THREE/92

U.S. $7.00
CANADA $8.00

Speaker Builder

THE LOUDSPEAKER JOURNAL

Good News

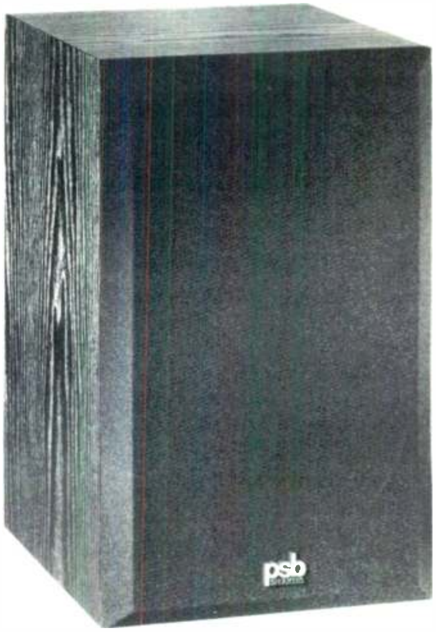
SENNET CONCEPTS is introducing several audio products for homes and vehicles. Included are the Model SH600, a six-speaker home-theater surround-sound package that retails for $350; the Model SH200, a three-piece stereo clock radio system, $200; the Model SH88, an 8" home-passive subwoofer, $125; and the Model SH5, an indoor/outdoor two-way speaker system, $125.

For details, contact Sennet Concepts, PO Box 1630, Old Milford Rd., Milford, PA 18337-2630, (717) 296-2818.

Fast Reply #GG1414

PSB SPEAKERS introduces the Alpha, a two-way loudspeaker system utilizing a 6½" polypropylene woofer and a ½" polyflare dome tweeter. It is the first in a series to range in price from $199–$899. A narrow profile cabinet (8¾" across) features minimal baffle area, flush tweeter mounting, frame-less grille attachment; a beveled-edge grille eliminates protrusion causing reflections.

The Alpha crossover employs top-grade, high-voltage capacitors for unrestricted clarity and dynamics. The connectors are heavy-duty push terminals that accept up to 12-gauge cabling, as well as banana plugs including standard ¾" spacing dual-banana blocks.



The PSB Alpha is available for $199/pair. Contact PSB International Inc., 633 Granite Court, Pickering, Ontario, Canada L1W 3KI, (212) 888-6610.

Fast Reply #GG1436

MICROMATH SCIENTIFIC SOFTWARE has a free, 16-page color catalog containing information on all its scientific and engineering software, including programs for modeling, simulation, and curve fitting. It includes descriptions of plotting packages, ODE solvers, statistics, and software for Laplace transform inversion. Software for preparation of publications includes scientific word processors, math editors, and bibliographic software.

MicroMath products include Graph, to plot scientific and engineering data; Diffeq, a differential equation solver; and Minsq, a nonlinear, curve fitting and simulation program. Others are Laplace and Rstrip. The catalog also includes several other scientific and engineering products.

For a free catalog, contact MicroMath Scientific Software, PO Box 21550, Salt Lake City, UT 84121, (801) 943-0290, FAX (801) 943-0299.

Fast Reply #GG1415

TANNOY introduces the Tannoy Sixes loudspeakers which incorporate new thinking on enclosure structure, acoustic point source design, drive unit construction, and Tannoy's own internal DMT (Differential Material Technology). The units also offer improved standing-wave behavior, increased structural rigidity, and improved diffraction performance. Prices range from $279 for the Model 603 to $1,399 for the Model 615. The stands are an additional $149, and the base is $99.

For specifications, contact Tannoy, 141 Linden St., Suite G10A, Wellesley, MA 02181, (617) 239-1692, FAX (617) 239-0096.

Fast Reply #GG780

POLYDAX SPEAKER CORPORATION, a subsidiary of Audax Industries in France, introduces the TW 025 A, a soft-textile, 1" dome tweeter. It is available in two models, with and without ferrofluid, and features a catenary profile. The shape resembles a chain suspended from two equidistant points in space that is left to form a unique shape. The faceplate design is streamlined, the voice coil i wo layers, the diaphragm is replaceable, and the frequency range is 5kHz–20kHz. It has a sensitivity of 92dB 1W/1M and power handling is 50W RMS.

Contact Polydax Speaker Corporation, 10 Upton Dr., Wilmington, MA 01887, (508) 658-0700, FAX (508) 658-0703.

Fast Reply #GG646

CARVER introduces the SD/A-370 and SD/A-410 linear single-bit CD players. Both units offer near-perfect low-level linearity, almost ruler-flat frequency response, and signal-to-noise ratios of over 100dB. The SD/A-370, which sells for $699.95, is a 10-CD changer that can provide over ten hours of music without interruption. Features include 32-track random programming, Intro Scan, Random Play, remote control, and time edit. A special circuit, Soft EQ, prevents loss of sonic front-to-back depth, side-to-side imaging, and excessive brightness. The TLM-10 10-CD Magazines worth $34.95 are included with your purchase.

The SD/A-410, which costs $279.95, also uses a linear single-bit D/A converter and includes shuffle play, intro scan, audible search and scan functions, calendar display, coaxial digital output, and a headphone jack.

Contact Carver, PO Box 1237, Lynnwood, WA 98046, (206) 775-1202, FAX (206) 778-9453.

Fast Reply #GG694

DIGITAL CABLE RADIO (DCR) announces the addition of "Audiophile Audition," an innovative music and talk show that caters to audio buffs. Host, John Sunier offers a musical mix of classics and jazz with an emphasis on sonics, and interviews leaders in the audio or music fields.

DCR is a 24-hour, commercial-free audio service featuring CD-quality, digital sound transmitted to customer's stereos via cable TV. With 19 channels of music each formatted to a specific genre of music.

Contact Digital Cable Radio, 2200 Byberry Rd., Hatboro, PA 19040, (215) 957-8290, FAX (215) 443-9454.

Fast Reply #GG1432

AZ-USA announces the introduction of AZ Acoustical Barriers that are designed to stop noise transmission and fight resonance in a variety of applications. Available as either a ⅛"-thick, loaded vinyl barrier or as a barrier combined with ¼" acoustical foam, both are easy to install and maintain and resolve sound transmission and vibration problems. These 1lb/ft² loaded vinyl materials are Class 26 with a sound-transmission loss performance range of 13dB at 125Hz to 40dB at 4kHz. They do not contain lead additives, are not restricted by OSHA, and comply with UL and DOT specifications.

Contact AZ-USA, 1401 W. 76 St., Suite #250, Minneapolis, MN 55423, (612) 861-2290, FAX (612) 861-2103.

Fast Reply #GG1439

ETON AMERICA announces a new line of audiophile automotive speakers. The Car 200 Hex is a two-way system featuring a 4" Hexacone midbass, 1" neodynium, metal-dome tweeter, and complimentary dividing network for $499/pair. The Car 300 Hex is a two-way system featuring a 7" Hexacone midbass 1" neodynium, metal-dome tweeter, and complimentary dividing network for $569/pair.

Both systems feature midbass units using two layers of Kevlar with a sandwich of honeycomb Nomex. Contact Eton America, Box 9274, Grand Rapids, MI 49509-0274.

Fast Reply #GG456

WORLD AUDIO ENTERPRISES offers Multi-Amp 1.1 and 1.2, high-quality, easy-to-mount, small circuit boards containing components for op amp applications. Features include linear phase circuitry, complete range of gains with 25-turn trimpot, RF filtering, AC-coupled easily changed to DC, and on-board power-supply filtering. Made of glass epoxy with 2-oz. copper foil, 1% metal-film resistors, and bipolar coupling capacitors, the frequency response ranges from 7–550kHz in the Multi-Amp 1.1 and 0.7–550kHz in version 1.2.

Both have full power bandwidth of 200kHz, slew rate of 10V/μS, phase margins of >60°, distortion of <0.008%, and differing input impedances. Each model costs $30.

Contact World Audio Enterprises, 1550 N. Hayworth St. Suite #4, Los Angeles, CA 90046, (213) 876-9535.

Fast Reply #GG1433

CUSTOM SOUND furnishes custom-made polyfoam speaker grilles, which are made of acoustically transparent open-cell polyurethane foam rubber in any size, thickness, design, color, or quantity for custom, replacement, or OEM applications. Dealer price reductions are available.

For a free descriptive brochure, contact Custom Sound, PO Box 463026, Mt. Clemens, MI 48046, (313) 463-5039.

Fast Reply #GG1440

BRIGHT STAR AUDIO announces the Total Isolation System, an effective vibration and resonance control that incorporates high mass and high absorption to eliminate vibration-caused degradation of audio and video performance.

A Little Rock isolation pod ($99) provides a high degree of concentrated mass to the top of the component's chassis. This dampens destructive stored energy and forces it down into a Big Foot isolation platform ($149) where it is absorbed by a 45-lb. bed of sand.

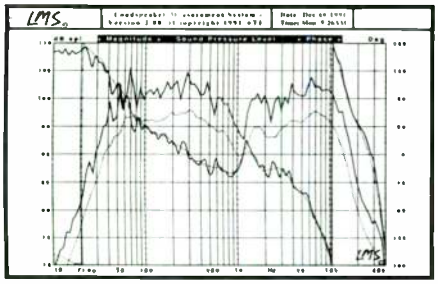
Contact Bright Star Audio, 2363 Teller Rd. #115, Newbury Park, CA 91320, (805) 375-2629, FAX (805) 375-2630.

Fast Reply #GG151

AUDIO TEKNOLOGY, INC. (ATI) introduces a Loudspeaker Measurement System (*LMS!*) which offers a cost-effective solution to electro-acoustic measurements. It comes with a calibrated, high-quality measurement microphone and offers all SPL readouts directly into absolute dB/SPL. You provide only a PC with a single open-expansion slot and benefit from features such as Speaker Parameter Measurement and designing crossovers with actual impedance and SPL. You can optimize these designs to your satisfaction. *LMS!* sells for $995.

Contact ATI, 7556 SW Bridgeport Rd., Portland, OR 97224, (503) 624-0405, FAX (503) 624-0194.

Fast Reply #GG624



MUSIC INTERFACE TECHNOLOGIES (MIT) has published Technical Note 101 entitled "Transportable Power in Audio Cables: Energy Storage Elements and the Power Factor" by Bruce A. and Timothy A. Brisson. In an effort to de-mystify audio cables, the authors have included mathematics for the engineer as well as non-technical, visual assists for the audiophile. Eight chapters, three appendices, and a two-color format further enlighten the reader as to the finer details of cables for the high-end audio industry.

For information, contact Music Interface Technologies, 3037 Grass Valley Highway, Suite 8212, Auburn, CA 95603.

Fast Reply #GG1438

Speaker Builder

Edward T. Dell, Jr. *Editor/Publisher*
Karen Hebert *General Manager*
Contributing Editors
Joseph D'Appolito Robert Bullock
John Cockroft David Davenport
Vance Dickason Bruce C. Edgar
Gary Galo G. R. Koonce
David Moran
Patricia French *Managing Editor*
Janine Jackson *Features Editor*
Margaret Gurney *Assistant Editor*
Christine Orellana *Graphics Director*
Graphics Assistants
Glenn Galloway Edward Roane
Katharine Gadwah *Circulation Director*
Circulation Assistants
Doris Hamberg Mary Weiss
Judy Walker *Advertising Assistant*
Techart Associates *Technical Drawings*

Advertising Rates & Schedules
Rally Dennis
PO Box 494
Peterborough, NH 03458
Advertising Phone: (603) 924-6710

Editorial and Circulation Offices
Post Office Box 494
Peterborough, New Hampshire 03458
(603) 924-9464 FAX: (603) 924-9467

Speaker Builder is published bi-monthly by Edward T. Dell, Jr., PO Box 494, Peterborough, NH 03458.

All subscriptions are for the whole year. Each subscription begins with the first issue of the year and ends with the last issue of the year. A sample issue costs $4 in the US, $5 in Canada.

Subscription rates in the United States and possessions: one year (six issues) **$25**, two years (twelve issues) **$45.** Canada add **$6** per year for postage. Overseas rates **$40** for one year, **$70** for two years. Subscribers residing outside the US and Canada are served by air.

To subscribe, renew or change address in all areas write to Circulation Department or call (603) 924-9464 for MC/Visa charge card orders. For gift subscriptions please include gift recipient's name and your own, with remittance. A gift card will be sent.

A Note To Contributors

We welcome contributions for possible publication in the form of manuscripts, photographs or drawings, and will be glad to consider them for publication. Please enclose a stamped, addressed return envelope with each submission. While we cannot accept responsibility for loss or damage, all material will be handled with care while in our possession. Receipt of material is acknowledged by postcard. Payment is on publication.

Speaker Builder (US ISSN 0199-7920) is published bi-monthly at $25 per year; $45 for two years, by Edward T. Dell, Jr. at 305 Union St., Peterborough, NH 03458 USA. Second class postage paid at Peterborough, NH and additional mailing office.

POSTMASTER: Send address changes to **SPEAKER BUILDER**, PO Box 494, Peterborough, NH 03458.

About This Issue

On page 10, **Roy Littler** offers an affordable alternative to expensive test equipment for the budget conscious. He describes how to measure speaker frequency response with a constant average power output and a separate sound level monitor in "A ⅓-Octave Noise Source."

A speaker is not a speaker without an enclosure says **Mark Florian** who gets down to the finer details in "A High-Quality Speaker Cabinet." Follow the author's plans on page 14 to construct a particleboard veneer sandwich that you can adapt for your own project.

