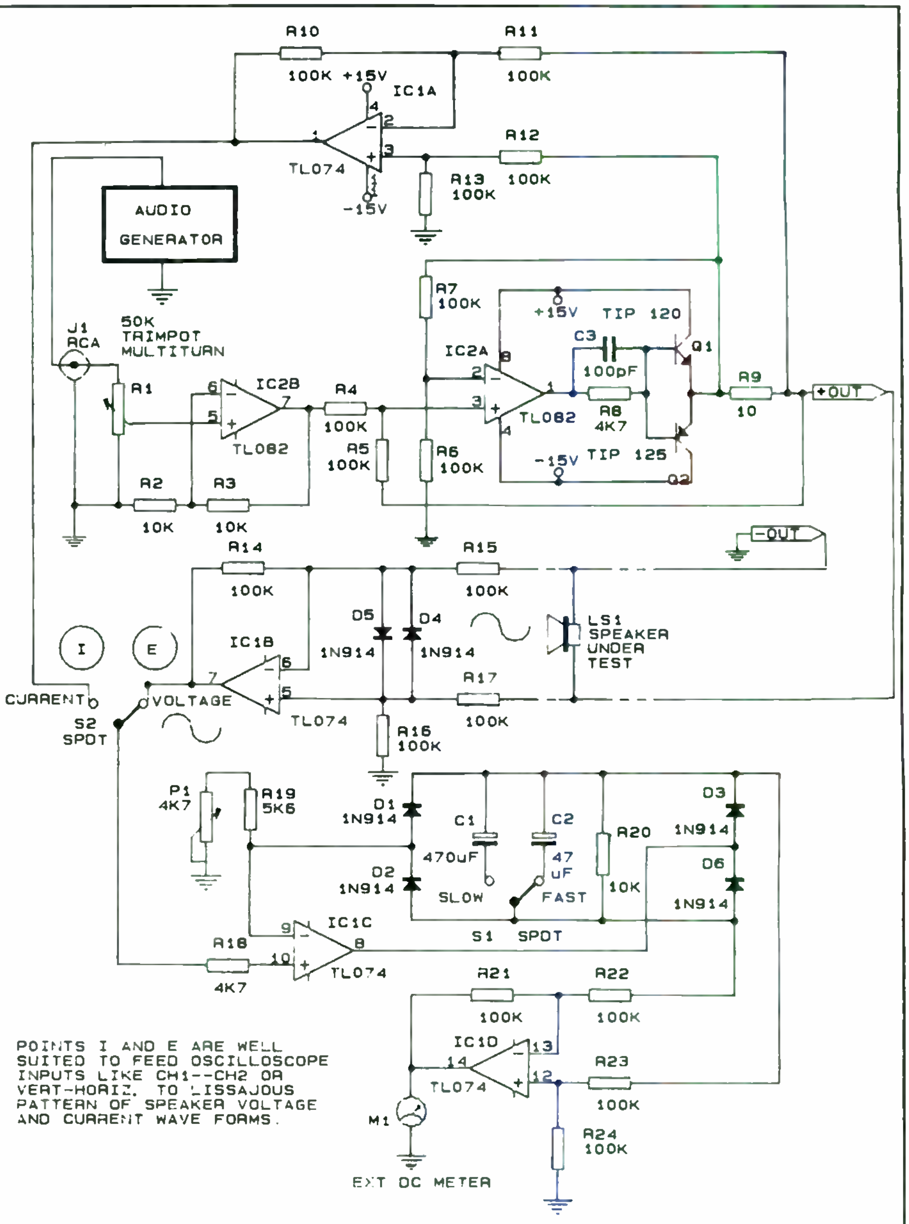
When measuring a driver's or enclosure's T/S parameters, a current drive source is of great help. In this case, resistance and impedance moduli can be read directly in the volts scale.

Frequencies involved in this process can be quite low—under 20Hz, for example—with some high-compliance drivers. (Consider Q_{TS} measurement, where it is necessary to read a frequency both above and below F_S.) We must face the problem of increasing inaccuracy with decreasing frequency when dealing with voltmeter readouts in the AC scale. A simple AC-to-DC converter (a full-wave precision rectifier) allows us to read them in DC scales, free from the trouble of coupling capacitors.

Another desirable feature is the possibility of displaying voltage and current waveforms on a CRT screen. This enables easy detection of the zero-phase points of driver resonance F_S, box-tuning frequency F_B, and impedance-peak frequencies F_L and F_H. It also conveniently shows distortion caused by suspension and magnetic nonlinearities, since the zero-phase straight line of Lissajouss patterns at resonance are very sensitive to harmonics.

The circuit in *Fig. A* comprises the current source and the AC-to-DC converter. The current source circuit is the same one that appeared in *SB* 3/92 ("Successful Circuit," p. 64), with a correction: R9 = 10Ω. To improve high-frequency stability, it also has a 100pF capacitor bypassing R8.

The AC-to-DC converter now has a differential amplifier which converts the R9 loudspeaker current to voltage, and a switch enabling both sinusoidal voltage and current measurements in a DC voltmeter scale. Filter capacitors C1 (slow) and C2 (fast) remove all signal components except the mean value after a full-wave precision rectification (2 × Peak/Π).



POINTS I AND E ARE WELL SUITED TO FEED OSCILLOSCOPE INPUTS LIKE CH1--CH2 OR VERT-HORIZ. TO LISSAJOUS PATTERN OF SPEAKER VOLTAGE AND CURRENT WAVE FORMS.

FIGURE A: Low-frequency AC-to-DC converter.

The four-wire link allows you to suspend the loudspeaker five or more meters from the bench without prejudicing accuracy due to voltage drop, even with small-diameter wire. If you need an RMS reading, multiply the mean value in the DC scale by 1.111.

can be directly wired across 10kΩ resistors without harm to circuit operation. You can even use a low-impedance analog voltmeter. A 10kΩ/volt VOM in the 1V scale is equivalent to a 10kΩ resistor. In this case, simply suppress the discharge resistor, since voltmeter internal impedance does the same job.

BREADBOARD OF HEALTH

To evaluate operation, breadboard the circuit using two TL082 dual-operational amplifiers or a single TL074 quad. Feed a 1V RMS sinusoidal signal with a frequency around 60Hz to the 100kΩ input resistors. Adjust the trimpot until you read 1V in the DC scale of a voltmeter connected to IC1D output and common. You now have an AC voltmeter calibrated to RMS readings for distortionless sinusoidal input signals.

For a peak-reading voltmeter, adjust the trimpot to 1.414V for the same 1V RMS input, or change the input to one peak volt and adjust it to a 1V reading. Remember, this will hold only for sinusoids, since the averaging circuit is intrinsically a mean-value indicator.

Change the frequency to the lowest value you expect to find. Check for proper circuit operation (i.e., a stable reading), and change the value of the large capacitor if necessary.

If you think using IC1A is unnecessary, move the two transient protection antiparallel diodes to IC1C inputs, coupling the input signal to the 4k7 resistor. All resistors should be metal film, 1%. ➽

Wayland's Wood World

PUTTING ON THE PRESSURE

By Bob Wayland

The simple, critical operation of clamping your enclosure when you are gluing it is also one of the easiest to mess up. For a strong joint the clamping must squeeze the glue into a thin continuous coating on the adjoining wood surfaces, at the same time expelling any trapped air. Finally, the clamping must hold the pieces together during the setting and curing of the glue. The requirement that we all get hung up on is that of ensuring we have the correct gluing pressure for a specific glue consistency.

The basics of obtaining correct gluing pressure are the subject of this note, I will cover the best glues later. Some general rules should be kept in mind. If you are using a thin, low-viscosity glue, use a lighter joint pressure. (This is also true of glues that become thin during curing.) Hardwoods, particleboard, and medium-density fiberboard (MDF) usually need a more viscous glue and require higher gluing pressures. Of course, in-between pressure is appropriate with glues of intermediate consistency. For the materials that we normally use in making speakers ("Building Materials for Speaker Cabinets," *SB* 3/93), this translates into pressures in the 100–250 psi range for hardwoods with specific gravity from 0.3–0.7, including black cherry, mahogany, sugar maple, red oak, and walnut. For particleboard, MDF, and woods with a specific gravity 0.7, the pressure can be increased up to 300 psi.

As with most things in woodworking, there is a simple basic principle: *Always assure the correct uniform pressure on the glue surfaces.* If you keep this in mind, the best way to clamp will usually be obvious. The two variables you can control are the amount of force applied by a clamp and the geometry of the application of that force. Then just put on the pressure.

HOW MUCH PRESSURE?

Clamps are the usual method for applying force. A previous column (*SB* 1/93) considered different clamps, now we need a bit more information. How much force can you expect to be able to apply with the different types shown in *Photo 1* and *Table 1*?

The maximum depends upon your upper-body strength and whether you use a little mechanical advantage. It is in trying to get the maximum force that one often encounters difficulty, however. Poorly made clamps can

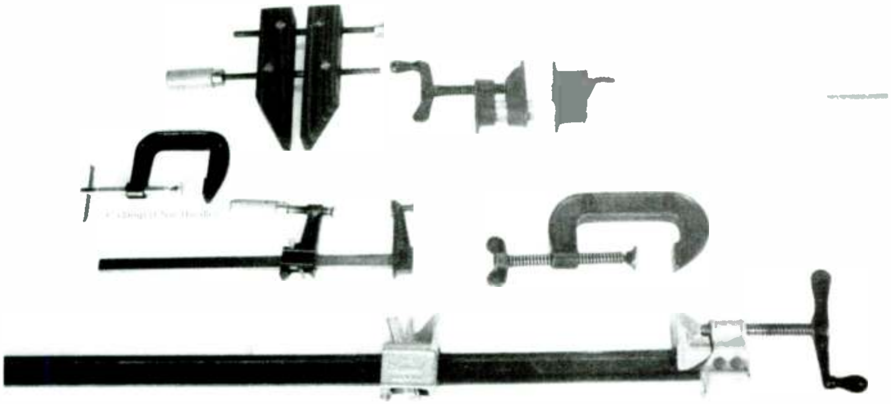


PHOTO 1: Common woodworking clamps.

TABLE 1

POSSIBLE CLAMPING FORCES

Clamp	Possible Force
Steel bar clamp	350–550lbs.
C-clamp (T-bar handle)	≈2,100 lbs.
C-clamp (butterfly crank)	≈1,100 lbs.
Hand screw clamp	≈ 900 lbs.
Steel I-beam clamp	≈2,000 lbs.
Pipe clamp (¾" pipe)	≈1,100 lbs.

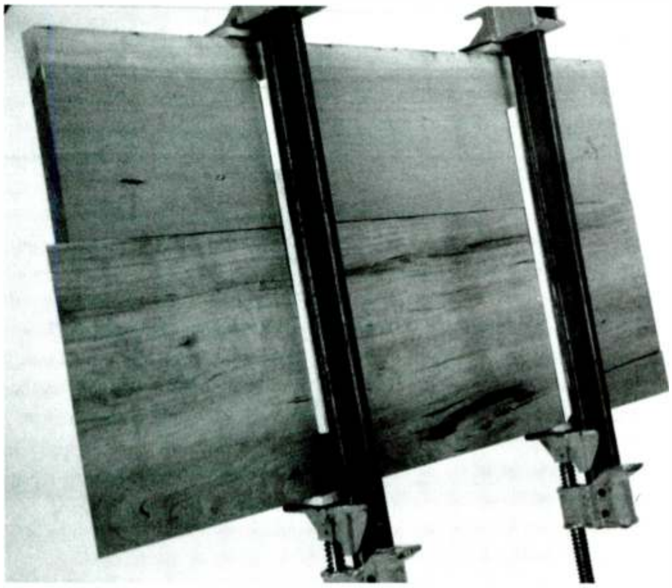


PHOTO 2: Spacers used to align the applied clamping force.