In "Bi-Amping the Sapphire II Sub I System," (p. 24) Contributing Editor **Gary Galo** lauds multi-amplification and imparts details on enhancing the popular Audio Concepts loudspeaker.

Examining everything you always wanted to know about capacitors, "Professor" **Richard Honeycutt** stands before the blackboard in "Caps For Passive Crossovers." On page 34, the brave North Carolinian dives into that murky pool of capacitive effects. Honeycutt flushes out such nagging questions as just how bad is the ESR in a modern electrolytic? and do capacitance and/or ESR change with applied voltage?

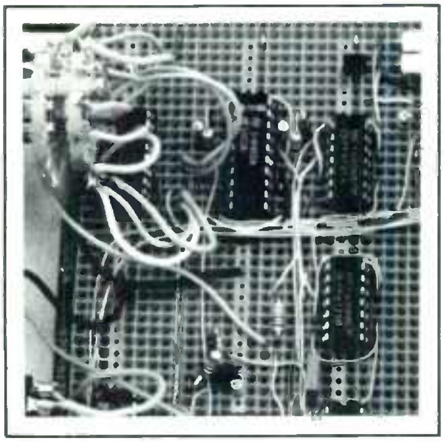
On page 42, **Victor Staggs** recounts his adventures in loudspeaker-system computer optimization in "The Disappearing Loudspeaker." His design accounts for box diffraction at low frequencies, woofer imperfections in the crossover range, and the tweeter's fundamental resonance. The result is a pair of compact near-field monitors equalized for close listening.

Catch audio gadabout **Dave Moran** fishing deep for bass on page 91. He really gets "down" in this month's soul-shaking low-hertz "Moran in the Market."

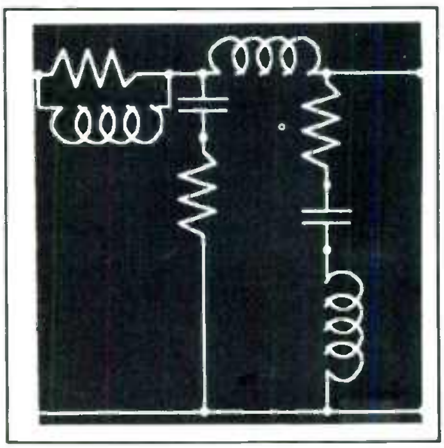
Cover photo: **Mark Florian**

SpeakerBuilder
THE LOUDSPEAKER JOURNAL

VOLUME 13 NUMBER 3 MAY 1992



10



42

FEATURES

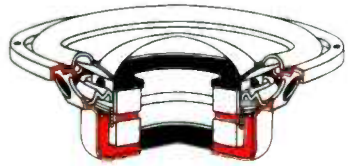
DEPARTMENTS

morel

high fidelity range

MW142

DPC Cone Double Magnet 5" Woofer



A 5" unit with a unique design, incorporating an extremely large diameter aluminium voice coil for such a small unit.

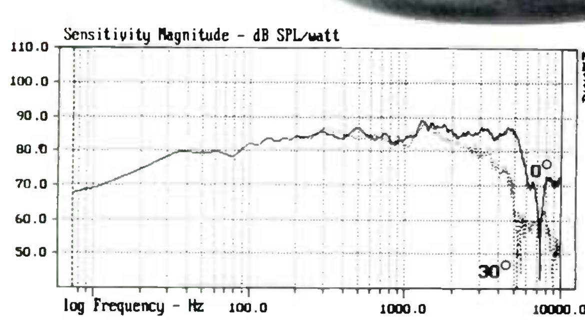
Vented double magnet system with steel chassis — damped polymer composite cone — rubber surround — smooth response to upper limit of 5 Khz with excellent on and off axis response and good roll off.

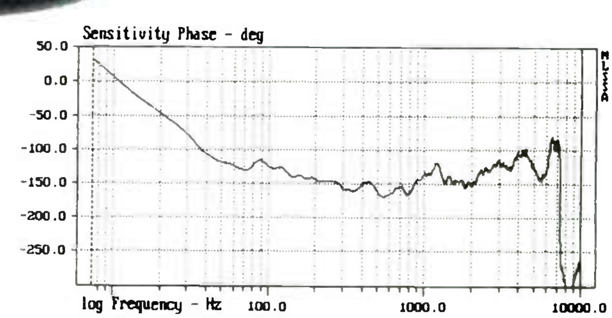
High power handling emanates from this unique design, resulting in a very impressive small bass/mid or mid range unit.

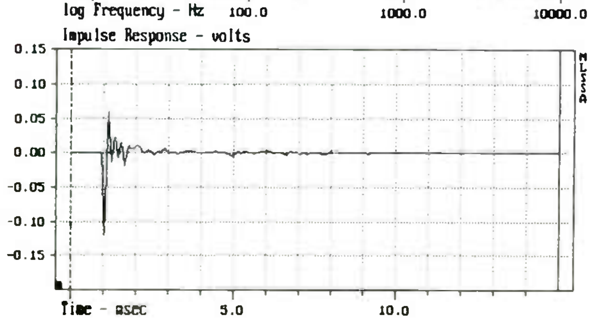
Specification

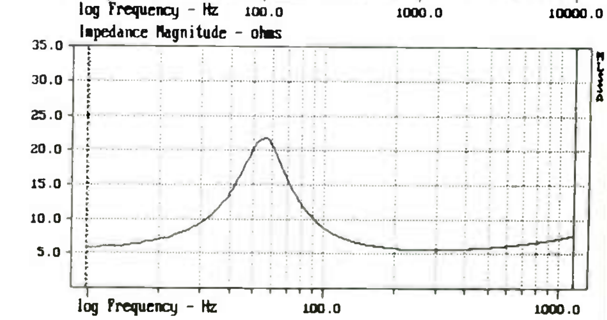
Overall Dimensions	⌀-142mm(5.5") x 52mm(2")
Nominal Power Handling (Din)	150 W
Transient Power — 10ms	1000 W
Voice Coil Diameter	75mm(3")
Voice Coil Type / Former	Hexatech Aluminium
Frequency Response	48-5000 Hz
FS — Resonant Frequency	52 Hz
Sensitivity 1W/1M	86 dB
Z — Nominal Impedance	8 ohms
RE — DC Resistance	5.2 ohms
LBM — Voice Coil Inductance @ 1 Khz	0.5 mh
Magnetic Gap Width	1.35mm(0.053")
HE — Magnetic Gap Height	5mm(0.196")
Voice Coil Height	12mm(0.47")
X – Max. Linear Excursion	3.5mm(0.137")
B — Flux Density / BL Product (BXL)	0.6 T / 5.0 NA
Qms — Mechanical Q Factor	2.14
Qes — Electrical Q Factor	0.62
Q/T — Total Q Factor	0.45
Vas — Equivalent Cas Air Load	7 litres (0.25 ft^3)
MMS — Moving Mass / Rmec	13gm / 2.06ns/m
SD — Effective Cone/Dome Area	90 cm^2
Cone/Dome Material	DPC (Damped Polymer Composite)
Nett Weight	0.97 kg

Specifications given are as after 24 hours of running.









morel (u.k.) ltd.
11 Foxtail Road,
Nacton Rd. (Ransomes)
Industrial Estate,
Ipswich, IP3 9RT
England
Tel: (0473) 719212
Telex:987601 Morel G

morel acoustics USA
414 Harvard Street
Brookline, MA 02146
Tel: (617) 277-6663
FAX: (617) 277-2415

morel®

Editorial

POEMS OR CODES?

"Alright, class, now all those who have ever bought an audio component without any emotion or feelings whatever, raise your hand."

"O.K. Now class, everyone who has ever bought an audio component with a strict, rigorously scientific set of criteria, with no feelings or emotional motive at all, please raise your hand."

The scenario above popped into my mind one day while reading the late W.H. Auden's opening commentary on Owen Barfield's wonderful book, *History in English Words.* Barfield* chose a career in the law, but wrote often and well as an amateur linguist.

Auden begins the Foreword:

"Many who write about 'linguistics' go astray because they overlook the fundamental fact that we use words for two quite different purposes; as a code of communication whereby, as individual members of the human race, we can request and supply information necessary to life, and as speech in the true sense, the medium in which, as unique persons who think in the first and second person singular, we gratuitously disclose ourselves to each other and share our experiences. Though no human utterance is either a pure code statement or a pure personal act, the difference is obvious if we compare a phrase-book for tourists traveling abroad with a poem. The former is concerned with needs common to all human beings; hence for the phrases given, there exist more or less exact equivalents in all languages. No poem, on the other hand, can be even approximately translated into any other language."

The controversy that has raged for over a decade regarding a reliable basis for evaluating the performance quality of audio reproduction equipment is, I believe, considerably illuminated by Auden's analysis. I notice that some on each side of the controversy accuse their adversaries of "nothing but" crimes. "Those Golden Ears are nothing but arrogant gurus." "Those Scientists care for nothing but numbers." Can this be so?

That hardy veteran Peter Walker maintains flatly that he does not listen to a new amplifier to check its quality before putting it on the market. If its measured performance meets his design goals, listening to it is only a matter of mild curiosity, supposedly. Few make so bold a show of scientific fundamentalism, however.

I have recently been struck by conversations with several who write for this magazine whose positions on the Golden Ear/Science controversy are distinctive and clearly on one side or the other. But all those who claim a scientific bias, in further conversation, talk animatedly about listening, experienced listening. And those who espouse subjective evaluation also talk with gusto about rigorous testing and careful measurements.

Those few who take a radical, absolute position on either viewpoint, too often resort to citing bad examples among their adversaries. This is testimony to the futility of making an absolute of either argument. The heat in many of the expressions is a clue to the impossibility of dissecting what happens when any human being makes a choice about which amplifier to purchase.

Can the sonic characteristics of any piece of audio gear be characterized totally by measurement? Can anyone make a genuinely informed decision about buying a preamp without considering the measured performance of the device?

Any dichotomy attempted by all this argument is false. The physical universe is a mess measured by any set of absolutes we might wish to use as yardsticks. Our idea of perfection is an abstract construct. It does not exist. The early theologians, faced with the difficulty of filling in the blanks about the deity, thought up a lot of these. But I rather doubt that very much confirmation of these lofty assertions is readily available nearly two centuries later.

However much respect I have for the scientific achievements of our wonderful epoch, I nonetheless refuse to make precise scientific fact the exclusive basis on which I decide what sounds best in my system.

I cannot resist one further Auden quote. "True understanding is unattainable without both love and detachment, and we can only learn to view anything with detachment by comparing it with other things which are both like and unlike it. We cannot understand the present without a knowledge of the past, our native land without having spent some time in a foreign country, our mother-tongue without a working knowledge of at least two other languages. Without such knowledge, our love of ourselves at the present moment, of our country, of our language, remains an ignorant idolatry, exemplified by the Frenchman who said: 'The great advantage of the French language is that in it the words occur in the order in which one thinks them.'"

Could it be that protagonists on both sides of the controversy are too enamored of their own languages?—E.T.D.

* Barfield was an Oxford classmate and lifelong friend of C.S. Lewis. Reprinted, with permission from *History in English Words,* Owen Barfield. Original copyright, 1953 by Faber & Faber, London. Copyright 1967 by Owen Barfield. Republished in 1985, 1988 by Lindisfarne Press, RR.4, Box 54, Hudson, NY 12534.

A ⅓-OCTAVE NOISE SOURCE

BY ROY LITTLER

Loudspeaker frequency response is measured as the output sound intensity, while driving the speaker from a variable frequency broadband noise source. The noise-source center frequency is set to a series of points through the audio spectrum, and output sound intensity is measured at each setting.

To take these measurements requires a pink noise source, a spectrum analyzer, and a sound level meter. Because this test equipment is beyond the means of many amateurs, there's a need for home-built equipment that contains all the facilities required to measure speaker frequency response. The equipment I describe here is a simple design which provides these facilities; it's comprised of a ⅓-octave noise source with a constant average power output and a sound level monitor.

Before going any further, I will go over some relevant theory.

LOUDSPEAKER TESTING. When speaker frequency response is measured using a single-frequency sinusoidal source, the measured response has a series of null points due to cancellation of sound waves, which take paths of differing lengths in their transmission to the receiving microphone. To lessen the effects of wave cancellation, the source driving the speaker is made a randomly changing band of frequencies, centered on the set test frequency. The frequency band is made ⅓-octave wide, as a compromise between single-frequency testing and having a too-broad band, which would cause loss of detail in the measured response. The center-frequency setting is usually incremented by a ⅓-octave for successive measurements.

The source should have constant power output on all frequency settings to avoid the inconvenience of having to calibrate the generator on each setting.

BANDPASS FILTERS. A ⅓-octave filter has a passband ⅙-octave on each side of the center frequency. Hence, the passband is proportional to the center frequency; increasing frequency broadens the passband. In electrical terms, the filter has a constant Q value. The equivalent Q value is found as follows: the unit bandwidth of a ⅓-octave filter is

$$2^{1/6} - 2^{-1/6} = 0.23$$

and the Q value is the inverse of unit bandwidth, that is, 1/0.23 = 4.3.

In order to sharpen the filter cut off, several filter sections may be connected in series. When several sections are used, the Q values of each section must be reduced in order to have the required overall Q value. The Q value of each section is reduced according to the expression $(2^{1/N} - 1)^{1/2}$, where N is the number of sections in series. In the following design, two similar sections are used, hence N = 2, reducing the Q value by 0.64, making Q = 2.7 for each section.

NOISE GENERATION. A random-number generator is the most practical noise source; its output has constant spectral density. This means that frequency bands of fixed width, taken anywhere in the spectrum, have similar noise power content. This evenly distributed noise is called white noise.

If white noise is used as the input source for a ⅓-octave filter, increasing the center frequency will cause the average

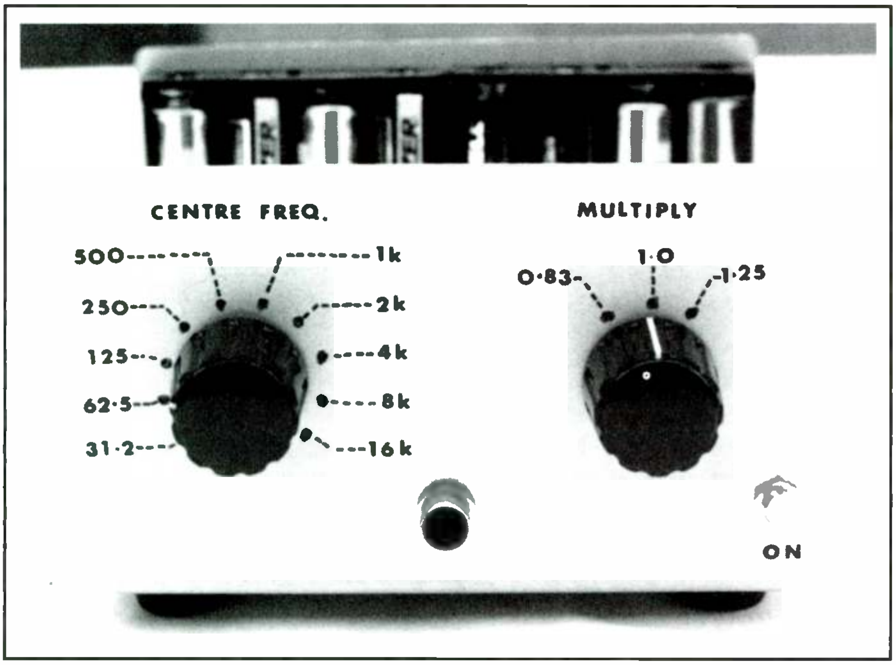


PHOTO 1: Front panel of the noise source.

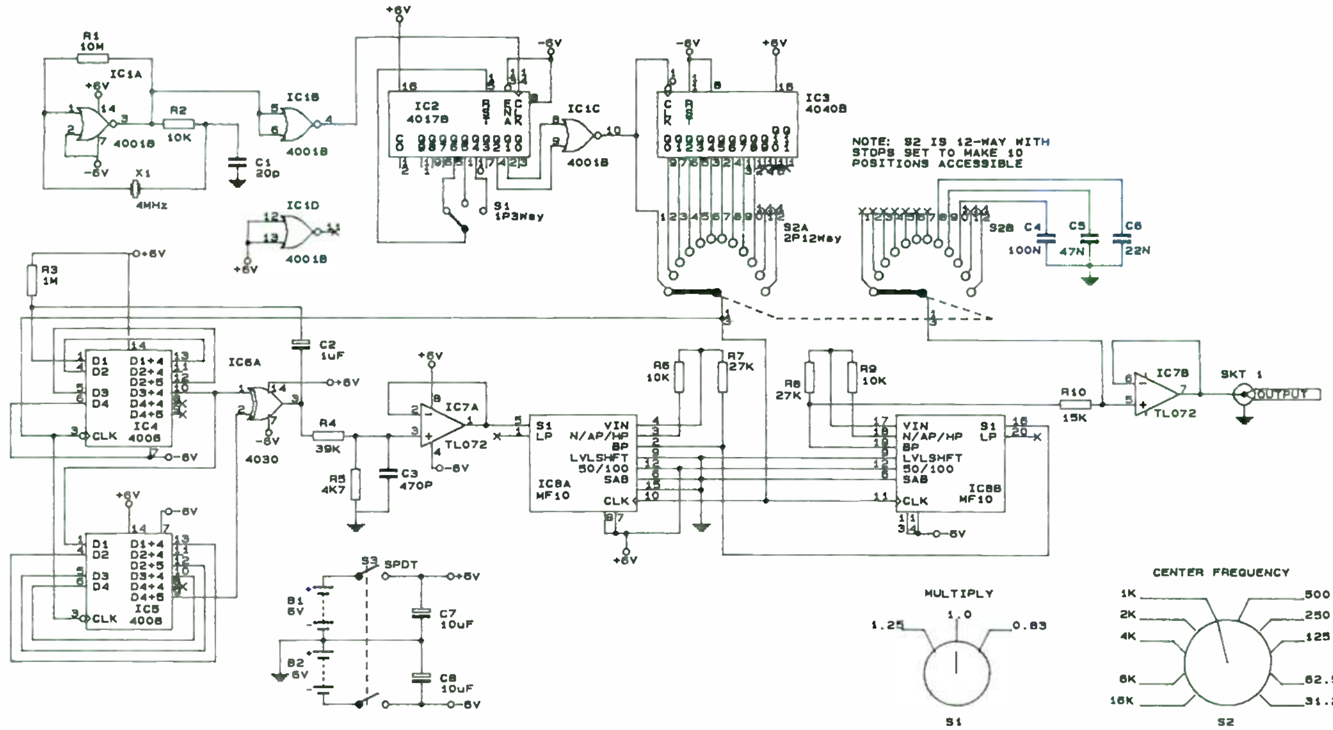


FIGURE 1: The ⅓-octave noise source.

noise output to increase, due to broadening of the passband.

To keep the filter-output constant when the center frequency is changed, pink noise is used as the source. Pink noise, which has a spectral density inversely proportional to frequency, is made by attenuating white noise at 3dB/octave throughout the audio spectrum.

The white noise is attenuated by a low-pass filter network. Attenuation at 3dB/octave is not a natural occurrence

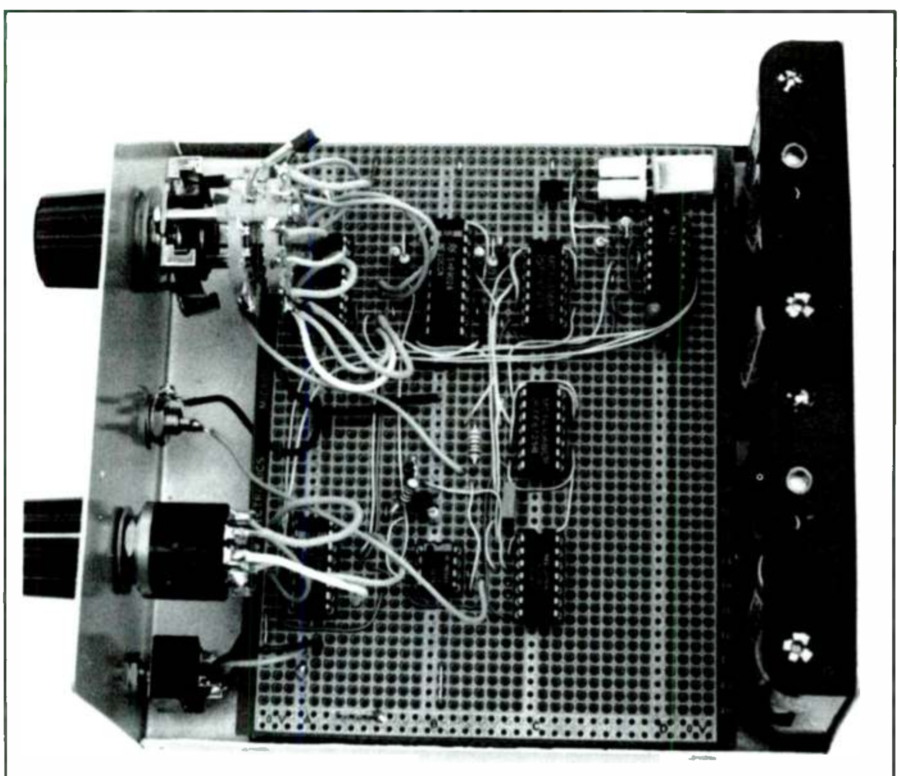


PHOTO 2: Inside the noise source.

TABLE 1

NOISE SOURCE PARTS LIST

QTY.	REF.	PART	COMMENTS
Capacitors			
1	C1	20P	8XAA
1	C2	1µF	
1	C3	470P	
1	C4	100N	
1	C5	47N	
1	C6	22N	
2	C7, 8	10µF	
Integrated Circuits			
1	IC1	4001B	
1	IC2	4017B	
1	IC3	4040B	
2	IC4, 5	4006	
1	IC6	4030B	
1	IC7	TL072	
Resistors			
1	IC8	MF10	
1	R1	10M	
3	R2, 6, 9	10k	
1	R3	1M	
1	R4	39k	
1	R5	4k7	
2	R7, 8	27k	
1	R10	15k	
Miscellaneous			
2	B1, 2	6V	
1	SKT 1	RCA JF	
1	S1	1-pole, 3-way	
1	S2	2-pole, 12-way	
1	S3	SPDT	
1	X1	4MHz	parallel resonant

All resistors 5% metal film, ¼W. Polarized capacitors—15V electrolytic. Nonpolarized capacitors—50V polyester.

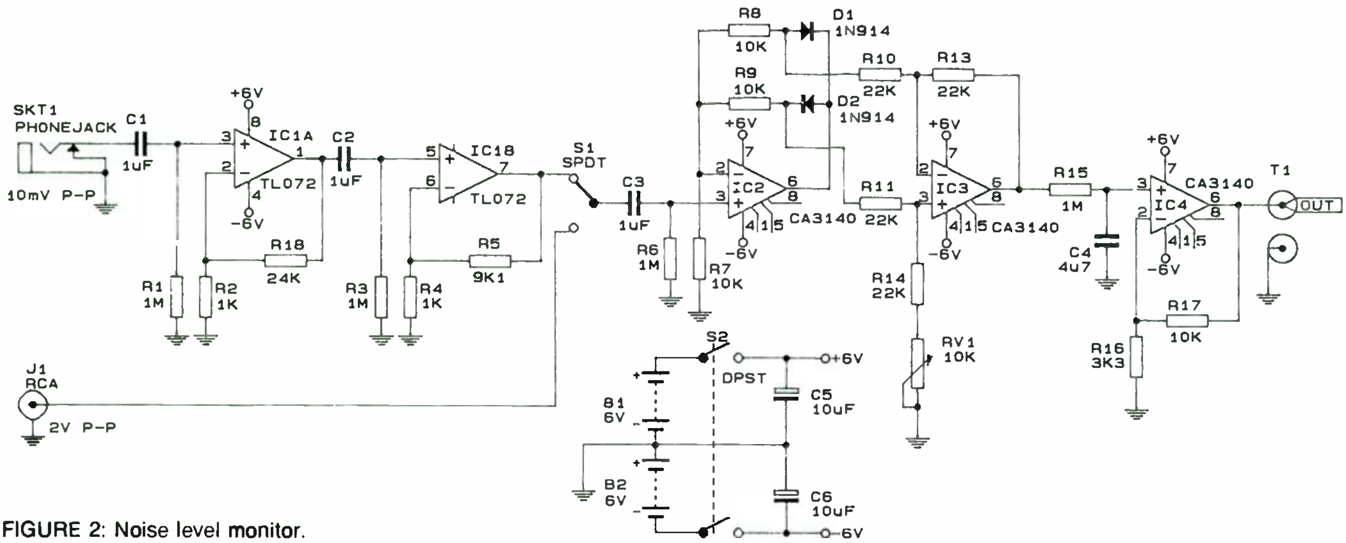


FIGURE 2: Noise level monitor.

in electrical networks, but suitable networks which approximate to 3dB/octave have been developed. Careful layout is required, because frequency components throughout the audio spectrum are present simultaneously, and the power density changes by 2^{10} across the spectrum.

When noise is not required simultaneously throughout the spectrum, which is when only a ⅓-octave band of noise is required, then there is no need for a pink noise source. A white noise source with variable output power is sufficient, provided the noise generator and filter are coupled, so that power density halves as the center frequency doubles.

The spectral density of noise from a digital random-number generator is directly proportional to the generator clock frequency. This relation between noise density and clock frequency occurs because the average power content of a random bit stream is constant; doubling the clock frequency doubles the noise bandwidth and halves the noise density. This type of variable-density noise source is used in the present design, coupled to a filter with a center frequency set by the same clock signal.

DESIGN OUTLINE. A switched capacitor filter (SCF) is used in its bandpass mode, with the passband set to ⅓-octave. The SCF center frequency is controlled by a clock input, which is set to 50 times the required center frequency.

A digital random-number generator is used as a white noise source; it's driven by the clock used to set the SCF center frequency. Use of one clock input, to control both center frequency and noise density, causes the filter to have constant power output. In effect, changing the clock frequency causes the white noise source to have the same power density as pink noise, at the filter frequency.

A crystal-controlled clock is used, to avoid the need for frequency measuring equipment. A level monitor is provided, to measure the noise source output, and a microphone amplifier is included for speaker measurements.