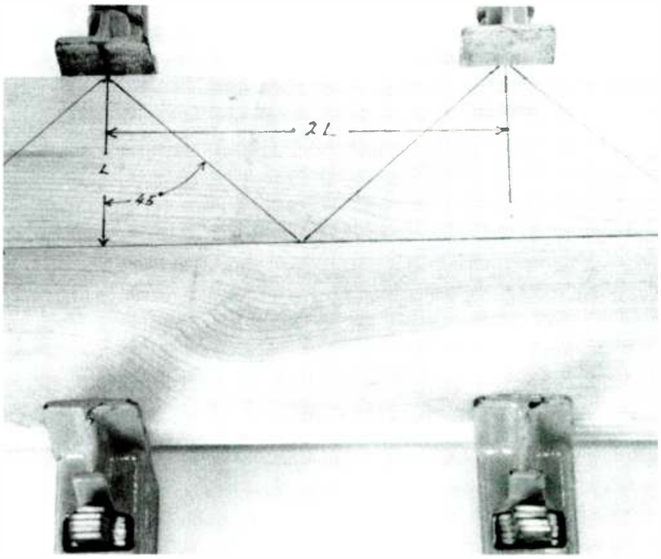


PHOTO 3: Use of a spreader to provide uniform pressure on a glue joint.

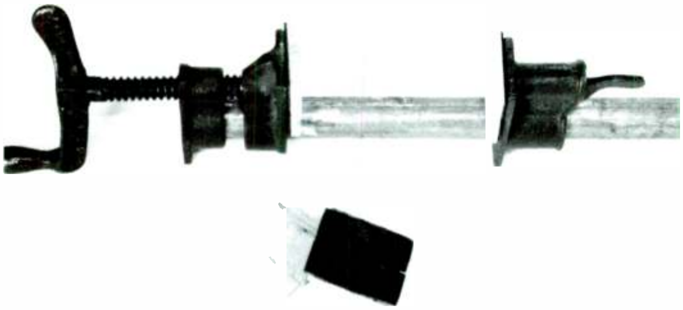


PHOTO 4: Clamping pads with magnetic holders.

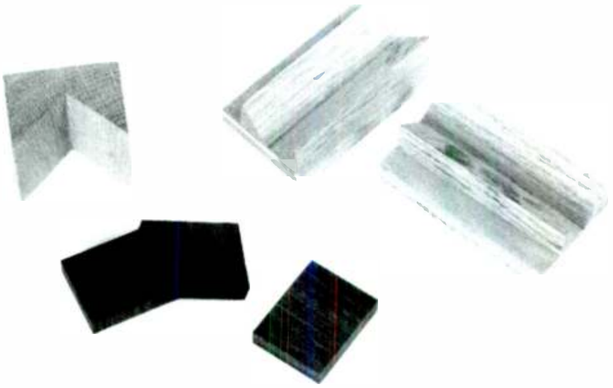


PHOTO 5: Corner clamping pads.

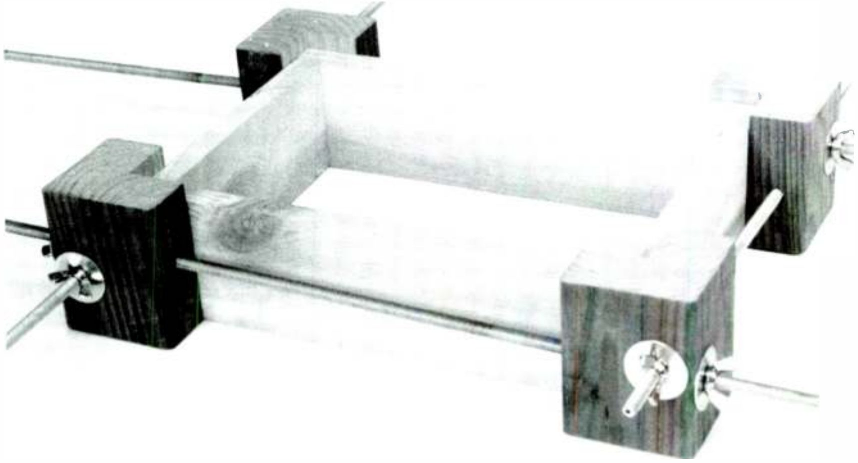


PHOTO 6: Using the corner clamping assembly.

strip threads; bars can permanently distort; pressure plates can break. The problem is the variety of cheap, inadequate clamps out there: clearly, caveat emptor!

Oft-encountered problems in clamping are insufficient pressure and incorrectly applied force. When you apply force be sure the force is perpendicular to the glue surface. Usually, when using bar, pipe, and I-beam clamps, this means placing a spacer to elevate the boards above the bar or pipe such that the axis of the drive screw is in line with the boards being glued (*Photo 2*). Also, if the body of the clamp isn't perpendicular with the glue line, the two pieces being glued will slip. Most of the time we don't want this. If what you are gluing isn't aligned properly, this off-perpendicular application of clamps can be used to force the pieces back into alignment.

The problem of getting enough pressure on the glue line is one of geometry and bonehead physics. If you are gluing up a corner joint and apply a clamp directly to the boards, the resulting uneven pressure will probably cause poor bonding, with glue-starved areas next to ones where insufficient pressure is applied. This also happens when clamping a solid wood strip to the edge of particleboard or MDF. The secret is to use an additional scrap board as a spreader (*Photo 3*), which spreads out the pressure generated by the clamping point of contact. Just how the pressure is spread depends upon the stiffness and the grain patterns within the spreading board.

Luckily we don't need even a 10% answer; a factor of two is good enough. You can get this by assuming that the force is spread out

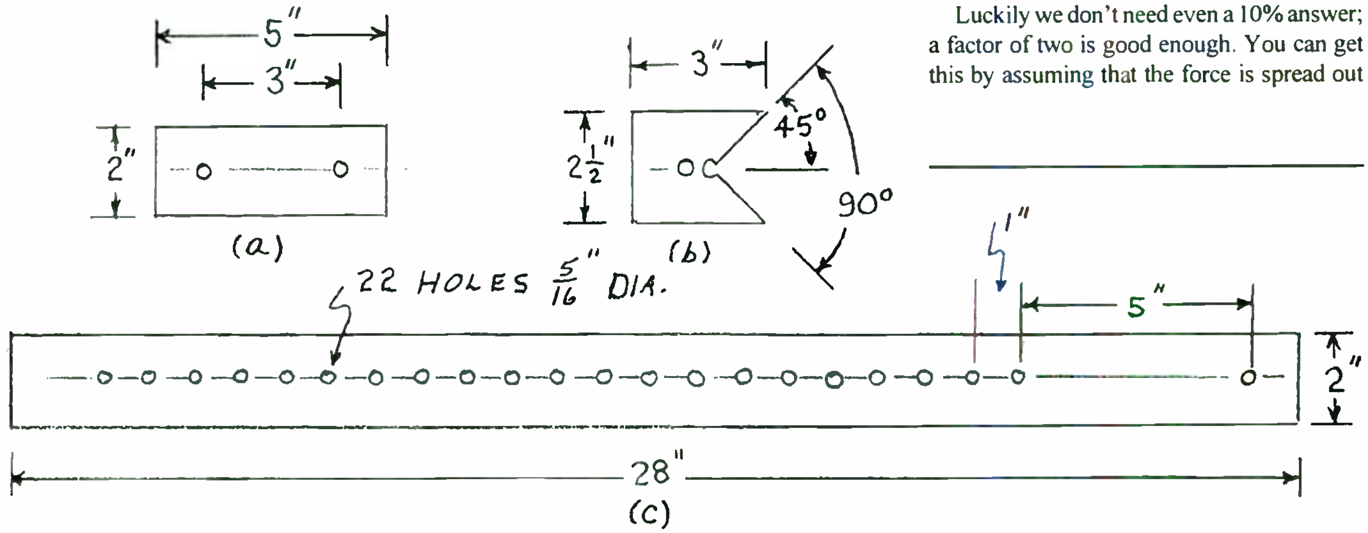


FIGURE 1: Construction details for grille frame clamp. You need to make two each of (a) and four each of (b) and (c). Use 6/4 hardwood if you have it for (a) and (c), but 4/4 will do. It is best to use 8/4 hardwood for (b). All holes are 5⁄16" diameter. Bolt together using 5⁄16" bolts using wing nuts and washers for the corner pads (b). The 22 holes on 1" spacing for (c) should be countersunk for bolt heads on the underneath side. The clamp should be assembled and used as shown in *Photo 8*.

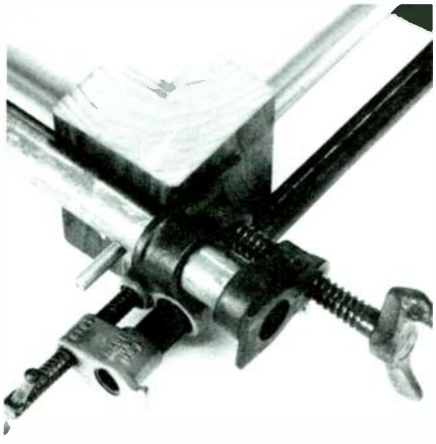


PHOTO 7: Using pipe clamps to increase clamping pressure on the corner clamping assembly.

at 45° from the clamp (*Photo 2*). Because we are dealing with a 45° triangle, the pressure is just twice the width of the board times the board thickness divided into the clamping force. The spacing of the clamps is about twice the width of the spreader board. Keep these ideas in mind when clamping and you will move a long way toward meeting the simple principle above.

JIGS AND EFFECTS

The normal system of clamps (i.e., those shown in *Photo 1*) can be made much more useful with the addition of a few homemade jigs. One of the easiest to make is pads for the pressure plates of the clamps. Normally you will place a pad of wood between the clamp and the boards being clamped to protect the latter. Unfortunately, the pads are always falling out. Most stationery or hardware stores sell magnetic strips which you can cut to shape and glue to the backs of the pads (*Photo 4*), holding them in place while you set up the clamps.

Another simply made pad helps in applying diagonal pressure, e.g., when you must straighten an out-of-square enclosure box. Start with a square cross-sectioned block and cut a 90° notch at 45° to one face, or glue 45° corner braces onto a scrap pad of wood to form the 90° notch (*Photo 5*). Just put one pad on each of the opposing corners and the clamp on the back surfaces opposite the 90° notches. Yes, I know that if the box is not square, things are off a bit, but you are only using this device to make small corrections.

An extension of the corner pads makes for a useful enclosure-clamping jig. Many of my woodworking friends consider this overkill, but it is the only technique I really trust when making a special enclosure. The idea is to use 4″ × 4″ wooden posts as a combination corner and load-spreading clamp. First cut four sections of the post as long as, or longer than, the depth of the enclosure. From one corner of each post, cut out one-fourth. Drill a series of 7/16″ holes in each pad. Through these holes place 3/8″ threaded rods and secure the pads in place using fender washers (#4, 5/16″ I.D., 1 5/8″ O.D.) and 3/8″ wing nuts (*Photo 6*).