TABLE 2

NOISE LEVEL MONITOR PARTS LIST

QTY.	REF.	PART	COMMENTS
Capacitors			
3	C1–3	1μF	
1	C4	4μF7	
2	C5, 6	10μF	
Integrated Circuits			
1	IC1	TL072	
3	IC2–4	CA3140	
Resistors			
5	RV1, R7–9, 17	10k	
4	R1, 3, 6, 15	1M	
2	R2, 4	1k	
1	R5	9k1	
4	R10, 11, 13, 14	22k	
1	R16	3k3	
1	R18	24k	
Miscellaneous			
2	B1, 2	6V	
2	D1, 2	1N914	
1	J1	RCA	
1	SKT1	phone jack	¼" jack socket; 2-pole contacts closed
1	S1	SPDT	
1	S2	DPST	

All resistors 5% metal film, ¼W. Polarized capacitors—15V electrolytic. Nonpolarized capacitors—50V polyester.

NOISE SOURCE. *Figure 1* shows the ⅓-octave noise source. A 4MHz square wave is generated by IC1a.[1] The output of IC1a is counted down by IC2, which is a decoded 1 of N decade counter. Switch S1 sets N to 4, 5, or 6. When S1 is in its center position, N = 5, making IC2 output 800kHz. Switching S1 left or right from its center position, changes IC2 output to 640kHz or 1.2MHz respectively, which approximates to ±⅓-octave.

The output of IC2 is counted down further by IC3, which is a 12-stage binary counter. Switch S2 is set to select a system clock in the range of 1.562kHz–800kHz. The highest frequency setting is taken from IC2 output, and the other settings, descending in octaves, are taken from IC3 outputs.

The clock waveform applied to IC8 should approximate to a 1:1 mark/space ratio. The IC3 waveforms are 1:1, but IC2 waveform has a mark/space ratio 1:5, 1:4 or 1:3, depending on the setting of S1. To broaden the mark output from IC2, its outputs Q1 and Q2 are joined by IC1c, to form a clock waveform which has 1:2, 2:3, or 1:1 mark/space.

ICs 4, 5, and 6 form a random-number generator[2] except the clock frequency is variable. IC8 is the SCF.[3] It contains two separate two-pole filters, which are connected in series to form a four-pole bandpass filter. The Q value of each filter is set to 2.7 by a 27k:10k resistor ratio.

The ratio of clock frequency to center frequency is set to 50:1 by connection of IC8 pin 12 to +6V. Hence, with S1 central, the SCF center frequency ranges from 31.25Hz to 16kHz.

The SCF output is stationary between clock pulses, each clock pulse causes the SCF output to step to a new level, giving the SCF output a stepping motion. This motion generates a noise component at

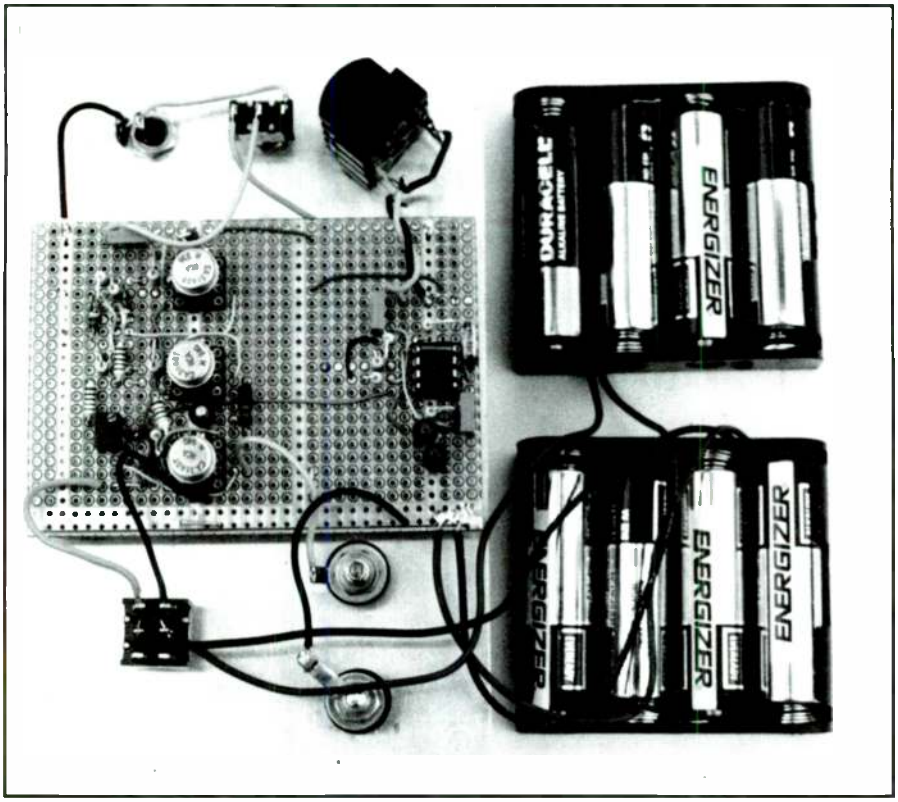
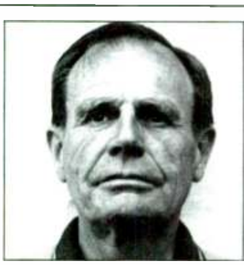


PHOTO 3: Underside of the level monitor panel and power source.

the clock frequency. The clock noise amplitude is 1/25th of the wanted noise in the SCF passband. When S2 is set to low frequencies, clock noise is in the middle frequency range where hearing is most acute. This makes the clock noise louder than noise in the low frequency passband; the sound is unpleasant.

Middle-range clock noise also limits measurement of speaker low-frequency output to about 26dB below the speaker midrange output; below 26dB, clock noise becomes predominant. For these reasons, the SCF output is passed through a low-pass filter on the three lowest settings of S2; this filter attenuates the clock noise. Builders may omit this filter if the sound of pure 1/3-octave noise has no aesthetic appeal.



About the Author

Roy Littler obtained his degree in electrical engineering from Manchester University, England, in 1955, after service in the Royal Air Force. Since graduation, he has pursued a career in electronic engineering and extends this to hobby interests in audio engineering. He is presently employed by the Australian Defense Department.

LEVEL MONITOR. The monitor is shown in *Fig. 2*. ICs 2 and 3 form a precision full-wave rectifier.[1] RV1 is set to make the rectifier have similar amplitude outputs for both positive and negative inputs. To make this adjustment, a 2V p-p sine wave is applied to the phono-socket input, and IC3 output is viewed on a cathode-ray oscilloscope (CRO). RV1 is then adjusted to make the rectified half-sine waves have equal amplitude. The rectified signal is averaged by a low-pass filter, which has a 4.7-second time constant. This choice of time constant is a compromise between smoothing low-frequency fluctuations while retaining tolerable waiting time for the output to settle when the center frequency is changed. IC3 provides gain to give a 0–2V output. IC1 is used as a microphone amplifier.

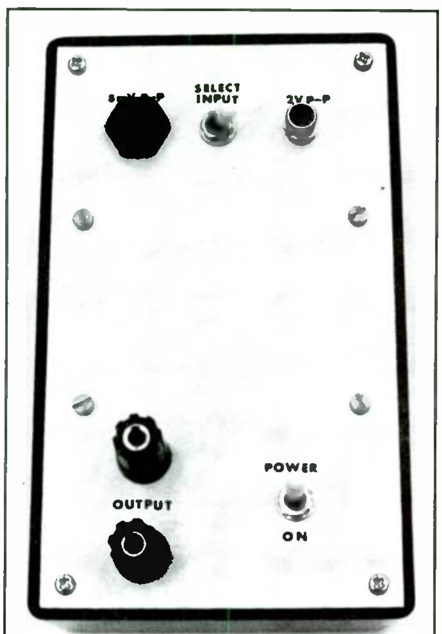


PHOTO 4: Panel of the level monitor.

TESTING. With the monitor connected to the noise source, measure the monitor's DC-output voltage using a moving-coil voltmeter. If the equipment has been assembled correctly, the measured output will be constant at about 1.8V on all frequency settings. On the lower frequency settings, the level monitor time constant is too short to completely smooth the random fluctuations. When you need high accuracy, take several readings at 5-second intervals and calculate their average value.

Having checked the equipment performance, you can take speaker measurements by connecting the noise source to the speaker amplifier auxiliary input and connecting a microphone to the amplifier in the level monitor.

IMPROVEMENT. I have made the design as simple as possible, using readily available components. This simplification has caused the MF10 SCF to be used below its full capability. By connection of pin 12 on the MF10 to ground, instead of +6V, the MF10 can be set to have its center frequency at 1/100th of the clock frequency, which gives the advantages of halving the clock noise and separating the clock frequency from the filter passband. If the MF10 clock frequency is raised, the crystal-clock frequency must be raised to 8MHz, and ICs 1 and 2 must be changed to CMOS HC-series ICs.

REFERENCES

1. Lancaster, Don. *CMOS Cookbook*, Howard W. Sams, 1988, pp. 279–281, pp. 399–400. (Available from Old Colony Sound Lab; see ads this issue).
2. Muller, Bernhard. "A Stereo Noisemaker," *SB* 4/84, pp. 16–20.
3. National Semiconductors, *Linear Databook*.

A HIGH-QUALITY SPEAKER CABINET

BY MARK FLORIAN

When my very first issue of *Speaker Builder* arrived, it contained a piece in Craftsman's Corner based on the Swan IV design (*SB* 3/90, p. 62) that immediately caught my attention. And as I read through subsequent issues, the Swan IV system's name kept appearing. Well, eventually my curiosity was aroused and I wrote to Madisound for a reprint of the article.[1] I was impressed with the design and decided to build a pair to replace my Large Advents.

While reading the article, I discovered that Part II unfortunately did not describe how to build the enclosures as pictured. I did some research, and after digging through several back issues of *The Woodworker's Journal*, I found an article that described how to build boxes by using a particleboard/veneer sandwich surrounded by hardwood on all the edges.[2] On the tops and bottoms, the hardwood is mitered at the corners. On the sides, it is placed along the long edge. The front and back fit between the sides and are held in place with wood biscuits. By resizing the cabinets and maintaining the same interior dimensions, I could adapt this carcass construction technique to the Swans.

In this article I'll describe some useful hints as well as some helpful sources. I'll explain how I modified the parts dimensions so everything fits properly and how the use of jigs can help save time. At the outset, I should mention that this is not a project for a beginning woodworker—it is far more involved than it appears.

Before purchasing any materials, I laid the whole thing out to scale on graph paper to be sure nothing was missing and the dimensions were correct. The various parts must fit exactly in order to get seamless joints between the hardwood and lamination. Merely tightening the clamps to close a poorly cut joint will not help, nor will lots of glue. It took about seven months of part-time work to complete, and the learning curve was steep the whole way. Several times I had to stop to do some research about a technique or jig before proceeding, halting progress until I could resolve the problem. In the end the tedious planning and research paid off but, as usual, it took far longer than I anticipated.

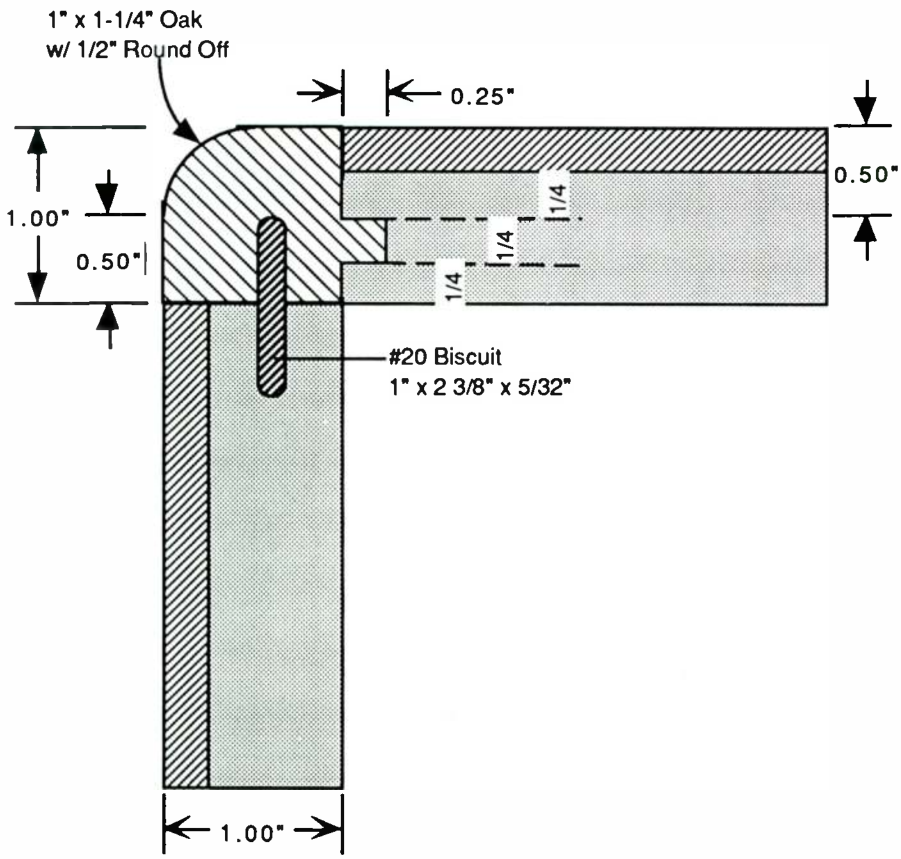


FIGURE 1: Front view showing side to top/bottom joinery.

CHISELS AND BLADES. First, I resized the cabinet pieces to account for the hardwood edging. This involved subtracting 2″ from the length of the front, back, and sides, 1″ from the top, and 1″ from the bottom, to account for the inch-thick hardwood. Similarly, these pieces were narrowed to account for the hardwood at the front and rear edges. The tops and bottoms were similarly modified so the internal volume did not change.

The corner joinery is detailed in *Fig. 1*. I used a tongue-in-groove joint to attach the hardwood to the particleboard/plywood lamination on the tops, bottoms,

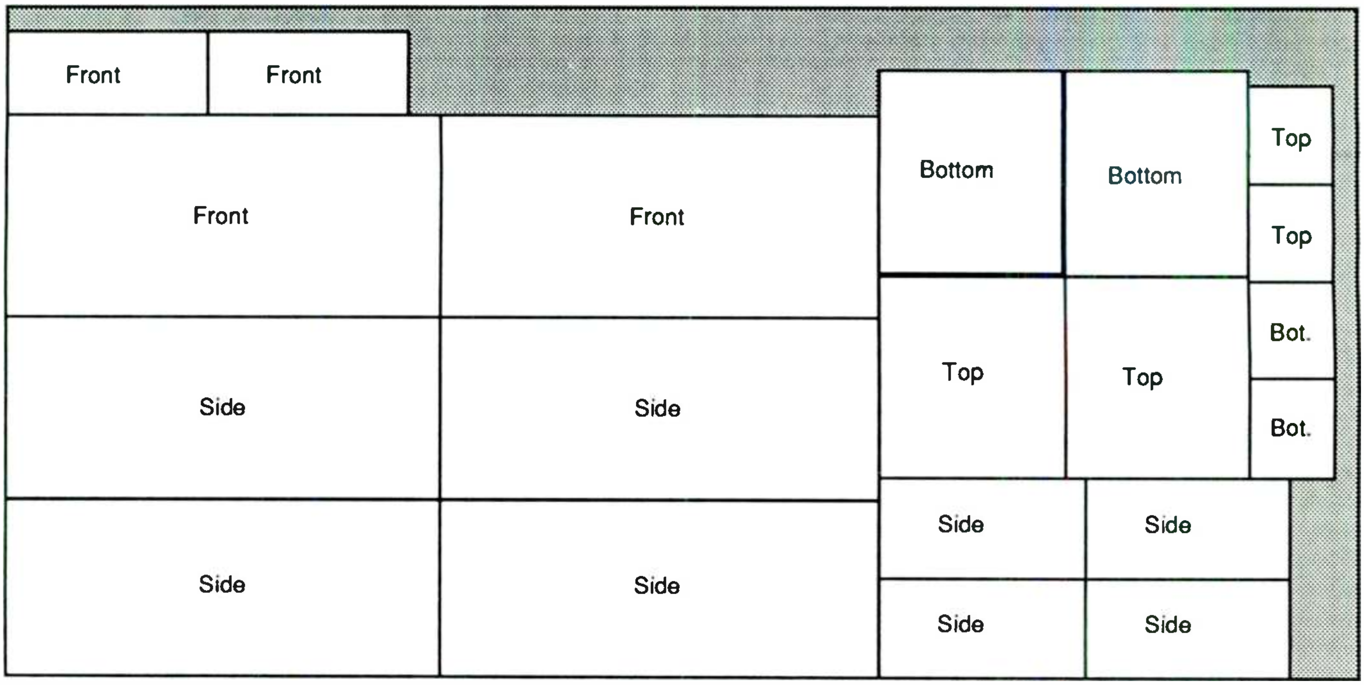


FIGURE 2: Revised cutting guide for the veneer.

TABLE 1

PARTICLEBOARD/OAK PLYWOOD DIMENSIONS

Required	Part	Satellite	Bass
4	Top, bottom	6"W × 7¼"D	14½"W × 13"D
4	Sides	14¼"H × 7¼"D	31¼"H × 13"D
4	Front, back	14¼"H × 6"D	31¼"H × 14½"D

and sides. This increases the strength of the joint and provides a larger gluing surface. The front and back pieces fit inside the two sides and are held in place with wood biscuits. The top and bottom are then attached to the open ends of the box, again with biscuits.

Since oak and particleboard are hard, dense materials and the tolerances required here are tight, I recommend that before cutting anything, you sharpen hand planes, chisels, and pencils. Michael Podmaniczky's article in *Bench Tools*[3] is helpful for learning how to sharpen chisels and plane blades.

Carbide-tipped saw blades and router bits are essential. High-speed steel is a waste of money when cutting oak and particleboard—it simply will not last very long. I used a 60-tooth carbide tablesaw blade when cross-cutting the ¼" plywood to prevent splintering. Ripping the oak and particleboard will go faster if you use a ripping blade instead of the fine-tooth type. I used a Dyanite 28-tooth combination blade. Though I have not tried one, Freud makes a thin-kerf ripping blade, the LU-87, which would reduce the load on your saw since the carbide tips are smaller and require less horsepower. Use saw blade stabilizers on your table saw for a better cut, especially if you use thin-kerf blades. These tools can be ordered from Woodworkers Supply (see Sources).

BLACK PAINT. With the right blades in hand, check your table saw to see that it is set up properly. Mark Duginske describes a step-by-step procedure for tuning up a tablesaw in *Power Saws and Planers: The Best of Fine Woodworking.*[4] This is critical to ensure that the cuts are perpendicular and square. With everything tuned-up and sharp, you're ready to start.

I used industrial-grade particleboard, though I later found that ¾" MDF (medium density fiberboard) was available and would probably be easier to staple, if you use staples to clamp the particleboard and plywood together while the epoxy sets. The cutting guide for the oak plywood and dimensions for all the pieces are shown in *Fig. 2* and *Table 1*, respectively. The cutting guide for the particleboard is the same as in the original article, with the pieces re-sized per

TABLE 2

OAK HARDWOOD DIMENSIONS

Required	Part	Satellite	Bass
8	Sides	14¼"	31¼"
8	Top & bottom width	8"	15"
8	Top & bottom depth	9¼"	16½"

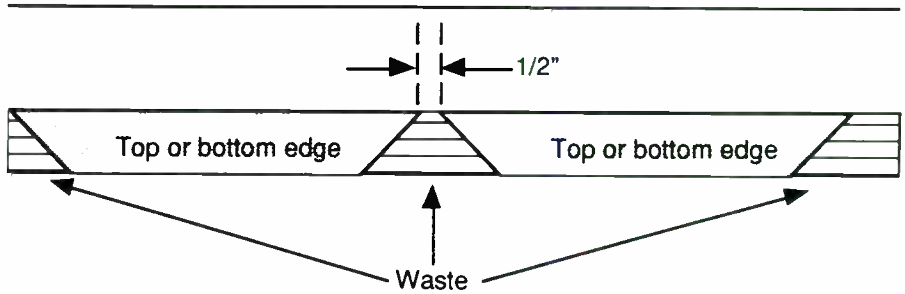


FIGURE 3: Layout of pieces on oak hardwood.

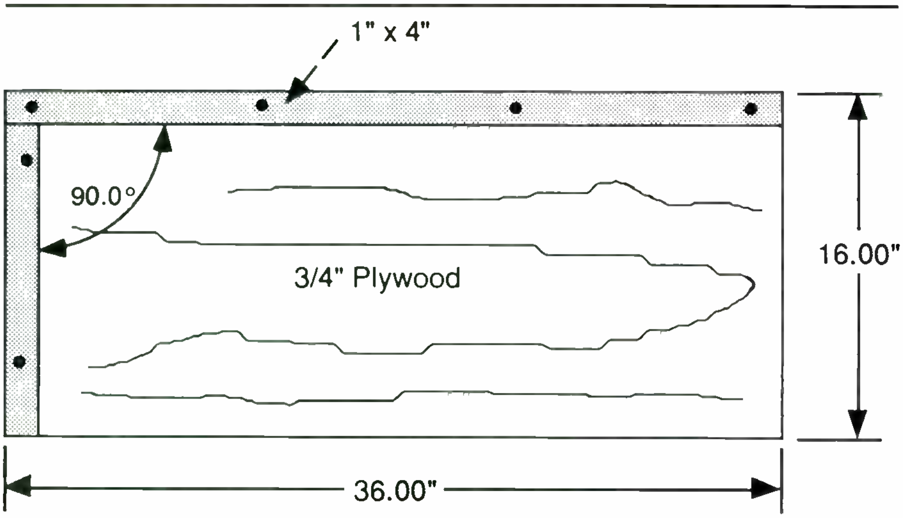


FIGURE 4: 90° jig used to align veneer plywood and particleboard.

Table 1. Note that four additional pieces are cut for the fronts and backs of the satellites and two more for the bass enclosure fronts. With these extra pieces, there was not enough room on the sheet of oak plywood to include the backs of the two bass enclosures, so I used some pine plywood I had and painted it black.

PRECAUTION. *Table 2* lists the oak hardwood lengths required. All pieces are 1¼″ × 1″. When you lay out these pieces to determine how much wood is needed, leave at least ½″ between each miter (as shown in *Fig. 3*). The miters could be reversed to save some wood, but since I used uncut lengths when cutting the tongue, I was unable to flip the pieces around. Working with shorter pieces would probably be easier.

The lumber store where I bought the hardwood sells it by the board-foot, instead of the commonly finished sizes such as 1 × 4 or 2 × 6, so I took along a tape measure. I was fortunate to find a piece that was approximately 1⅜″ × 7″ × 6′ from which I could almost get all the pieces I needed. I purchased another smaller board of the same thickness to allow room for the inevitable consequences of Murphy's Law and knots. All together, I bought about 63 board-feet of hardwood.

The lumber store ran the boards through their planer at a 1¼″ thickness. I then ripped the boards into strips 1″ thick on my table saw. A featherboard and hold-down device are helpful in keeping the board up against the fence and reducing the possibility of kickback. And always wear safety glasses or goggles when using power tools!

LONG STRIPS. Now is a good time to invite a friend over to show him what speaker building is all about. The uncut pieces of hardwood and particleboard are very heavy, and having someone there to help guide the board and strips is a big help. I later found a couple of undersize strips which caused problems at assembly time. Check the strips as you rip them for correct dimensions. You should now have several long strips that are 1¼″ wide × 1″ high.

I cut the plywood/particleboard pieces slightly oversize and later trimmed to final size after the epoxy cured. I cut all similarly sized pieces at the same time to ensure consistency. The plywood's veneer side should be *up* if you are using a table saw and *down* if you have a hand-held circular saw. The epoxy is fairly gooey and the first couple of times I tried to staple the pieces together, the plywood slid on the particleboard.

SAME WIDTH TRIM. To prevent this, I built a corner jig out of a scrap piece of ¾″ plywood onto which I nailed two pieces of 1 × 2 lumber at set at 90° using a framing square (*Fig. 4*). The jig helps hold everything square while you staple. Incidentally, I had to use ⅜″ staples. The ½″ staples recommended in the article bent when driven into the particleboard. As suggested, I cut popsicle sticks into sections about an inch long and drove the staples through them to prevent dinging the plywood.

Epoxy smears on the plywood can be cleaned up later with lacquer thinner. If the epoxy touches the veneer face of the plywood, it will inhibit the stain from penetrating, resulting in an uneven finish. If the spots are sanded well however, they are hardly noticeable. After gluing all the panels, I stacked them horizontally on the floor to allow the epoxy to set.

When the laminations have dried, pull out the staples using a small, sharp-tipped screwdriver with a small wood dowel as a fulcrum. Lift the staples just enough to grab them with a pair of pliers, to avoid marking the veneer. Next, trim the panels to their final width, putting the square edge set by the corner jig against the table saw fence. To ensure

Continued on page 18

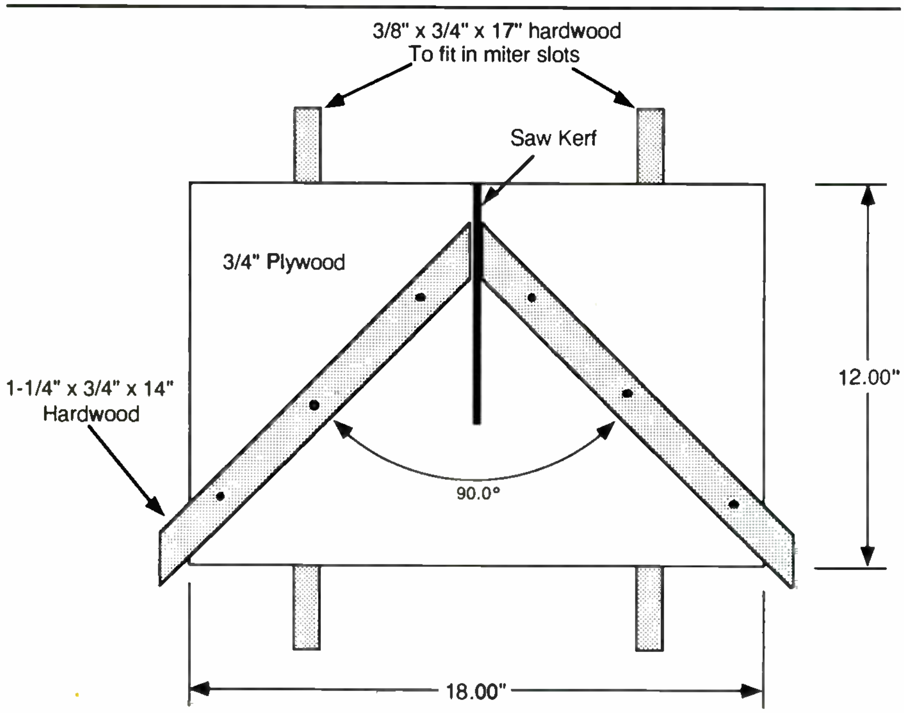


FIGURE 5: Miter jig for table saw.