To keep any squeezed-out glue from attaching the pads to the enclosure, either put waxed paper between the pads and the enclosure or soap the surface of the pads. (If you use soap, be sure to wipe off any excess before use.) Selectively tighten the wing nuts until the enclosure is square and firmly clamped. This will not provide enough clamping pressure, so you will need to use additional clamps; e.g., steel I-beam clamps. Using the corner pads as spreaders, you can readily apply the necessary pressures (*Photo 7*).

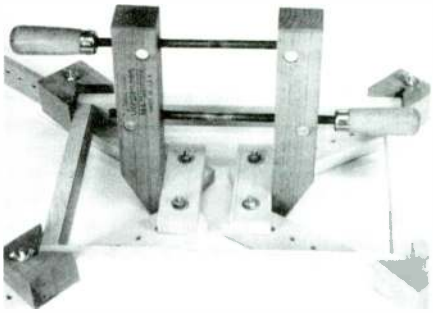
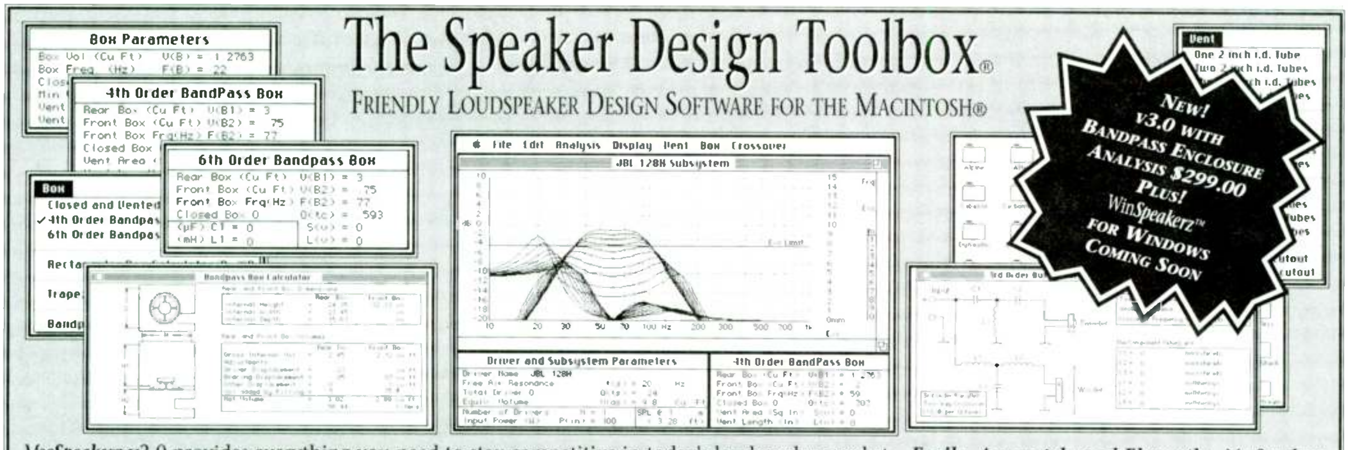


PHOTO 8: Using the grille frame clamp.

Tighten the clamps in a sequential manner to avoid creating unequal tensions that might distort the shape of your enclosure. It is a good idea to use the methods discussed in *SB* 6/93 to ensure that the enclosure remains square (straight). You can use this same approach even if your enclosure is not rectangular and has a number of unusual angles, designing and cutting the corner pads to accommodate your special shape.

You can make a simple jig for clamping the often fragile grille-cloth frame. It is not something that I came up with but is an idea that has been around for a long time. I believe it was originally designed to aid in making picture frames. If you decide to make one, you will have a dual-use jig. *Figure 1* illustrates the construction details. You can make the legs longer, if needed. One of the jig's advantages is its self-aligning capability; if you have cut the frame members accurately

Continued on page 72



Tools, Tips & Techniques

VENTED BOX PARAMETERS

Several years ago, D.B. Keele, Jr. gave us a couple of equations that made it easy to design vented loudspeaker systems with the aid of a hand calculator. The first equation came up with the specs for an optimum box, the optimum in this case being the flattest possible frequency response. The equations require us to input three speaker parameters: f_{SA}, Q_{TS}, and V_{AS}. (Keele's equations are based upon variations of the Thiele/Small alignments.)

All is well if the results of this equation satisfy the designer, but in some cases the box size or the frequency response of this alignment will not be satisfactory. Anticipating this, Keele presented another equation based upon the selection of the box size (V_B) and proceeding from there. The designer could then pick out a box size that he or she deemed an acceptable compromise, and then calculate the ripple magnitude at the bottom of the frequency response to find out if the box is usable.

We can save a lot of time if the question of the magnitude of the peak or dip of the response could be considered first and a maximum (of ±dB) decided upon. These limits would indicate the largest and smallest boxes that would have an acceptable ripple. The upper and lower limits of f_3 would also be known. Everything inside the limits would compose a potentially feasible system, and it would be clear early on if the woofer at hand was suitable for a vented design.

If Keele's second set of equations is worked backward we come up with the desired results:

$$V_{AS}/V_B = \text{Antilog}_{10}\,[(R/20)/(Q_{TS}/2.6)]^{(\text{INV}).35}$$

where R = ripple magnitude in ±dB (peak or dip)

Antilog is achieved (at least on my Texas Instruments calculator) by punching the INV key followed by the LOG key and waiting for the answer. Calculate (INV).35 by punching INV, Y^X, .35 = and waiting for the answer.

V_{AS}/V_B is the ratio of the woofer to the enclosure. The net size of the box may be found by dividing $7V_{AS}$ by V_{AS}/V_B. You can use liters or ft.^{3} or in.^{3}.

Also use V_{AS}/V_B to find f_3 and f_B:

$$f_3 = (V_{AS}/V_B)^{.5} \times f_{SA}$$

This is just another way of saying that $f_3 = \sqrt{V_{AS}/V_B} \times f_{SA}$.

$$f_B = (V_{AS}/V_B)^{.32} \times f_{SA}$$

This exponent is achieved by first punching the Y^X key and then .32 =. f_{SA} is the free air resonance of the woofer.

For a walk-through demonstration I picked up a Carbonneau catalog and chose a 10″ speaker that looked appropriate—the model CG-258R. f_{SA} is 26Hz (±15%—we'll use the nominal figure), Q_{TS} is .414, and V_{AS} is 6.19 ft.^{3} To be practical, let's pick +1dB and −1dB for the upper and lower ripple limits. We'll do the upper one first:

Divide +1 by 20 (0.05). Press the INV key and then the LOG key and wait for the answer (1.1220185). Divide this by .414 (this is Q_{TS}, 2.7101895). Divide again by 2.6 (1.0423806). Press the INV key, then the Y^X key, then .35 =. After a short pause the answer (1.1259102) will appear; this is V_{AS}/V_B. (Actually this is the V_{AS}/V_B to achieve a box that will yield a +1dB ripple.) All other conditions will have a different V_{AS}/V_B.

To find the net V_B for this setup, divide 6.19 (V_{AS}) by 1.1259102 (our V_{AS}/V_B of the moment). We get a box with a net volume of 5.4977742 (say 5.5) ft.^{3}

To arrive at f_3 (which is the half power, −3dB point), take the square root of V_{AS}/V_B (1.0610892) and multiply it by f_{SA} (26). This is about 27.6Hz.

For f_B (the frequency the box is tuned to via the duct) again take V_{AS}/V_B, punch the Y^X key, then .32 = and wait (1.0386786). Again multiply this by 26 (f_{SA}) and get 27Hz. Since we went to so much trouble, I'll give you the dimensions of a duct that will probably come close to realizing the speaker we worked out together. A duct with an inside diameter of 3″ and a length of about 2¼″ would probably work. A duct with an inside diameter of 4½″ and a length of 6″ would also work, and because of the low frequency involved would probably be a better choice. The large diameter would give lower wind velocity.

The −dB ripple is found in the same way. You can work it out from the above using the different figure. I'll give the calculator readouts for each step: −1/20 = −.05. (To get −1, punch 1, then ± key.) The antilog_{10} is .8912509, which divided by Q_{TS} is 2.15278. That divided by 2.6 = .8279923. At this point punch the INV key, the Y^X key, and .35 = and wait again until you see .5831623. This is the V_{AS}/V_B for the −1dB ripple.

The square root of V_{AS}/V_B, times 26 is 19.8 (say 20) Hz. This is f_3.

$(V_{AS}/V_B)^{.32}$ = .8414971. That times 26 = 21.9 (say 22) Hz. This is f_B.

Anything between those two extremes should be a viable speaker with <±dB peak or dip. Note: V_B is 10.6 ft.^{3} (It's big, but it goes to 20Hz.) A duct with an inside diameter of 4½″ and a length of about 4⅜″, or 6″ and 8¾″, respectively, would work.

John Cockroft
Sunnyvale, CA 94087

Craftsman's Corner

HEAD SPEAKER

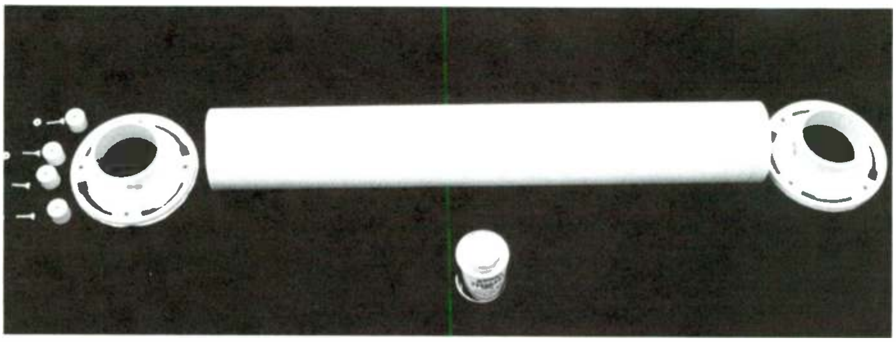


PHOTO 1: The Simpline parts.

OK, you asked for it—here is a photo of my construction of "The Simpline," by John Cockroft (*SB* 2/93, p.14). His cabinet construction was too difficult for me (my most sophisticated machining tool is a rusty hacksaw). Instead, a trip to Home Depot produced an almost-ready-made box, cut to length. The tube is 4″ PVC pipe ($9.42), schedule 40 (one piece of pipe, 10′, will build four speakers). The top and bottom pieces are, ahem, 3–4″ closet flanges ($2.11); the feet are 1½″ caps ($0.21). Stuffing is from an old pillow. Pretty cheap and quick, yes?

I modified the Radio Shack driver per the article. I didn't lean it against a wall, the feet hold it off the floor. I tried different spacings but it didn't seem to matter and the feet are in keeping with my aesthetic theme.

All in all, it's reasonable for what it is—the perfect speaker for the bathroom. Mr. Cockroft is right about the amount of bass from such a small driver, but it is muddy. It was an interesting one-afternoon project.

W. Werner
Severn, MD 21144

P.S. Note the clever use of color photography. The speaker is white, the background black.

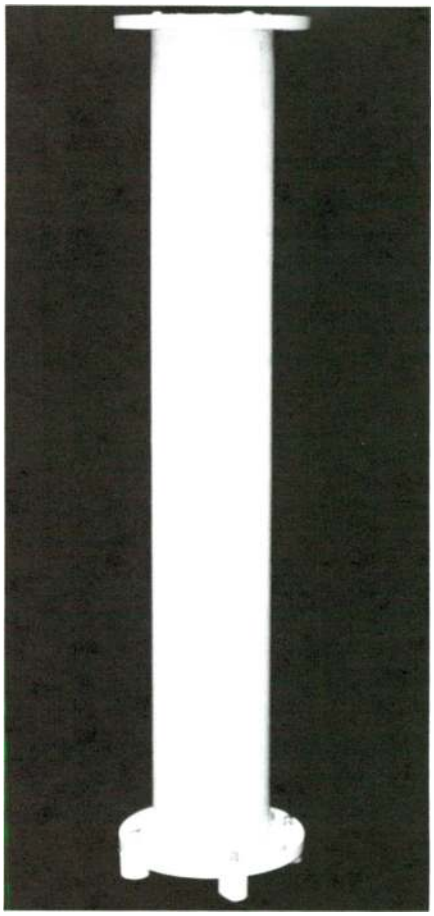


PHOTO 2: The Simpline complete.

SB *Mailbox*

PHOTO SWAP

In Wayland's Wood World, *SB* 6/93 ("Let's Get This Straight," p. 50), *Photo 5* and *Photo 6* were transposed. *Photo 5* depicts a saw blade check on a piece of 2 × 2 stock; *Photo 6* is a 2 × 4 crosscut with a 0.1° error. We apologize for any confusion this may have caused.

INCORRECT FORMULAS

My letter in *SB* 6/93 ("Accurate Measurements," p. 57) included some formulas which contained lower-case L's as well as l's, which led to some confusion. The correct formulas are:

$$Q_{MS} = F_S \times \frac{R_O}{(F_2 - F_1)}$$

(as printed), and:

$$V_{AS} = V_{BT} \times \frac{(F_H^2 - F_M^2) \times (F_M^2 - F_L^2)}{(F_H^2 - F_L^2)}$$

$$F_{SB} = \frac{(F_L \times F_H)}{F_M}$$

$$\text{Adjustment Factor} = \frac{(F_S \times F_M)}{(F_H \times F_L)}$$

$$H_A = \frac{F_M^2}{(F_L \times F_H)}$$

$$\alpha_A = \frac{(F_H^2 - F_M^2) \times (F_M^2 - F_L^2)}{(F_H^2 \times F_L^2)}$$

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THE IMP REVISITED

I recently saw a demonstration of Bill Waslo's IMP audio analyzer ("The IMP," Part I, *SB* 1/93, p. 10; Part II, *SB* 2/93, p. 30; Part III, *SB* 3/93, p. 36; "The IMP: Measuring T/S Parameters," *SB* 4/93, p. 38; "The IMP Goes MLS," *SB* 6/93, p. 40) at a meeting of the Cape Town Hi-Fi Club. I did years of research on audio measurement using impulse noise as a source and wrote a paper on the subject, published in 1991 by the South African Institute of Electrical Engineers. The IMP system is impressive. Mr. Waslo has done a wonderful job, and my comments are intended as constructive.

The source signal is important. Mr. Waslo uses a rectangular unit pulse, which is calculated to have its first zero in the frequency domain around 25kHz. The main problem with this pulse is the presence of the high-frequency lobes that continue out to a theoretical infinity. While this may not be a problem in the loudspeaker itself, electrical networks will definitely suffer from the presence of the high-frequency energy. Furthermore, the sharpness of the pulse will have a tendency to shock the system, producing adverse effects (*Figs. 1* and *2*).

Figure 1 is the unit pulse as observed on IMP along with the FFT transform. *Figure 2* shows a wider pulse, which has the effect of reducing the first zero in order that the high-frequency lobes can be readily observed (the vertical scale being adjusted for ease of viewing). Practically, with the pulse Mr. Waslo actually uses, these higher-frequency components can be a number of megahertz.

As the pulse gets narrower the first zero will move out along with lobes, and the frequency over the range 0–20kHz will flatten. In the limit, as the width of the pulse tends to zero, the frequency response will be flat to infinity. This can easily be proved mathematically. The Fourier transform of the general waveform $g_{(\omega)}$ can be shown by the equation:

$$g(\omega) = \int_{-\infty}^{\infty} f(t) \cdot \exp(-j\omega t)\, dt \qquad (1)$$

The inverse of this transform is then:

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} g(\omega) \cdot \exp(j\omega t)\, d\omega \qquad (2)$$

For the unit pulse, see *Fig. 3*. The single element (t) has an amplitude A and extends from –t/2 to +t/2. Applying the Fourier transform, (t) is 0 outside the above limits of (t). Then

$$g(\omega) = \int_{-t/2}^{t/2} A \cdot \exp(-j\omega t)\, dt \qquad (3)$$

$$= A \cdot t \frac{\sin \omega t/2}{\omega t/2}$$

If the pulse has an area of unity, i.e., A · t = 1, Then

$$g(\omega) = \frac{\sin \omega t/2}{\omega t/2}$$

or

$$g(\omega) = \frac{\sin x}{x} \text{ where} \qquad (4)$$

$$x = \omega t/2$$

This, then, is the unit pulse. The value at x = 0 is unity; the first zero occurs at π and

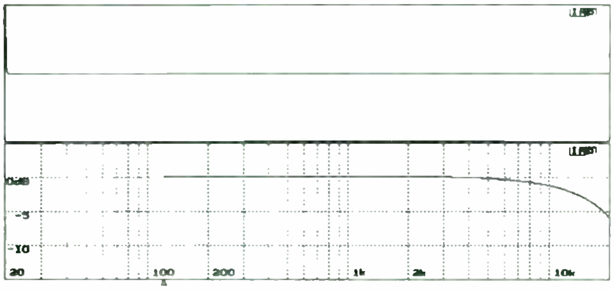

FIGURE 1: Unit pulse as observed on IMP with FFT transform.

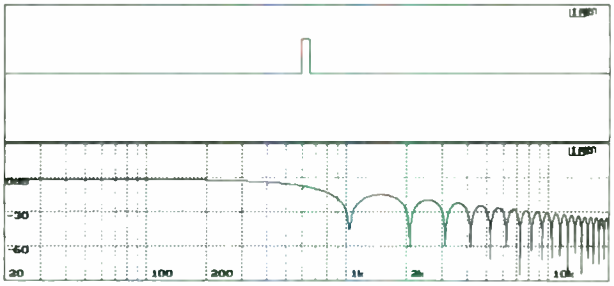

FIGURE 2: A wider unit pulse.

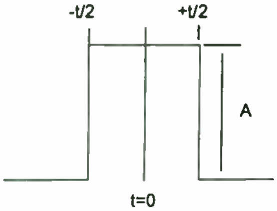

FIGURE 3: Unit pulse. As width of the pulse tends to zero, frequency response will be flat to infinity.

subsequent zeros at multiples of π. If t is reduced and A is increased, to maintain the unity of the pulse, the first zero in the spectrum will move out. In the limit as t tends to zero and A tends to infinity,

$$g(\omega) = 1 \qquad (5)$$

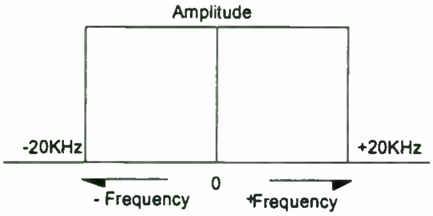

FIGURE 4: Idealized frequency spectrum.

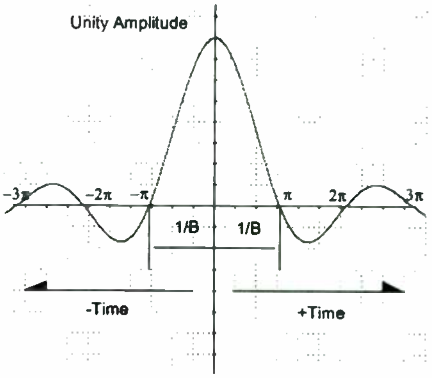

FIGURE 5: Theoretical time-domain pulse.

and the amplitude and phase response will be flat to infinity.

If we know the frequency spectrum we can obviously calculate the time-domain function. *Figure 4* shows an idealized frequency spectrum. It ranges over the bandwidth B, which is twice the desired frequency range, the negative domain being required for the transform. The limits are therefore from –20kHz = –ω0/2 to 20kHz = ω0/2.

From the Fourier transform of the general waveform,

$$f(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} g(\omega) \cdot \exp(j\omega t)\, d\omega \qquad (6)$$

The limits being reduced from "–∞" to –ω0/22" and "∞" to "ω0/2" and gω being unity or flat between the limits,
Then

$$f(t) = \frac{1}{2\pi} \int_{-\omega 0/2}^{\omega 0/2} \exp(j\omega t) \cdot d\omega \qquad (7)$$

where

$$-\frac{\omega 0}{2} = -\frac{2\pi B}{2} \text{ and } \frac{\omega 0}{2} = \frac{2\pi B}{2}$$

$$f(t) = \frac{1}{2\pi} \int_{-\pi B}^{\pi B} \exp(j\omega t) \cdot d\,\omega$$

$$= \frac{\sin \pi Bt}{\pi Bt} = \frac{B \sin x}{x}$$

If B is taken as unity height of the pulse then:

$$f(t) = \frac{\sin x}{x} \qquad (8)$$

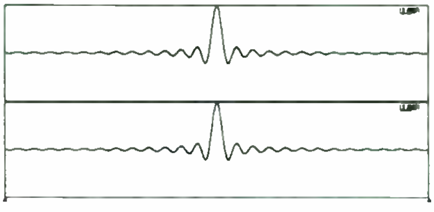

FIGURE 6: Display of sin x/x pulse on IMP screen.

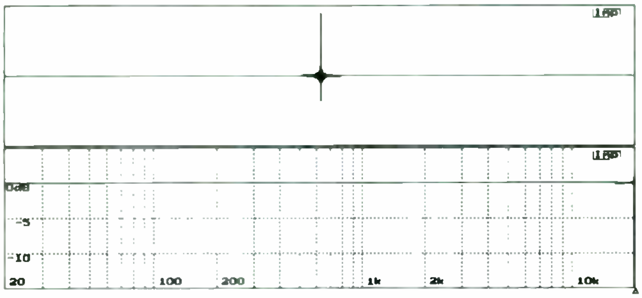

FIGURE 7: A sin x/x pulse producing the frequency-domain bandwidth flat to 20kHz.