ZALYTRON

YOUR ONE STOP SHOPPING CENTER

FOCAL. VIFA Cabasse LA REFERENCE EN HAUTE FIDELITE SCR

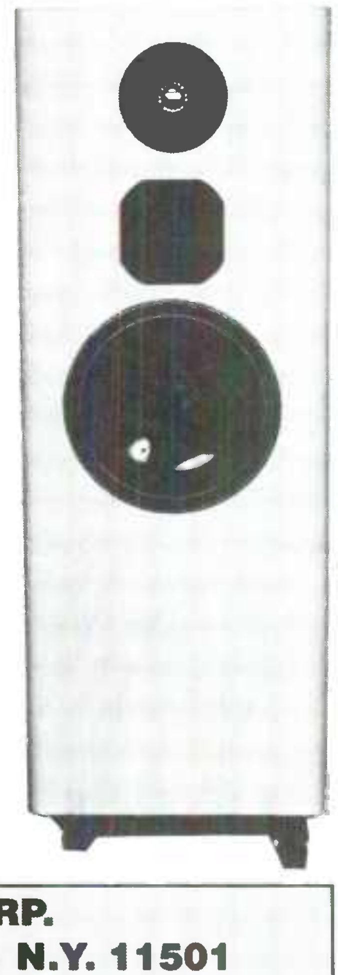
seas morel ZALYTRON CUSTOM WOOFERS PHILIPS

Polydax speaker corporation l'atelier audio FRANCE ACCUTON

Zalytron's Exclusive NEW! TPX SYSTEMS

This year Polydax introduced the new titanium tweeter model DT101 and the new TPX mid range MDA 100. However what I really felt we needed was a professional design to get the most out of these new drivers. So I spent the money myself to have Frank Luscher of the New York Audio Society design two systems around these drivers. Well the results were well worth the effort.

These systems must be heard to be appreciated. The tweeters and mids sound great and the bass is excellent. However the secret is in the crossover and the cabinet construction and design. Without Mr. Luscher's special crossovers you cannot get these drivers to sound right. Without the strong Zalytron cabinets the bass is gone. The mid range also needs a special sub enclosure design to sound good. Thus when you buy the system at Zalytron you are assured of success. These are our own exclusive designs and are only available at Zalytron.



WRITE OR CALL FOR THE LATEST CATALOG INFORMATION

ZALYTRON INDUSTRIES CORP.
469 JERICHO TURNPIKE, MINEOLA, N.Y. 11501
TEL. (516) 747-3515 FAX (516) 294-1943

Our warehouse is open for pick-up 10AM to 6 PM daily, Monday thru Saturday
UPS orders shipped same day • Minimum order $50.00
FOR ALL YOUR SPEAKER NEEDS OR INQUIRIES - WRITE OR CALL ASK FOR ELLIOT

Fast Reply #GG1131

Continued from page 16
consistency, trim all pieces of the same width without moving the fence.

JUGGLE STRIPS. I used my router to trim the length of the bass enclosure pieces with the aid of an edge guide depicted on p. 212 of Patrick Spielman's book, *Router Jigs and Techniques.*[5] They are too long to cut accurately on a table saw with a miter gauge. I considered building a shooting board, which is like a sled that fits on a table saw, but a friend suggested we try his radial-arm saw. Unfortunately, we were unable to get a perfectly square cut, even after repeated adjustments. The router did the job accurately, however, and did not splinter the veneer.

A quick, reliable check for square is to measure the diagonals from one corner to the opposite one. If the two are the same, the panel is square. Make sure the edge of the veneer plywood and particleboard lie in the same vertical plane. Otherwise, the joints will not fit tightly and a gap will result. Check all the pieces with a small try square and recut any that are not true. Place all same-size panels atop one another. They should all have the same dimensions. If not, something is amiss and must be corrected before proceeding. Group each speaker's pieces together and label them.

Inspect the hardwood strips and mark any sections with loose knots. Small, tight knots can be positioned so they are not exposed. Select the best face on each piece and mark it. Next, lay out the pieces on the oak with a pencil, leaving at least ½″ between miters as shown in *Fig. 3.*

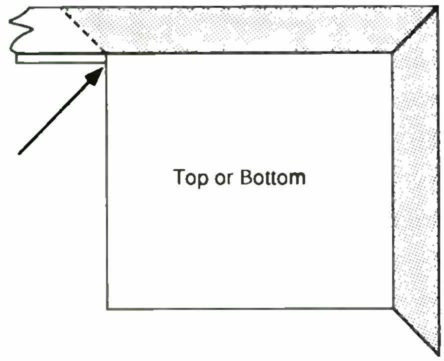


FIGURE 6: Mark oak trim at inside edge for correct length.

You will probably have to juggle them to minimize wasting wood and to work around the marked-out knots.

TIME SAVER. Next, the 6′ pieces were cut into smaller sections for easier handling. Those with radial-arm saws will have an easier time than I did. I improvised on my table saw by attaching a 1 × 4 × 8″ piece of maple with a couple of wood screws to the miter gauge on my table saw. I used maple because I already had a piece, but any sqaure-edged wood will do. Glue a piece of 100-grit sandpaper to the face to prevent the oak from creeping down the fence when cutting. A couple of hand clamps are also a good idea to hold the oak tight against the fence so your hands are out of the way.

Next, tongues were cut into the hardwood using a router table. As a reference, Patrick Spielman provides plans for an excellent router table in his book. It is a time-saver when making repetitive cuts since you set up the fence only once ensuring accuracy.

As shown in *Fig. 1*, the tongue is ¼″ wide and starts ¼″ from the bottom of the oak facing. I actually reduced the tongue to 7/32″ wide so it would slide more easily into the groove cut in the particleboard. This made aligning the mitered joints at the corners of the tops and bottoms easier. My router table is set up using an Incra-Jig, which allows the fence to be moved in exact 1/32″ increments. It saved me a lot of time in measuring.

SEAT THE JOINT. To ensure that the wood stays in constant touch with the router table and fence, use a featherboard clamped to the table to hold the hardwood against the fence and a small block above it spring-clamped to the fence. Run all the pieces through on one setting and then reset the fence to remove the remaining 17/32″ on the other side, leaving the tongue. You will now have several pieces of hardwood with a tongue cut along one side.

Now cut the groove in the side panels on the router table using a ¼″ carbide straight bit. Be sure to put the particleboard side of the lamination against the fence! Cut a groove in a piece of scrap particleboard and check it with a piece of the oak edging. When they fit together, the tops should be even or the oak should be a little higher. The latter can be fixed with a router jig that I will describe later. When the adjustments are correct, run all the side panels through on their long sides only; and the tops and bottoms on all edges so they are grooved all the way around.

Next, assemble the side panels with the corresponding oak pieces. If the oak piece happens to be on an end, then you have only one cut to make. Otherwise, cut the piece free from a longer strip at either of its ends but leave the other one long. Fit a side and oak piece together and make it flush on one end. Tap the oak edging into the side with a hammar and wood block to fully seat the joint. There should be no gaps. If it refuses to fully close, mark it and correct it with a wood chisel or rasp.

STRESS TEST. If a tongue is too deep, preventing the joint from closing, remove some of it with a small block plane. Any small lips left on the inside face of the edging where it joins the tongue can be eliminated with a sharp wood chisel.

When the joint closes tightly, mark the other end of the edging so it is flush with

Continued on page 20

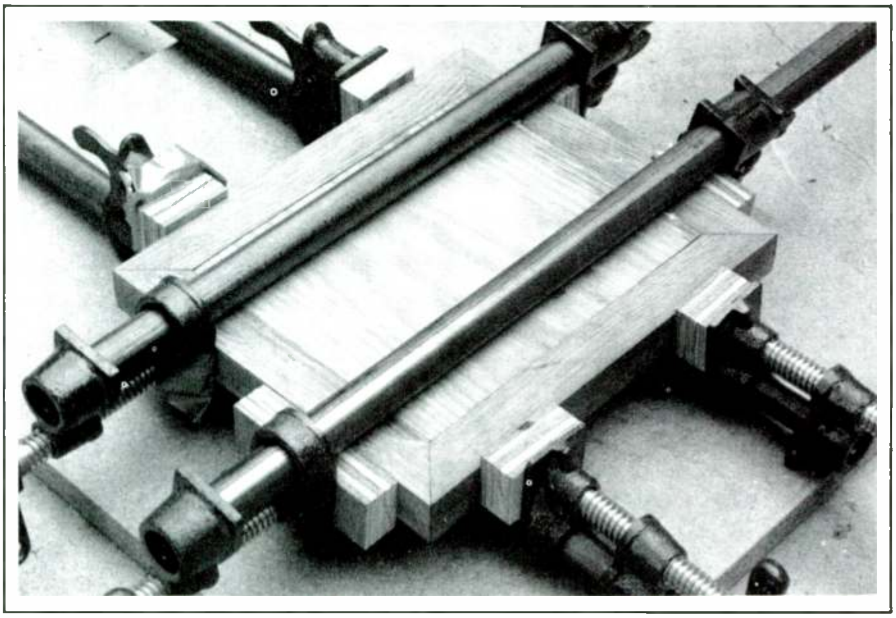
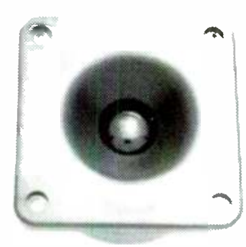
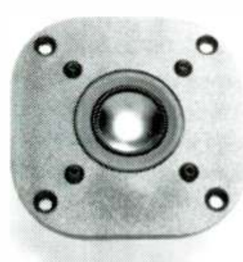


PHOTO 1: Alternating the pipe clamps helps prevent warping the panel.

Transducer Technology

Bringing you the finest from around the world...





4352 J Spring Valley, Dallas, Texas 75244 **(214)490-9933 FAX (214)404-1806**

Fast Reply #GG401

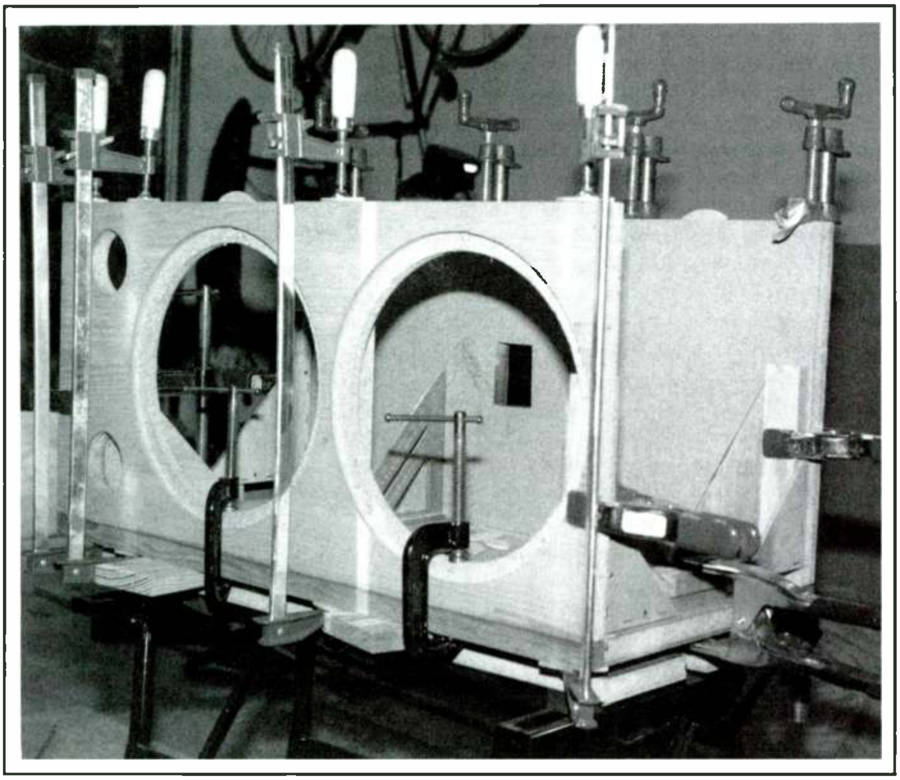


PHOTO 2: Two corner jigs are used at each end of the bass enclosures to hold the front and back square. Note the use of C-clamps to prevent distorting speaker cut-outs.

Continued from page 18

the length of the side and cut it. Repeat for the other side. I chose to measure right on the wood, instead of with a ruler, to minimize errors. Mark these two pieces of oak and their respective sides and repeat for the remaining seven sides. When finished, you will have eight sides with two pieces of oak edging per side, for a total of sixteen.

Now we glue the oak edges to their sides. Although the epoxy is necessary for the panels, it is too messy for the joinery work. I considered several types of glue and did a little stress test.

COMING TOGETHER. Using yellow wood glue, I glued a scrap piece of particleboard edge to edge with a ¾″ piece of oak making a simple butt joint (no tongues or biscuits), and clamped it overnight. The next day I clamped it to the side of my workbench, with the joint right on the edge. Using my hammer I could not break the joint. Consequently, I decided that wood glue, with the additional strength from biscuits and tongue-in-groove joints, would hold up quite well.

Next, coat the particleboard groove with wood glue and let it sit until tacky. Then coat the oak piece. Recoat the particleboard edge with glue and assemble the two, tapping with a hammer and wood block to close the joint. Make sure the two ends are flush. Repeat for the other side and then clamp. Use two bar or pipe clamps for the satellite panels and four for the bass panels. Wood biscuits are handy as clamp pads to prevent marks on the oak.