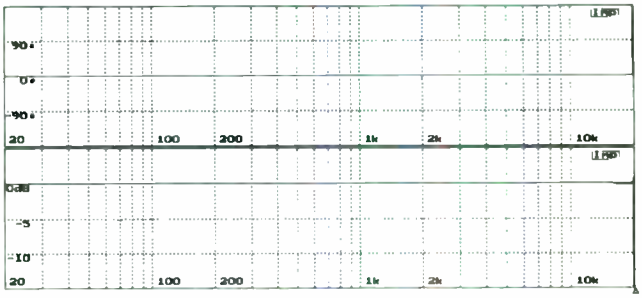

FIGURE 8: Phase and frequency response.

where

$$x = \pi Bt = \frac{\pi \cdot t}{25t} = \frac{\pi}{25}$$

The equation of the pulse is then given by:

$$y = \frac{\sin \pi/25}{\pi/25} \text{ for values of t} \qquad (9)$$

This, then, is the perfect time-domain pulse. It has a peak amplitude B of unity centered on t = 0. The first zeros are at π and –π. The pulse width between the first zeros is 2/B and represents the required spectrum in the positive domain. In this case:

$$\frac{2}{B} = \frac{2}{40 \times 10^3} = 50\,\mu\text{sec} = 20\text{kHz} \qquad (10)$$

Figure 5 shows the theoretical pulse. The sin x/x pulse has its peak amplitude on zero seconds and the measurement starts (in the case of a size of 4,096) at –2,048–+2,048. *Figure 6* shows the display of the sin x/x pulse on the IMP screen. If this pulse is transformed the resulting frequency span will be about 400Hz, but the energy is very high. A more practical example (*Fig. 7*) depicts a sin x/x pulse producing the frequency-domain bandwidth flat to 20kHz. *Figure 8* shows the phase and frequency response. *Figure 9* illustrates the case for a 10kHz pulse showing the phase and frequency response. In all these cases the pulses were synthesized using a simple QBASIC program.

```
CLS
OPEN "SINX.DAT FOR OUTPUT AS #1
FOR X = -2047 TO -0.999999
Y = SIN(X) / (X)
PRINT #1, USING "+#.######"; Y
NEXT
FOR X = 0.999999 TO 2047
Y = SIN(X) / X
PRINT #1, USING "+#.######"; Y
NEXT
CLOSE
```

The file "Sinx.dat" can now be edited by inserting the four-line header information. The value of Y must be inserted at time = 0, as the program is unable to calculate this (it can't divide by zero) at the point where x = 0. For the above case Y = +1.000000.

You can play around with multiples of X to get different frequency spans. In a practical system each desired frequency span should have its own sin x/x pulse. *Figure 10* illustrates further reasons for using this pulse.

The figure shows the IFFT of the unit pulse; the upper trace being an expansion of the left side of the lower trace. This is, of course, the sin x/x pulse produced by the inverse transform. The left of the lower screen shows the positive time and the right the negative. If this is now FFTed it will produce an erroneous result, because of the mixup in the time domain signal. This cannot happen if the source signal is the sin x/x pulse, as this is the pure pulse.

As both the negative and positive halves of the pulse are required for the transform, the pulse belongs in the center of the time domain. Furthermore, the transform requires the whole of the pulse for the mathematical calculation in the FFT. Cutting out portions of the signal and FFTing will produce a distorted result. This is very important around time = 0, because most of the information exists around this critical area. Mr. Waslo's waterfall maps cannot work because this critical area is gradually removed and transformed. Instead, the waterfall shows a gradual distortion of the signal that has nothing to do with the loudspeaker or room acoustics. When transforming a signal it is better to use the 1/? octave, averaging to smooth out the result.

Waterfalls or spectral maps are produced by transmitting a single pulse and taking a number of separate measurements while the sound produced by the original pulse dies away. Reverberation time measurements can be done in this way to calculate RT 60.

Because of the high accuracy of the sin x/x pulse producing a frequency response over the desired range within 0.1dB, the system can be useful in testing amplifiers, audio transmission chains, and so on, as well as for aligning tape machines and comparing tape brands. Experimentation with the system will uncover many uses.

Some possible features for a "deluxe" IMP system:

- A sin x/x pulse for very high accuracy of results for acoustic and electrical measurements.
- A time-domain display with graticule for oscilloscope-type measurements, which can be switched to a single full-screen display. The screen can be scrolled both horizontally and vertically. Normal simple oscilloscope-type controls are provided for vertical and horizontal measurements, and crosshatching on screen markers makes measurement easy.
- Harmonic distortion in the frequency domain can be calculated for decibels and percentages by placing crosshatching markers over the peak of the fundamental and calculating. A handy facility is a marker to peak command.
- The frequency-domain display with on-screen crosshatching markers can be switched to single full-screen display scrollable both horizontally and vertically. Gain and range facilities are provided.
- Source and input triggering are provided with trigger delaying.
- Multiple buffers will enable construction of true waterfall maps and the capture of long-signal transients. RT60 measurements can be carried out by the same method.
- The 1/? Octave facility provided enables sound measurements and graphic equalizer tweaking of loudspeakers.
- The built-in calibration facility enables sound-level measurements.

Mr. Waslo has gone further than anyone to produce an affordable testing system, and I congratulate him. It is a pity that he didn't go all the way, because he clearly has the ability to do so.

Philip P.N. Thompson
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Bill Waslo responds:

I'm pleased to hear of the Cape Town club's demonstration of my IMP system, and appreciate Mr. Thompson's kind comments about the IMP. I can't say I'm in agreement, though, about the sin x/x being the ideal test source signal. As Mr. Thompson correctly points out, the ideal mathematical square pulse has frequency content extending out to theoretical infinity; so he must be aware that the sin x/x time function consists of oscillations (nonconstant values) that cover all of eternity! In order to get that abrupt cutoff of high-frequency energy (absolutely), the measurement can never be finished.

In fact, the frequency spectrum of an ideal rectangular pulse, which Mr. Thompson sees as a problem, is of the same shape (x being frequency) as the time-domain function of the sin x/x he proposes (with x being time). This fact may not be immediately obvious in his IMP FFT plot of the widened pulses, because IMP shows frequency-domain data in decibel (log magnitude) format over a log format frequency scale.

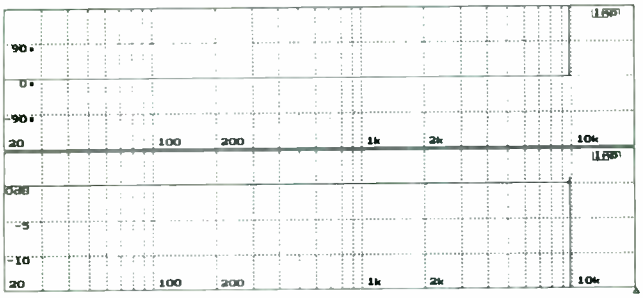

FIGURE 9: Phase and frequency response of a 10kHz pulse.

FIGURE 10: IFFT of sin x/x pulse produced by inverse transform.

It is a basic dual principle of time/frequency domain theory that a limited time signal in the time domain possesses frequency content to infinite frequency and that a limited frequency signal in the frequency domain corresponds to content over infinite time. For less absolute, real-world use, of course, we can limit the measurement time at some point, with less ideal, but usable, results; in similar real-world use, we can also limit the spectrum of the test pulse, quite a bit more easily (and often it can't be avoided), by nearly trivial low-pass filtering.

Mr. Thompson seems to object primarily to the energy bandwidth and "sharpness" of the rectangular pulse. The energy bandwidth could, admittedly, have been better dealt with on the board. IMP exists primarily to allow loudspeaker measurements at the absolute lowest possible cost; I had assumed that any competently made power amplifier used with it would have basic RC filtering at its input to prevent misbehavior due to RFI, rapid rise times, or any out-of-band signal with which the amp might not be able to contend.

In any case, more advanced users could easily add an external low-pass to band-limit the test signal; IMP's use of a cal-probe input to reference the input spectrum that is applied to a unit-under-test means that response variations of the test signal, such as a low-pass filter's slow rolloff, will not affect the measurement (the final response is the output spectrum divided by the input spectrum).

Contributing Editor G.R. Koonce convinced me not to assume that all amplifiers would incorporate such input limiting, so we issued a suggested modification (SB 3/93 Mailbox, p. 66), adding a rolloff capacitor across the test pulse output jack; this limits the input slew rate to the amplifier and provides first-order band-limiting. The input to IMP already incorporates rudimentary but quite effective third-order anti-aliasing low-pass filtering. I have used IMP quite successfully to measure active parametric equalizers and filters; they have not suffered from the effects of the pulse.

I see no advantage to using a sin x/x as compared to a simple causally band-limited pulse output. The sin x/x is merely what would result if an impulse were passed through an ideal brick-wall linear-phase low-pass filter; it rings forever, and the near total lack of energy at the high end of the band greatly complicates any attempt to eliminate amplifier effects by "cal"ing (can't divide by zero).

The flat response and linear phase might be an advantage if no reference cal spectrum were taken, but then amplifier and smoothing filter effects could not be removed, and accuracy would suffer. It is far more accurate, and expedient, to measure what a test signal is, and correct for that, than to go to extreme means to make it approach some sort of ideal. A pulse band-limited by a linear filter will also never quite get back to zero (in theory), but it sure gets close a lot faster than a pulse that has hit a ringy brick-wall filter.

Mr. Thompson fears that the sharpness of the pulse will have a tendency to shock a system, producing adverse effects. Slew-rate-induced effects are easily dealt with by simple RC rolloffs or other band-limiting, and I know of no other effects that might be considered adverse. If the system under test rings when it is shocked, that is its impulse response, which is directly convertible to the frequency response—that's what we want. Unless the response is nonlinear (i.e, changes when level is reduced), the shock response is the information desired and not a problem.

The band-limited pulse is causal—it starts at the beginning and doesn't waste acquisition samples. It is easily generated by a monostable flip-flop and analog linear filter (the sin x/x requires a D/A), and on well-behaved speakers it stimulates a damped output, making possible the removal of echo effects for quasi-anechoic measurements (the main purpose of IMP), which would be quite impossible using the sin x/x.

The sin x/x shape Mr. Thompson derived by IFFTing the IMP frequency response of a rectangular test pulse was actually caused not by any flaws in the pulse, but by a frequency-domain "window" function that is applied in IMP version 1.11 through 1.22 to all frequency-domain data before any IFFT operation (in effect, a zero-phase low-pass filter). The window is used to minimize noise effects that can result near the Nyquist frequency at 30.7kHz (which is also the first pulse response zero) when a cal is performed; these effects happen due to the low cal-response energy near the zero—the resolution-limited effects of dividing some very small numbers by each other at the supersonic frequencies. If this window were not applied, you would get the usual pulse back, as you must.