If you use pipe clamps, cut a couple of 1″ sections of foam pipe insulation and fit them over the pipe to avoid marking the veneer. This prevents them from making contact with the veneer. Alternate the clamps, one on top, the next on bottom, to even out the force. Tighten securely, but do not overdo it or you risk warping the panel. With a clean, damp rag, wipe up any excess glue. Glue up as many as possible, depending upon your supply of clamps, and let them sit overnight.

USEFUL TIPS. Next, cut the miters on the remaining oak pieces. While accurate miters can be obtained using a 45° plastic triangle to set the miter gauge, I made a miter jig on a shooting board that slides on my table saw (*Fig. 5*). This allows each miter to be cut on its respective fence: the left miter on the left fence and vice versa. This results in a better joint in less time when the two pieces are joined. *Table Saw Techniques* by Roger Cliffe includes full instructions for building the jig.[6]

To assemble the edging around the top, start with one end of the oak, placing the tongue towards the fence, and cut a miter. Position the mitered piece into the groove of its corresponding lamination and place it so that the inside edge is flush with a corner. Mark the oak piece to the correct length on the inside edge at the other corner (*Fig. 6*). Cut it a little long at first.

Work your way around the lamination, again measuring directly as you go. When all four pieces have been cut, assemble them around the lamination and check the joints. Trim those that are a little long, rechecking each one. This is tedious work, but if the cuts are accurate and the lamination square, you should be able to put all the oak around a top or bottom without any gaps in the miters. If you make a mistake, you can use the piece for a bottom where it will not show.

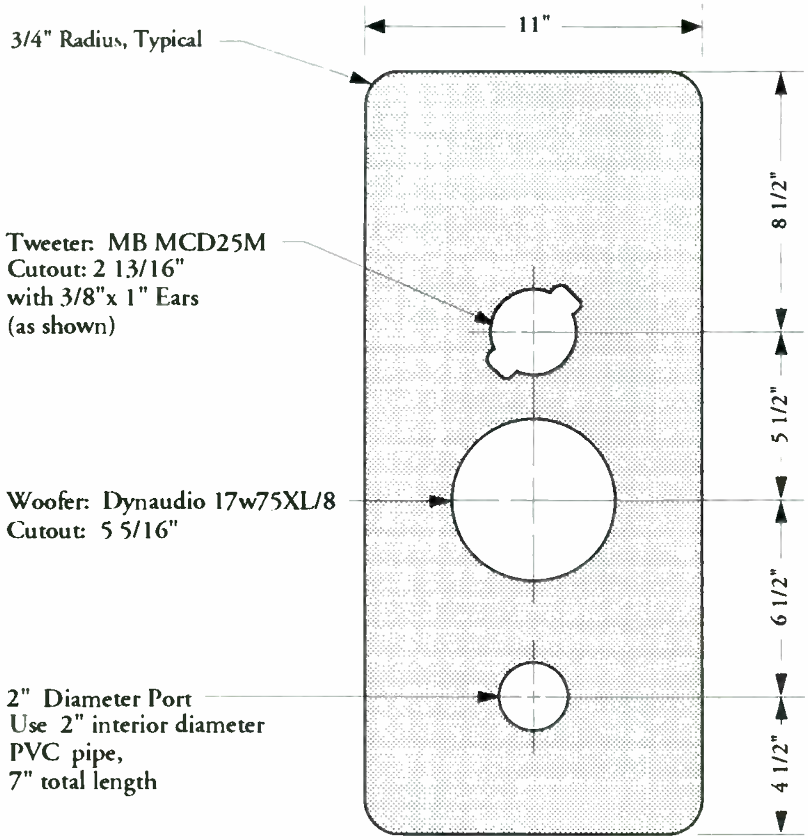
Extra oak is a good idea in case you cut one too short. If the oak is slightly higher than the lamination, don't worry. It can be trimmed later with the router. Work carefully and take your time. Glue the oak facing around the lamination using the method described above. A band clamp is useful to hold everything in place while the glue sets. I used four short pipe clamps as shown in *Photo 1*. Repeat for the remaining seven tops and bottoms.

A NIFTY JIG. Lay out the mounting holes for the drivers and ports per the original article. I used a circle-cutting jig (Black & Decker) on my router and a ¼″ straight bit. This works so well that the port tubes fit snugly into their holes, requiring only silicone sealant around the joint inside to ensure air tightness. Be sure to rout out the recesses for the tweeter and woofers *before* cutting the holes for the drivers when using the circle-cutting jig. In order to protect the tweeter dome, I used an Avery 4″ × 3″ self-adhesive label stuck over the front to temporarily seal it and prevent dust from getting inside. This was also useful when it came time to trim the tweeter flange. I cut the Focals' recesses freehand with the router and made adjustments with wood chisels.

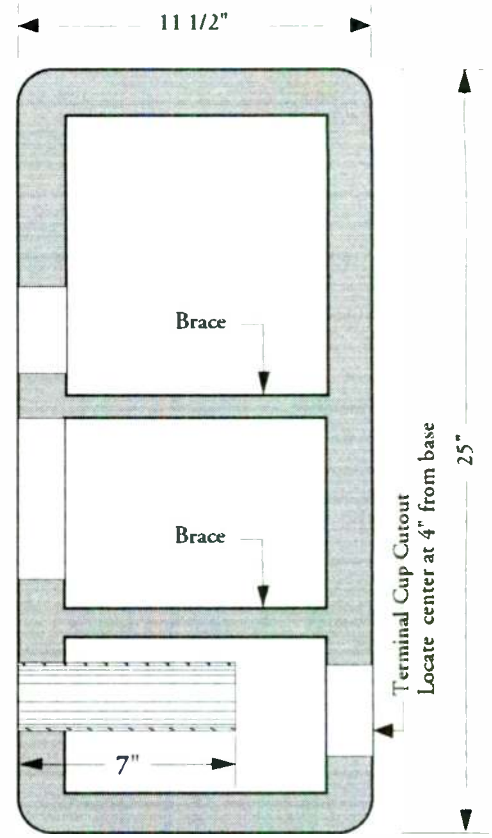
If I were doing this project again, I would build a template, using the appropriate bearing guide in my router. Doing recesses free hand took three times as long as following a template. Since you must do this task four times, a template can be a real time saver. Spielman explains how to use templates in his book.

Also, I would mount the ports on the rear panel in another version. Jim Bock

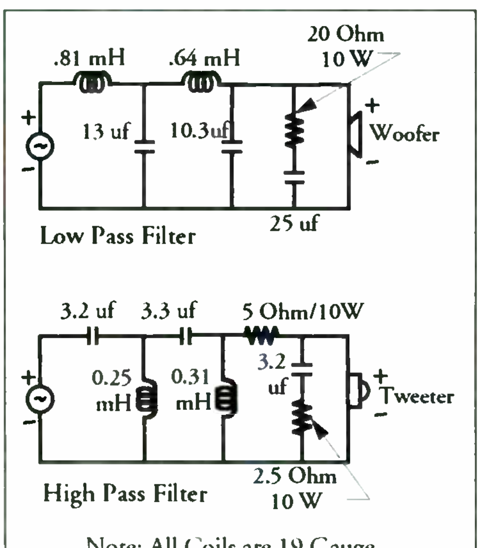
Continued on page 22



FRONT BAFFLE

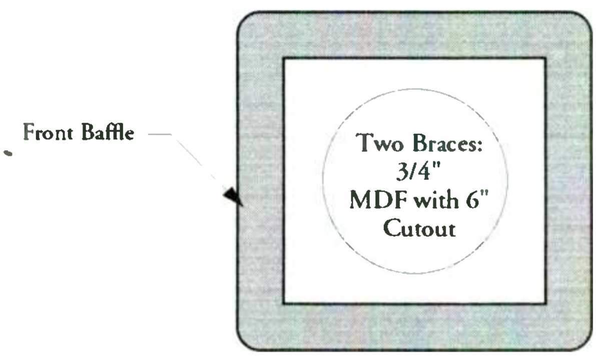


SIDE SECTION



CROSSOVER NETWORK

Typical Cabinet Construction: Two Layers
3/4" Medium Density Fibreboard (MDF)



TOP SECTION

The Black Dahlia Mk II

Designed by Dick Olsher

A&S SPEAKERS

3170 23rd Street; San Francisco, CA 94110
Telephone: 415-641-4573 Fax: 415-648-5306

Continued from page 20
says he has built boxes both ways with no difference in the sound. Finally, either drill holes or cut out a piece in the rear panels for the binding posts.

Now, you should have a top and bottom with oak edging, two sides with oak edging, a front with holes cut for the drivers, and a back.

Spielman's book offers a nifty jig for flush trimming on page 254 which can be adapted to any router. This allows you to trim the oak edging perfectly flush with the veneer, without damaging it. It would take a long time to do this with sandpaper, so luckily it is a snap with the router. I shortened the jig to fit on the small tops and bottoms of the satellites. When using it, keep constant pressure on the jig base to prevent the router from tipping and gouging your work. If you use this method, practice on a piece of scrap to get a feel for the proper pressure. You will be amazed at how well it works and how quickly it accomplishes the task.

BEARING GUIDED BIT. Now comes the fun part: biscuit joinery. Biscuits (sometimes referred to as wafers) create strong, self-aligning, easy to make joints. When the two pieces are glued and clamped, the biscuits will swell as they soak up the glue, resulting in a tight joint. A three-wing 5/32″ slot cutter and a bag of biscuits are all you need. (I used size #20.) Both can be ordered from Woodworkers Supply or at your local lumber or hardware store.

The process is simple. Starting with the satellites, assemble the front, back, and two sides, holding everything together with clamps. Remember the front and back fit between the sides. Align the pieces until the joints are flush and, using a combination square, draw a straight line across the joint on both pieces. I used three biscuits on the satellites and five on the bass enclosures, spaced equally along the length (height) of the fronts and backs. Continue until all four edges are marked. The lines will enable you to reassemble the pieces in the same order.

Set the slot cutter depth so it cuts a slot in the center of the particleboard, about ⅝″ down from the veneer surface. Since the slot cutter's diameter is smaller than the diameter of a biscuit joiner's blade, you must enlarge the cut a bit on each side of the line. Move the router a little to the left and right of the line. The slot depth is preset since the bit is bearing-guided.

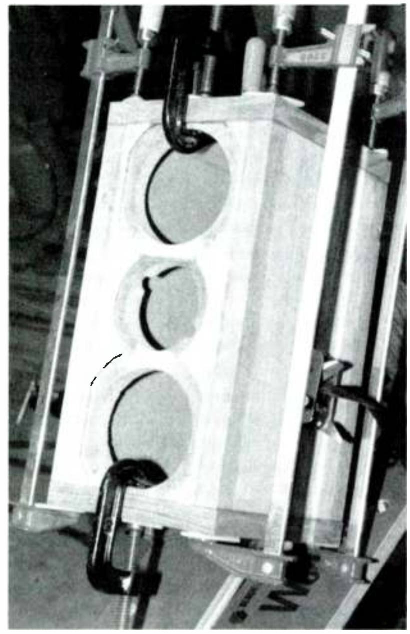


PHOTO 3: Six clamps are used to hold the top and bottom in place while the glue sets. Notice the band clamp between the two bar clamps.

PLYWOOD TRIANGLES. I clamped the sides vertically in a Black & Decker Workmate® to cut the slots, centered on each line. Clamp the front and back flat to a workbench when cutting their slots. After you have finished cutting, assemble the box with biscuits following the pencil lines as a guide. Use the glue technique described earlier and put glue into the slots on both sides.

Assemble the parts as follows: Place one side horizontally on a bench and insert the biscuits in the glued slots on both edges. Then add the front and back, aligning the slots with the biscuits. Next, put biscuits into the glued slots on the front and back and place the other side down on top of this. Adjust the pieces until both ends are flush—this is essential for the top and bottom to fit properly. Make sure the box is square, and clamp it. Repeat for the remaining three boxes.

In assembling the bass enclosures I had difficulty keeping the fronts and backs square with the sides while I clamped the box. I would get all the clamps on, only to check for square inside the box and find a corner out of square. Trying to do both sides at once is difficult. A friend suggested I build triangular corner jigs out of scrap 1 × 2s and ¼″ plywood to hold the front and back pieces squarely while I put the clamps in place. I set the 1 × 2s at a 90° angle using a framing square and stapled the plywood triangles to the 1 × 2s. If you make one of the plywood triangles smaller, you can use spring-clamps to hold the jig securely in the corner.

I made four of these and they really helped. On the bass cabinets, I found that it is easier to position one side, glue its edges, install the biscuits, and then position the front and back using my corner jigs. Next clamp the front and back to the side with pipe clamps as shown in *Photo 2*. Remember to use only enough pressure on the pipe clamps to close the joint, and then a little more. When this dries, spread glue on the remaining edges for the other side, insert the biscuits, and then clamp the remaining side.

PREVENT A CRACK. By now, light is beginning to appear at the end of the tunnel. Fit a top or bottom to one of the satellite boxes. Check around the edges to make sure it is flush on all sides. If the box or the top is not square, then make some adjustments. When all is aligned properly, use the square and draw pencil marks for the biscuit slots. I used one biscuit in the middle of the front and rear of the satellite and two equally spaced for each side. On the bass enclosures, I used three per panel.