After all, the FFT and the pure IFFT are exactly convertible, aren't they? (In the IMP/M 2.0 version, the window is applied only if the cal operation is performed. If you transform a pulse, then IFFT it without

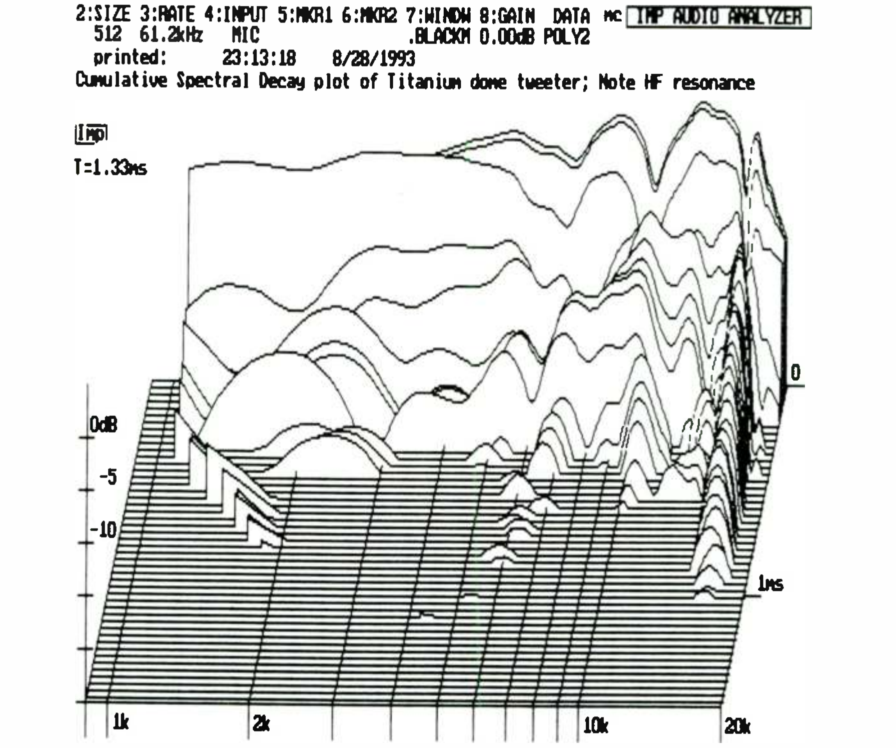
```
2:SIZE 3:RATE 4:INPUT 5:MKR1 6:MKR2 7:WINDW 8:GAIN  DATA MC IMP AUDIO ANALYZER
  512  61.2kHz   MIC                   .BLACKM 0.00dB POLY2
  printed:      23:13:18   8/28/1993
Cumulative Spectral Decay plot of Titanium dome tweeter; Note HF resonance
```

FIGURE 1: Cumulative spectral decay plot of Titanium dome tweeter. The CSD waterfall is useful in highlighting delayed or resonant behavior in drivers.

normalizing it via a cal response, you get it back with no ripples). Also, note that the FFT (as opposed to the theoretical Fourier transform) operates on periodic data only, and treats all time data as periodic: hence the wiggles at the right side of the screen in Mr. Thompson's IFFT.

If you FFT this data correctly (it is sin x/x-like, so you can't truncate as you can a pulse) by including all the time data (in his Fig. 10, set the right time marker, MKR2, to 512), you do not produce an erroneous result, other than that due to the window IMP applied. The erroneous result he reports is, in fact, not avoidable by but precisely because of the sin x/x shape of the data, which covers all time and can't be approximated by the time-truncated sample.

Mr. Thompson and I evidently refer to different things when we talk of waterfall plots. The waterfalls in IMP (and in DRA's MLSSA and other speaker analysis systems) are more properly referred to as cumulative spectral decay plots or cumulative decay spectrograms. (Figure 1 is an example of this type of plot.) They work—and quite well.

A true frequency response is a measure of the Fourier transform of an impulse system for all time; any attempt at representing "frequency response versus time" is but an approximation, though it can be quite a useful one. The curve at the back of the plot is the frequency content of the entire impulse response (or as much as was included in the measurement). Successive curves show the frequency content of successively later fractions of the impulse response, which indicate the contributions to the cumulative plot that are made later in time—in other words, because of group delay and ringing, just as would occur if a tone burst were applied at each frequency.

Of course there is some spectral leakage from cutting into the initial part of the time response, and the significance of the later curves becomes questionable at the lower frequencies because of the shorter valid time data included (IMP abruptly cuts off the curve at the low end as the waterfall progresses, to avoid presenting meaningless data). But we can't talk about pure frequency response as a function of time; this plot is an extremely useful invention (not mine) that allows for easy identification of resonances and reflection problems.

I am not an acoustical engineer by trade, but as I understand it, RT60 doesn't consider frequency content at all; isn't RT60 measured via the time-energy curve? While I suppose it could be derived at least qualitatively via a waterfall, I'm afraid I don't grasp what Mr. Thompson means by "taking a number of separate measurements while the sound produced by a pulse dies away." If he is measuring pressure over time, that is just the IMP time response, from which a TEC curve could be made; which is not any kind of spectral map. If he is measuring frequency energy, he must be measuring it through filters to isolate the frequencies (or else doing windowed short-time FFTs of sections of it).

Which brings us back to truncation and spectral leakage, or the decay time of the filters. I have made such plots in communications work, but they seem to be of little use in speaker research; the time periods involved must be considerable compared to the period of the frequencies being measured. The waterfalls in IMP are not meant to be "frequency response versus time"; they are meant to indicate the time source of the cumulative contribution to the final response. And the plot shown does in fact have a great deal to

REFERENCES

1. Bergman, J.M. and L.R. Fincham, "The Application of Digital Techniques to the Measurement of Loudspeakers," JAES (Vol. 25, 6/77): 377-379.

2. Janse, C.P. and A.J.M. Kaiser, "Time-Frequency Distributions of Loudspeakers: The Application of Wigner Distribution," JAES (Vol. 31 4/83): 198-223.

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Reader Service #4

do with the loudspeaker—as a waterfall of a driver that has a ringy cone (like many dome tweeters) demonstrates.

A Deluxe IMP is in preparation, but the change made is for improved noise immunity. The MLS option (new software, add-in board) uses a maximum length sequence (pseudorandom noise, basically a periodic sequence of positive and negative pulses) as the stimulus; a limited version of what is done in the DRA Labs' MLSSA system. I did this to overcome the major flaw of the rectangular pulse stimulus, which is its low energy content and high crest factor (the sin x/x is little better in this regard). An MLS measurement provides rejection of noncoherent noise and distortion and can be band-limited as easily as the rectangular pulse.

Unfortunately, the memory size on the IMP board is not so easily extended, so longer signal transients (as might be encountered in large halls) still cannot be accommodated. Actual source triggering would be a big plus, of course; but it would require an entire hardware redesign, greatly increasing the cost and complexity. The existing IMP accommodates sound-pressure-level sensitivity measurements with some additional effort (as I shall describe in an upcoming SB article, along with the MLS upgrade).

As Mr. Thompson correctly points out, overlaid screens are nearly essential. The new 2.0 software accommodates them, as it does on-trace numbered markers (HP style, not crosshatching). IMP can currently make limited harmonic distortion measurements (for products below the Nyquist limit) using a sine-wave generator and the Blackman time window function—the markers read out the difference in their decibel values, which can easily be converted by a user to percent: -40dB = 1%.

I thank Mr. Thompson for his effort, his suggestions, and his stimulating letter. If he uses and looks at the IMP system in its entirety, and in its full scope, I think he will agree that the rectangular pulse is indeed an appropriate stimulus.

CURRENT EVENTS

Thanks to Dick Campbell for his kind words ("Smoking Ban," *SB* 6/93) about my Prism V articles in *SB* 4/93 and 5/93, and his interesting remarks on the demands on capacitors used in passive crossovers. He specifically comments on the capacitor used in the paralleled notch filter; but the demands on this capacitor are no greater than those on any other capacitors in the crossover network that are connected in series with the drivers.

All capacitors in my passive crossover,

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Reader Service #2

World Radio History

including the 30µF cap in the notch filter, are high-voltage Solen metallized polypropylene, which have an extremely low equivalent series resistance (ESR). As Mr. Campbell correctly notes, ESR is a source of heating in the capacitor that may lead to its destruction.

I asked Denis Oullet of Solen Electronique, Inc., about the "ripple current" specification of Solen's capacitors. He didn't provide such a specification, but while we were on the phone he measured a 30µF metallized polypropylene capacitor for its ESR and differential of frequency (df) performance with a Stanford Research model SR720 LCR meter. The measurements are as follows:

	ESR	df
100Hz	0.0046Ω	0.00008
1,000Hz	0.0025Ω	0.00045
10,000Hz	0.0023Ω	0.0040
100,000Hz	0.0030Ω	0.130

These ESR measurements confirm the desirability of using metallized polypropylene capacitors to assure low ESR and high current handling. Mr. Ouellet also affirmed that electrolytic capacitors exhibit much higher ESR and much poorer current handling capability as a result.

Although I was formerly unfamiliar with the issue of "ripple current," I am satisfied I have nothing to worry about.

Randy Parker
Lancaster, PA 17601

CHAMBER MUSIC

When I read Walter D'Ascenzo's article "The AR-1 Rejuvenated" (*SB* 2/82, p. 7), I never expected to find myself with another AR product, much less a 30-year-old one. However, I now am using a pair of AR-3a cabinets as woofers. I have bypassed the internal crossover and am powering the 12" drivers directly from my Adcom 555. Can Mr. D'Ascenzo tell me if all of the modifications he describes are necessary for operation with a cutoff at 18dB/octave at 100Hz? In particular, will the chamber resonance he refers to be a factor in my application? Thanks.

Les Winter, PE
New York, NY 10003

Walter D'Ascenzo responds:

It does not surprise me that inquiries still come in regarding the AR woofer system. Whether you have the AR-3, the AR-3a, or the AR-1 cabinet, the price of admission buys you a well-made enclosure even if the drivers themselves are no longer useful.

The AR 12" woofer, used in the AR-3a speakers through to the AR-10pi speakers, use the steel-frame driver with a vacuum-formed cone and a natural-rubber half-roll surround. These drivers have a low-resonance, long-throw performance, but they are plagued with surround aging in the form of dry rot. You can replace either the woofers or the surrounds.

My response to your enquiry is an emphatic yes, yes! It is vital that the woofer be capable of reasonable performance well into the midrange, and at a 100Hz crossover you will still get "bleed." The modifications I suggest in my article should control resonances and ringing, allowing the woofer to reproduce the lower midrange as accurately as possible with low coloration. As a hobbyist, free of the constraints manufacturers contend with, you can fine-tune the sound of the woofers to blend with the rest of your speakers. (Be sure to do the tuning without the crossover, as it is important to hear what you are doing. The crossover will interfere with that effort.)

The chamber resonance caused by the area under the dust cap is one of the modification's most important features. For a home demonstration of the effect, just cup your hands over your mouth and talk or sing.

Placement is another important issue. Cosmetically, the subwoofers should be hidden. For the best sonics, place them next to your main

speakers and align them by time by lining up the woofer voice coils with voice coils of the drivers in your main speakers. The results will more than repay the effort.

One final consideration: don't limit yourself to the "magical" 100Hz crossover. If you can adjust the crossover point, experiment. You may find that, in your system, another crossover point will provide a more optimum synergism with your main speakers. If you can successfully lower the crossover point below 80Hz, it will sound as if your main speakers, not your woofers, are reproducing the deep bass. The ploy is especially convincing if you can camouflage the woofers without compromising performance.

High-end audiophiles please note. The Adcom 555 power amplifier may cost far less than your silver-plated, big-bucks amplifiers, but it will impose an iron grip on good woofers and fill out the bottom end without mush or overhang. Happy POOGEing!

CUTTING EDGE

I found Bob Wayland's article on materials for speaker cabinets ("Wayland's Woodworking World," *SB* 3/93, p. 56) most interesting, especially his tabulation of the modulus of various panel materials.

I had been using particleboard for most of my speaker projects but when I learned that speaker builders favor MDF, I used it in my latest project. I was delighted by its excellent machine ability and its tendency *not* to dull saw blades. My *subjective* impression, however, is that it is not as stiff as particleboard. Does MDF have a lower modulus?

David J. Meraner
Scotia, NY 12302

Bob Wayland responds:

You seem to raise two questions in your letter, one overt and one covert. Let me address your covert question first. The remark I made about particleboard (PB) and medium-density fiberboard (MDF) dulling cutting tools quicker was in relation to the effects on tools cutting solid wood. The question you raise about the difference between the effects on tools of PB versus MDF is considerably more complex, mainly because there are so many different types. If you wish to establish the relationship for a particular PB and MDF, I suggest that you start with two identically sharp saw blades and run a series of wear tests, one blade cutting PB and the other cutting MDF. Please let me know what you find.

I did not provide a value for the modulus of elasticity for PB because of its great variance. The American National Standard for Wood Particleboard, ANSI A208.1, lists 15 grades of PB with moduli values between 80,000 and 500,000 psi. If you bought the PB from a local lumberyard, it might be interior grade floor underlayment PB, grade 1-M-1, which has a modulus of 250,000 psi. My next question has even more teeth in it: was the PB manufactured to specification? Who knows?

Wood World

Continued from page 60

the frame will automatically be square. Another advantage is that only one clamp is needed, a hand-screw clamp (*Photo 8*). Be sure to make the jig out of a hardwood, such as maple or oak, as the pressures can easily split the corner blocks. Again, use wax paper or soap to protect against stray glue.

RECAPITULATION

Clamping doesn't have to be a Rube Goldberg adventure. KIS—keep it simple. Follow one simple rule: Always assure the correct uniform pressure on the glue surfaces. And be willing to be inventive.

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JBL L100 Century speakers, 12″ three ways, components are in excellent condition, cabinets are good, $500 for pair or best offer. (805) 392-9575.

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Audio Catalog #1—The personal library of the late Donald Aldous, previous editor of *Hi-Fi News* magazine, comprising hundreds of rare and out of print books pertaining to early audio and vacuum tube technology. Many are written by noted authors such as Olson, Crowhurst, Briggs, Moir, Babani and McLachian. Also included are many catalogs, magazines and ephemera. For catalog, please send $2 (US) or $4 (Foreign) to: *New Wireless Pioneers*, PO Box 398, Elma, NY 14059 USA, Phone/Fax: (716) 681-3186.

California Audio Delta CD transport and remote, best unit under $1,000, very low hours, absolute mint condition, original manual and shipping carton, why pay retail ($799), your cost $550. Harman-Kardon Citation 11, 5 band frequency equalizer/preamp, $75, needs work on one channel. Split shipping, 1-800-227-6121, 9-5 (CT).

Viking 85 tape deck, 3¾ and 7½; $25. General Radio BFO Model 1304-B, needs tuning knob, otherwise near mint, $100. AD-YU Phase Meter Type 405, needs work, $15. Systron RDB Counter Model 1304, lots of nixies, $40. Ballantine VTVM Model 300, $25. Heathkit VTVM Model IM-11, no handle, $20. McIntosh MC-250, $200. Heathkit FM stereo generator, Model 1G-37, with manual, needs alignment, $25. Dynaco PAT-4 preamplifier, $150. Dyna PAS Stereo Cost Cutters kit parts, chassis, front panel, transformers, controls, boards, and some other parts, $125. Pair of AR 4x speakers, $100. Bernoulli Box CDS-PC 20 by Iomega, only $10. Or make an offer for the whole mess. You pay shipping or if close enough, I'll drop it off. J. Curtis, (715) 341-5880.

Seck MK II, 18 channel recording console with power supply and case, $1,700; Whirlwind 100′ snake, 18 channel, $200; two Biamp EQ-230 15 band 2/3 octave graphic equalizers, $200/ea.; three-way P.A. speaker system set up for biamped operation, four cabinet system, very smooth and defined. Components include: JBL 15″ woofers, 7″ Audax pro mids and EV-ST350 tweeters. Passive crossovers at 5K, fourth-order LR crossover from 300-800Hz range, $1,600. Call John Del Signore, (518) 692-8282.

Beautiful three-way horn loaded loudspeakers 1½″ thick birch veneer cabinets, look and sound like Klipsch Lascalas, $850 or best offer. Adcom GFP-555, mint condition, $325 or best offer. Pete Groth, 384 Trestle Lane, Clinton Corners, NY 12514, (914) 266-4183, 6–9 p.m. (EST).

ORCA Prototype, custom two-way 17NDLB/C2-22 walnut cabinet with carbon loaded baffle outboard crossover by D'Appolito, $850. Call John (818) 707-1629.

Velodyne VLD 15-11 subwoofer, mint, $990. New Sony DTC-M100 SemiPro DAT recorders, $500. Power Amplifiers: New Hafler XL600s and two SoundCraftsman 200W. Sony CDP707ESD CD Player, $750; six new EVM15B II 15″, $150; Altec 418A 15″, $185; Kenwood 5300 tuner, $85; Thorens w/Rabco, 15″ woofers from '30s/'40s *Voice-of-the-Theater.* One lot Club/Disco stage lighting equipment, offer? Test Equipment: Tektronix Oscilloscopes—531A and 453; much other test equipment, parts, etc. Lot PC Compatible computer equipment and parts. Will fax/send list on request. John (602) 553-8207, or leave voicemail message.

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NAD 2600 power amp 150 WPC 500W bridged, soft clipping input level control, $350; Speakerlab K-Horn bass section with Marchand Electronic Crossover, $375/pr. (San Francisco Bay area only); Van Alstine Super PAS II vacuum tube preamp, wood cabinet, extra two 12AX7 (5751), four chinese 12AX7, one 12 × 4, $325. All plus shipping. (415) 768-1294.

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Hafler IRIS remote controlled preamp, $395; Hafler DH-500 power amp, $395. Randy (717) 898-3085.

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Amplifiers: Dynaco stereo 70 with polypro caps, $150; Harman-Kardon Citation 12, unknown condition, $50; HH Scott Model 250 mono, excellent condition, $50; Dynaco modified PAT-5, $50. All plus shipping. Lou Garner, 1400 S Valley View, #2004, Las Vegas, NV 89102.

JBL 2426 horn drivers, $250/pr.; 2307/2308 horn/lens assembly, $150/pr.; 2402 bullet tweeters, Alnico, $180/pr.; 2202 12" woofers, $225/pr.; "baby cheek" horns for 1" drivers, $150/pr.; 4612 OK monitor speakers, $500/pr.; single 4660 defined coverage speaker system, $450; Altec 411-8A, $375/pr. David (914) 688-5024.

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Marantz 250M power amp, $235; Aperture Interconnects (AudioLink): three - 1 meter and two -2 meter, $40/all 5; Advent 101 Dolby unit, $25; one Philips AD0162 T8 tweeter, $12; one Philips AD5060 SQ8 Mid, $15. (414) 458-2057, (4–8 p.m. CST).

DBX 3BX III Range Expander, mint condition, $250.; Egg crate foam, 15" × 15" and ½" to ¾" thick, for lining speaker cabinets, $2.50/sheet plus shipping. Nick (209) 583-6511 (PST).

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Altec studio monitors #9846, has 411 woofers, 511 horns, 806 divers, $750/pr. (will part out); #9849 cabinets, 12" + horn, $550/pr.; empty 9849 cabinets, $100/pr.; extra pairs 802, 806, 807, 902, 908, 32C horns, 1500Hz and 800 Hz crossovers and more. Trades possible. David (914) 688-5024.

McIntosh 2002, 200wpc amp, mint, $1,200; Carver S7+ mono amps, 575wpc, mint, $1,200; Soundcraftsmen PP4 205wpc amp, mint, $450; Harman-Kardon Citation 12 60wpc amp, excellent, $200; Eclipse 1250 12" woofers, $15/ea. Paul (914) 528-1510.

Advanced Akustic Compact Monitors from kit with Dynaudio 17W75 woofers and D-28AF tweeters. Worked flawlessly until a child poked in the tweeters. $100 includes COD shipping. SSI 3000 dolby pro-logic processor. Remote and amps for rear channels. Used one year, fine shape, $110 includes COD shipping. Larry Rasmussen (206) 435-5263.

WANTED

Information on Karlson loudspeaker products and related items. Also Transylvania Power Corp. products. I would consider purchasing 8" and 10" "ULTRA FIDELITY ENCLOSURES" and Transylvania products. Would like to correspond with anyone doing Karlson related design work. David A. Young, PO Box 87, Blockhouse, Nova Scotia, Canada, B0J 1E0. (902) 624-8008.

Used vifa, Dynaudio, Focal drive units at reasonable prices. Richard Glenn (617) 493-3158.

Caltech Music Lab needs gifts of several pairs of Quad ELS (original electrostat, not ESL-63) never overvoltaged. James Boyk (818) 395-4590, Fax (310) 470-9140.

One Focal 10V516 woofer. Jim (708) 425-6719.

Sumo Delilah crossover or Bryston 10 b or better. Call Andrew (514) 281- 7954.

DeCoursey crossover cards. Prefer 65Hz 18dB, 100 Hz, 18dB, 400 Hz, 18dB, 2200, 18dB. May be interested in other values; Mod Squad passive line drive age for $100; one Focal 8″ mid woofer 8K515; two Focal 8″ 8K516 mid woofers. Also source for MU metal. Call after 4 p.m. (CT) Charles, (901) 386-1121 (TN).

Manual or any information on Weston Model 785 Circuit Tester, about 30 years old. Will pay costs of copying, postage. Bill c/o Jaffa, PO 543, Sharon, MA 02067, (617) 461-3821.

Teac reel to reel. Desire model from following: X-200, X-300, X-300R, X- 2000M, X-2000R. Also seek a revolver turntable with standard tonearm. Larry (214) 226-9017. M–F, 1–9.

CLUBS

WANTED: SPEAKER AND AUDIO AMATEURS IN THE BRADENTON/SARASOTA/ST. PETERSBURG/TAMPA, FLORIDA AREAS. Would like to form a club and develop a lab for testing speakers/amps/preamps and passive and active crossovers or just to discuss speaker projects and ideas with. Angel Rivera, Bradenton, FL (813) 792-3870.

WANTED: PRO-SOUND HOBBYISTS OR CURRENT EMPLOYEES IN SOUND REINFORCEMENT FIELD to correspond with USAF serviceman in England. Looking for equipment sources and contact with anyone who has a passion for quality reproduction of live music. Rick Diaz, PSC 41, Box 6912, APO AE, 09464.

ELECTROSTATIC LOUDSPEAKER USERS GROUP is now a world-wide network for those interested in sharing valuable theory, design, construction, and parts source information. If you are interested in building, or have built, your own SOTA ESL we invite you to join our loose-knit organization. For information, send a SASE to: Barry Waldron, 1847 Country Club Dr., Placerville, CA 95667.

MONTREAL SPEAKER BUILDER CLUB Meets when it can. BYOB, discussions range from speaker design and testing to equipment modification. All welcome, contact Jeanne Mance, Montreal, Que, H2V 4J5 (514) 281-7954.

THE CATSKILL AND ADIRONDACK AUDIO SOCIETY invites you to our informal meeting. Join our friendly group of audio enthusiasts as we discuss life, the universe and everything! Tubers, Transistors, vinyl canyons or digital dots. No matter what your level of interest, experience, or preferences, you are welcome. Contact CAAS at (518) 756-9894 (leave message), or write CAAS, PO Box 144, Hannacroix, NY 12087. See you soon!

ALL PERSONS INTERESTED IN STARTING AN AUDIO/SPEAKER BUILDER CLUB IN THE SOUTHWEST MISSOURI—NORTHWEST ARKANSAS AREA, please send your name, address, phone number and something about yourself to: Greg McKinney, 900 S. Roanoke, Apt. #2, Springfield, MO 65806.

AUDIOPHILES IN THE DAYTON/SPRINGFIELD, OHIO AREA: We are forming an audio club. Please contact me if you're interested in construction, modifications, testing, recording or just plain listening to music. Ken Beers, 1756 Hilt Rd., Yellow Springs, OH 45387, (513) 767-1457.

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send a self-addressed, stamped #10 envelope to Tim Eding, PO Box 611662, San Jose, CA 95161.

THOSE INTERESTED IN AUDIO and speaker building in the Knoxville-East Tennessee area please contact Bob Wright, 7344 Toxaway Dr., Knoxville, TN 37909-2452, (615) 691-1668 after 6 p.m.

CLASSIFIED INFORMATION

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

FOR SALE: For readers to sell personal equipment or supplies.

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.

PRICING

All advertising is $1.50 per word, $10 minimum per insertion. Deduct 5% for a 8x contract. Please indicate number of insertions. Payment MUST accompany ad. *No billing for word classified advertising.*

AD COPY

A word is any collection of letters or numbers with a space on either side. *No abbreviations*; please spell out all words. Count words, not letters. Ad copy should be clearly typed or printed. Illegible ads will be discarded.

SUBSCRIBERS

Receive *free For Sale, Wanted,* and *Club* advertising; 50 words maximum; each additional word just $.20. Please 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, 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

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues includes monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-Bing equipment. New members welcome. Contact Frank J. Alles, (908) 424-0463, 209 Second St., Middlesex, NJ 08846; or contact Bob Young, (908) 381-6269, or Bob Clark, (908) 647-0194.

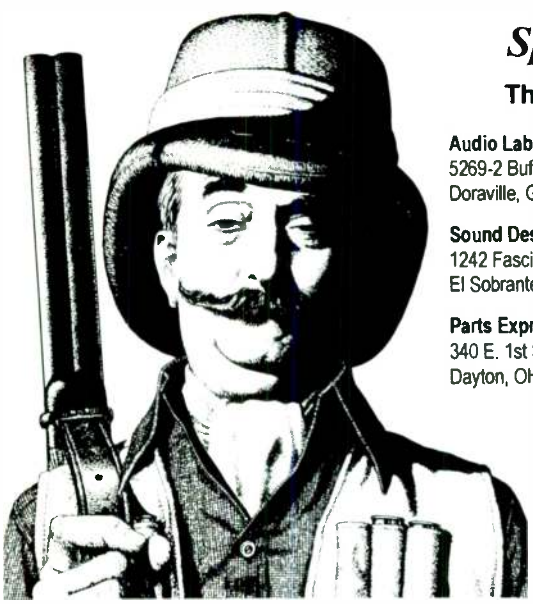
SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, A–B listening tests, equipment clinics, recording studio visits, and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 544-8453 or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

IF YOU ARE AN "ORGAN MUSIC LOVER" and like to test your audio system, **SFORZANDO** has room for a few more members. We have about three thousand "Live," on-the-spot cassette tapes that are not available in the stores. We are happy to lend them to you via the mail. Just ask EA Rawlings, 5411 Bocage St., Montreal, Canada, H4J 1A2.

THE BOSTON AUDIO SOCIETY the nation's oldest (founded 1972), seeks new members. Dues includes the monthly meeting notice and our newsletter, the *BAS Speaker* (6 times/year). Recent issues cover Carver, a/d/s; the founder of Tech Hi-Fi; Photo CD; plus visits from famous speaker designers; listening tests; measurement clinics; research investigations; and more. Back volumes available. Membership includes engineers, journalists, consultants, and music-loving audiophiles like yourself. For information write to PO Box 211, Boston, MA 02126-0002, USA.

THE INLAND EMPIRE AUDIO SOCIETY (soon to become) **THE SOUTHERN CALIFORNIA AUDIO SOCIETY—SCAS** is now inviting audiophiles from all areas of Southern California and abroad to join our serious pursuit for that elusive sonic truth through our meetings and the **IEAS'** official speaker, *The Reference* Newsletter. For information write or call, Frank Manrique, President, 1219 Fulbright Ave., Redlands, CA 92373. (714) 793-9209.

PIEDMONT AUDIO SOCIETY Audio club in the Raleigh, Durham, and Chapel Hill area is meeting monthly to listen to music, demonstrate owner-built and modified equipment, and exchange views and ideas on electronics and speaker construction. Tube and solid state electronics are of interest and all levels of experience are welcome. Kevin Carter, 1004 Olive Chapel Rd., Apex, NC 27502, (919) 387-0911.



ARIZONA AUDIOPHILE SOCIETY Located in Metropolitan Phoenix is a growing and active club in the pursuit and reproduction of recorded music. New members are welcome. Meetings are last Tuesday of each month. Receive monthly newsletter and biannual journal. Club discounts with local high- end audio dealers. Send inquiry to Arizona Audiophile Society, PO Box 13058, Scottsdale, AZ 85267, or call Bob Williams, (602) 944-5929.

PACIFICNORTHWEST AUDIO SOCIETY (PAS) consists of 60 audio enthusiasts meeting monthly, second Wednesdays, 7:30–9:30 p.m. at 4545 Island Crest Way, Mercer Island, WA. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130 or Nick Daniggelis, (206) 323- 6196.

THE COLORADO AUDIO SOCIETY is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five bimonthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 11685 W 22nd St., Lakewood, CO 80215, (303) 231-9978.

AUDIO SOCIETY OF MINNESOTA Now in its 15th consecutive year! Serving the many and varied interests of audiophiles in the upper midwest. Monthly meetings, tours, audiophile concerts, special guests. For information and a sample of our latest newsletter, write ASM, PO Box 32293, Fridley, MN 55432 or call our 24 hour "Audio Hotline," (612) 825-6806.

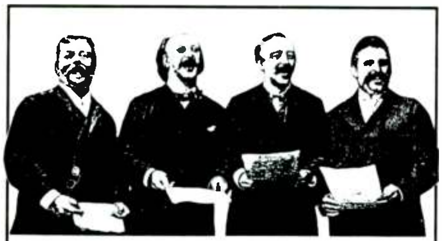
LONDON LIVE D.I.Y. HI-FI CIRCLE meets quarterly in London, England. Our overall agenda is a broad one, having anything to do with any aspect of audio design and construction. We welcome everyone, from novice to expert. For information contact Brian Stenning, 081-748-7489.

THE ATLANTA AUDIO SOCIETY is dedicated to furnishing pleasure and education for people with a common interest in fine music and audio equipment. Monthly meetings often feature guest speakers from the audio manufacturing and recording industry. Members receive a monthly newsletter. Call: Chuck Bruce, (404) 876-5659, or Eddie Carter, (404) 847-9296, or write: A.A.S., 4266 Roswell Rd. N.E., K-4, Atlanta, GA 30342-3738.

CONNECTICUT AUDIO SOCIETY is an active and growing club with activities covering many facets of audio—including construction, subjective testing, and tours of local manufacturers. New members are always welcome. For a copy of our current newsletter and an invitation to our next meeting, write to: Richard Thompson, 129 Newgate Rd., E. Granby, CT 06026, (203) 653-7873.

NORTHERN VIRGINIA BUSINESS CLUB. I am interested in turning my speaker building interests into a profitable business. So I'm organizing this Business Club to attract other Speaker Builder Entrepreneurs with the same interests. Call Frank Troy (703) 912-8226, M–F, 7:30 a.m. to 4:30 p.m.

THE WESTERN NEW YORK Audio Society is an active, long-established club located in the Buffalo area. We issue a newsletter and hold meetings the first Tuesday of every month. Our meetings attract many prominent manufacturers of audio related equipment. We are involved in all facets of audio—from building/modifying to exposure to the newest high-end gear, and the chance to hear more types of music. For information regarding our society, please write to WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.



WANTED

AUDIO CLUB MEMBERS

(Good singing voice not required)

Learn about the latest equipment, techniques and recordings through group meetings, tours and newsletters. Ask questions. Share viewpoints and experiences. *Have fun!*

If there's no club in your area, why not start one? Our club ads are free up to 75 words ($.20 per word thereafter). Copy must be provided by a designated officer of the club or society who will keep it current.

ESL BUILDERS GROUP is a new address for people who have built or want to build ELECTROSTATIC LOUDSPEAKERS and ASSOCIATED (TUBE) DRIVERS, or are just interested. We will concentrate on ESL-related building projects but also look at the theoretical aspects of acoustics and electronics. Interested? An answer is ensured, if you include some kind of compensation for postage and handling. Write to: Gunter Roehricht, Buhler STR.21, 7030 Böblingen, Germany.

THE PRAIRIE STATE AUDIO CONSTRUCTION SOCIETY (PSACS), meets every other month. Meetings feature audio construction, design, and analyses, blind listening tests, equipment clinics, autosound, lectures from manufacturers and reviewers. PSACS, PO Box 482, Cary, IL 60013, call Tom, (708) 248-3377 days, (708) 516-0170 eves.

WEST VALLEY AUDIO SOCIETY. We are starting a group interested in all aspects of high perfomance audio. West San Fernando Valley, CA. Contact Barry (818) 225-1341.

WASHINGTON AREA AUDIO SOCIETY Meetings are held every two weeks, on Fridays from 19:00 hours to 21:30 hours at the Charles Barrett Elementary School in the city of Alexandria, Va. Prospective members are welcome but must register in advance in order to be admitted to the meetings. No exceptions please. If interested please call Horace Vignale, (703) 578- 4929.

DO YOU LIVE NEAR LAWRENCE KANSAS? I am a student at the University of Kansas looking for other speaker builders within driving distance. I would like to exchange ideas and listen to other homebrew systems. Michael Marmor, 1520 Lynch Court #2, Lawrence, KS 66044, (913) 843-8993.

HI-FI COLLECTOR/HOBBYIST seeks "living letters"/audio pen pals from other states to correspond via reel-to-reel tape. Non commercial strictly; make up short monologues on subjects from vintage technology, with regional FM excerpts for background or equipment samples, from personal tales of yard sales scavenging success, repair/restoration tactics and strategies, favorite service centers, general ways to handle the burgeoning obsession with arcane hi-fi gear. All correspondence on 3", 5", 7" reels (¼" tape) will be cheerfully answered and tapes returned via parcel post. James Addison, 171 Hartford Rd., Apt. #7, New Britain, CT 06053.

THE LOS ANGELES AREA LOUDSPEAKERS DESIGNERS GROUP If you're just starting out or an experienced builder and would like to share ideas on speaker design and listen to each other's latest creations, give us a call. Geoffrey (213) 965-0449, Edward (310) 395-5196.

MEMPHIS AREA AUDIO SOCIETY being formed. Serious audiophiles contact J.J. McBride, 8182 Wind Valley Cove, Memphis, TN 38125, (901) 756-6831.

THE HI-FI CLUB of Cape Town in South Africa sends a monthly newsletter to its members and world-wide subscribers. To receive an evaluation copy of our current newsletter, write to: PO Box 18262, Wynberg 7824, South Africa. We'll be very pleased to hear from you.

Ad Index

Reader Service #18 ➞

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