Cut the slots, glue the edges, and install the biscuits. On the satellites, I used a bar clamp on each corner and a band clamp around the middle (*Photo 3*). The bass enclosures are really heavy and difficult to move. I positioned them vertically on the Workmate and opened the jaws on the working surface so that it was as wide as the cabinet. To place the bass cabinet at a comfortable working height which allowed it to move around, I positioned the pipe clamp slider so it engaged underneath the Workmate's surface (*Photo 4*).

Seven pipe clamps are ideal for this task: one at each corner and one in the middle of each side and at the back. The

About the Author

Mark Florian received a B.S. in Electrical Engineering from the University of Texas at Austin in 1983, and has been involved in electronics since disassembling a family TV set at the age of seven. When not building something or taking things apart, he enjoys tennis and catamaran sailing.

bass enclosure and (especially) the satellite front panels are particulary fragile because of the large cutouts. An overtightened pipe clamp in the middle of one would probably crack something. Therefore I used a large C-clamp or a bar clamp in the middle to glue the fronts to either the top or the bottom.

COOL OFF TIP. The Focal midranges will fit in their recesses only if part of the satellite-oak edging is removed. I placed a driver on its face, traced its outline, and used a sharp wood chisel to make the recess. Another option is to increase the height of the satellite so that ¼″ or so exists between the top of the midrange flange and the oak edging. This would increase the volume of the box a small amount and make recessing the midranges easier. I am not sure what effect this would have on the sound, but I believe it would probably be minimal.

Last, I drilled four holes to accommodate #10-24 T-nuts into the bottom of each bass enclosure near the corners to insert spikes. An appropriately sized stop-collar or a small piece of tape wrapped around the bit will avoid drilling too deep. I made spikes out of hardware store hanger bolts. One end of these has wood threads, which I ground to a point using a bench grinder, and the other has #10-24 machine threads to fit the T-nut.

A T-handle tap wrench, from a tap and die set, is perfect for holding the hanger bolt. You can easily shape the opposite end of the bolt to a sharp point, since you can rotate the whole thing in your hands. This also allows you to safely handle the bolt, as it becomes very hot from grinding. Do not allow the tip to overheat, or the temper of the metal will be ruined. Periodically, hold the tip in the breeze created by the grinding wheel to cool it. I used a flat washer, lock washer and #10-24 nut to securely tighten the spike into the T-nut, firmly anchoring it to the enclosure.

TRY STAINS. I used two methods to attach grille frames to the enclosures. For the bass cabinets, I drilled 9/16″ holes at each corner and in the middle, along the edge, to accommodate a ball-and-socket fastener. The ball end of the fastener is mounted on a short piece of ½″ dowel, with the other end mounted on the grille frame. This positions the frame about a ½″ from the cabinet's front panel to minimize diffraction.

Since these fasteners would be too large for the satellites and detract from their appearance, I drilled a ¼″ hole in each corner and used ¼″ dowels which were painted black and then glued to the frame. The length determined allowed for a ½″ stand-off to match the bass enclosures. The frame in both cases is ¼″ hardboard, with as much material removed as possible, covered with black speaker cloth from Radio Shack.

FINISHING. I rounded off all the oak edges using a ½″ roundover bit and sanded all the surfaces using a finishing sander, starting with 100-grit sandpaper and working up to 220. Next, I vacuumed out the cabinets and wiped all external surfaces with a tack rag to remove any last traces of dust. For finish, a couple of coats of Minwax Golden Oak stain was used, which looks really nice. Incidentally, the circles from the driver cutout are a good place to try different stains.

Having chosen to separate the tweeter and midrange crossovers to minimize crosstalk, I wired the crossovers and drivers using four lengths of #24-gauge Kynar-insulated wire-wrap wire for an equivalent gauge of #18 (silver-plated OFHC wire available from Interstate Wire Co., Inc.). I used ten pieces of #24 for an equivalent gauge of #14 in the bass units. I did not braid the wires because of the time required, but I did find that when twisted together, the wires stay in place.

After reading about the Litz concept in POOGE-2 (*TAA* 4/81, p. 10), I decided to give it a try. Finally, I lined and stuffed the cabinets as in the original construction article.[1] I soldered all connections and used silicone sealant to seal the drivers into the cabinets. Black trim wood screws secured the D-28 and T-nuts hold woofers and midranges aided by #8-24 × 1¼″ machine screws. A cordless drill/driver with an adjustable clutch was very useful for this operation.

The satellites made their first sounds during October 1991. I am amazed by how much more I hear from them than from my old Large Advents, which served for the past 11 years. Barbra Streisand's voice on the *Broadway* album is so rich in detail I would swear

Continued on page 85

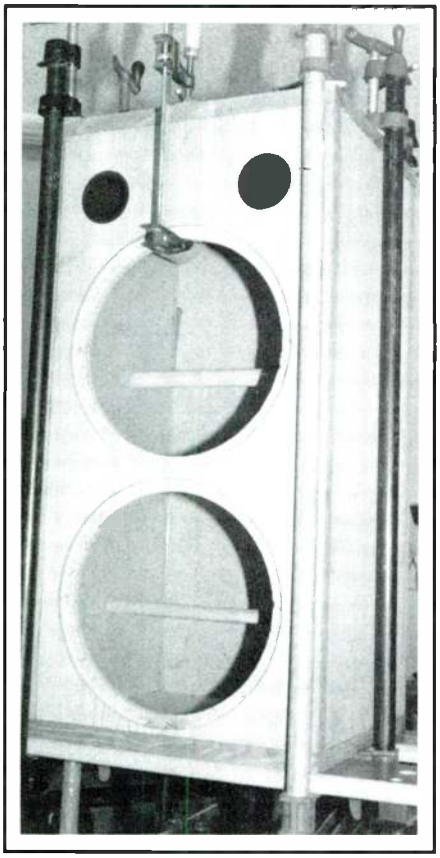


PHOTO 4: The completed bass enclosure. The surface of the Workmate is sandwiched between the top and the clamp to aid in assembly and prevent tipping. Side clamps fit in between the Workmate tops.

ACKNOWLEDGEMENTS

I offer special thanks to my friend Bert Jones for his help and guidance and to Jim Bock for answering my many questions. And thank you, also, Messieurs D'Appolito and Bock, for designing this system and publishing it.

SOURCES

Woodworkers Supply of New Mexico
1108 North Glenn Road
Casper, WY 82601
(800) 645-9292
Saw blades, router bits and biscuits, Incra-Jig

Meniscus
2442 28th St. SW, Unit D
Wyoming, MI 49509-2158
(616) 534-9121
Drivers, crossover components, grille fasteners

Interstate Wire Co., Inc.
10820 Sanden Dr.
Dallas, Tx 75238
(800) 442-0073 (Texas)
(800) 527-0010 (Elsewhere)
Kynar-insulated wire, spiral wrap

REFERENCES

1. D'Appolito, Joe, and James Bock, "The Swan IV Speaker System," *SB* 4, 5/90.
2. "Modular Coffee Table and Bar," *The Woodworker's Journal*, Jan./Feb. 1985, pp. 50-53.
3. Podmaniczky, Michael S., "Sharpening with Oilstones," *Bench Tools: The Best of Fine Woodworking*, Taunton Press, 1990, pp. 105-109.
4. Duginske, Mark, "Tuning-Up Your Table Saw," *Power Saws and Planers: The Best of Fine Woodworking*, Taunton Press, 1990, pp. 71-75.
5. Spielman, Patrick, "Router Jigs and Techniques," Sterling Publishing Co., 1988.
6. Cliffe, Roger, *Table Saw Techniques*, Sterling Publishing Co., p. 77.

BI-AMPING THE SAPPHIRE II SUB-1 SYSTEM

BY GARY GALO
Contributing Editor

Since I reviewed Audio Concepts' Sapphire II satellite speakers (*SB* 6/90, p. 56) and the Sub-1 subwoofer (*SB* 3/91, p. 51), the Sapphire II/Sub-1 combination has become my reference loudspeaker system. My opinions expressed therein have been reinforced many times since. These loudspeakers still offer unsurpassed performance in their price range, and easily outperform many systems costing much more.

DEDICATED AMP. I've been a believer in multi-amplification for over a decade. My old transmission line loudspeaker systems were tri-amplified for many years, with superb results. Over the past two years, I've become an advocate for satellite-subwoofer systems. Keeping the tweeter and midrange drivers completely isolated from the vibrations of the woofer and its enclosure results in greater midrange clarity and detail. Of course, you won't reap the full benefit of separate woofer enclosures if you simply place the satellites on top of the subwoofers. You really need to isolate the satellites by placing them on separate stands. I believe satellite-to-subwoofer crossovers should be designed to allow this flexibility in placement. Audio Concepts has done this with the Sub-1 crossover. I have my system set up with the Sub-1s right behind the Sapphire IIs.

As fine a system as this is, I believed that even greater potential could be realized with multi-amplification. Fortunately, it's neither necessary nor desirable to pursue tri-amplification with this system. First, the Sapphire II's passive crossover characteristics would be difficult, if not impossible, to simulate actively. Second, because of the relatively high crossover frequency, Audio Concepts was able to use the highest quality passive components. The physical sizes of the inductors and capacitors aren't a problem, and premium quality parts aren't outrageously expensive in the smaller values required.

Bi-amplification from the subwoofer to the satellites, on the other hand, makes a great deal of sense. It's desirable to have a dedicated amplifier for the midrange and treble. Confining one amplifier to the region above the lower midrange will normally reduce IM distortion in it. A separate amplifier can handle the bass frequencies. But there's a benefit I consider much greater, particularly if you have a high performance amplifier with extremely low IM distortion under any conditions.

WORTHWHILE MODIFICATION. The Sub-1's first-order high-pass filter contains 200μF of series capacitance. Since the load is approximately 5Ω, the result is a −3dB point of around 160Hz. It would be nice if the 200μF capacitor did nothing more than roll off the frequencies below 160Hz. Unfortunately, it makes its presence known in another way. The capacitor inserts a substantial impedance between the amplifier and the loudspeaker. At 160Hz, the reactance of a 200μF capacitor is around 4.97Ω ($X_C = 1/2\pi FC$). This series impedance is nearly identical to the loudspeaker impedance. No matter how low your amplifier source impedance may be, you are still faced with an effective speaker-Z-to-

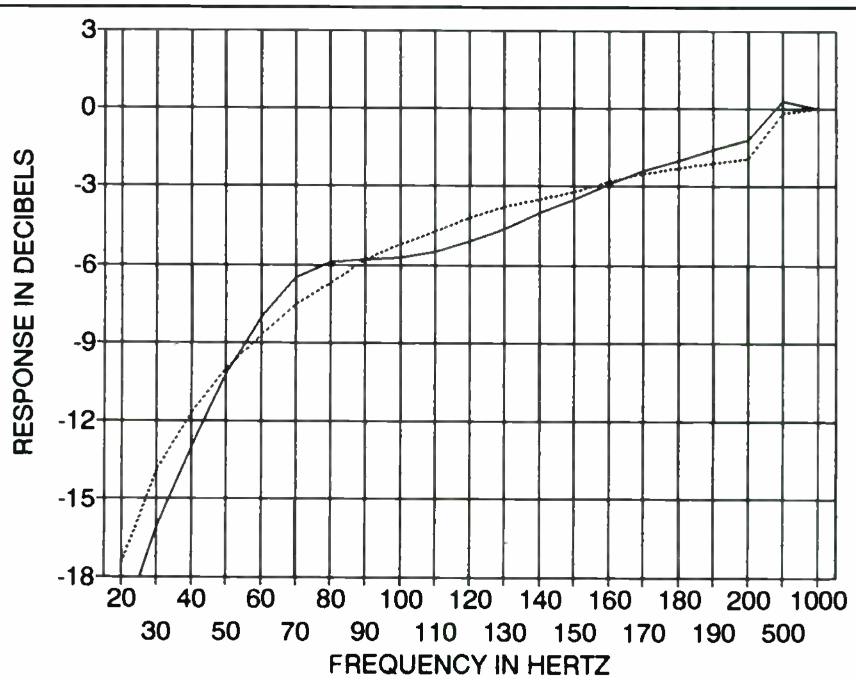


FIGURE 1: The solid line shows the Sub-1's high-pass crossover characteristic driving the Sapphire II. The dotted line is the filter characteristic of the modified Adcom GFA-585 power amplifier.

amp-Z ratio of around 1:1. The resulting damping factor is, of course, around 1 (see the sidebar on damping factor).

This isn't nearly as horrible a situation as you might think. Midbass drivers require far less control from the amplifier than woofers. This is partly because their excursions are extremely small by comparison. It's fortunate that this is true. If it weren't, the series impedances of the low-pass crossover components in the satellite would ruin the performance of the system. But it's still better to have as direct a connection from the satellite crossover back to the amplifier as possible. The improvement in amplifier control of the midbass driver should produce a worthwhile improvement in lower midrange clarity. The only way to achieve this improvement is to dispense with the passive high-pass filter. The crossover must be accomplished ahead of the amplifier. The only impedance between the amplifier and the satellites should be the speaker cable.

The 200μF cap is made from four 50μF electrolytics bypassed by a 2μF Chateauroux polypropylene and a 0.47μF Wonder-Cap. It would be better to make the high-pass filter entirely from high-quality film capacitors. I pointed this out in my review, and Audio Concepts' Mike Dzurko agreed that this would be a legitimate modification for those who might be able to hear the difference. This would add around $100 to the cost of the Sub-1 system, but it still wouldn't be as beneficial as dedicating a separate amplifier to the Sapphire IIs.

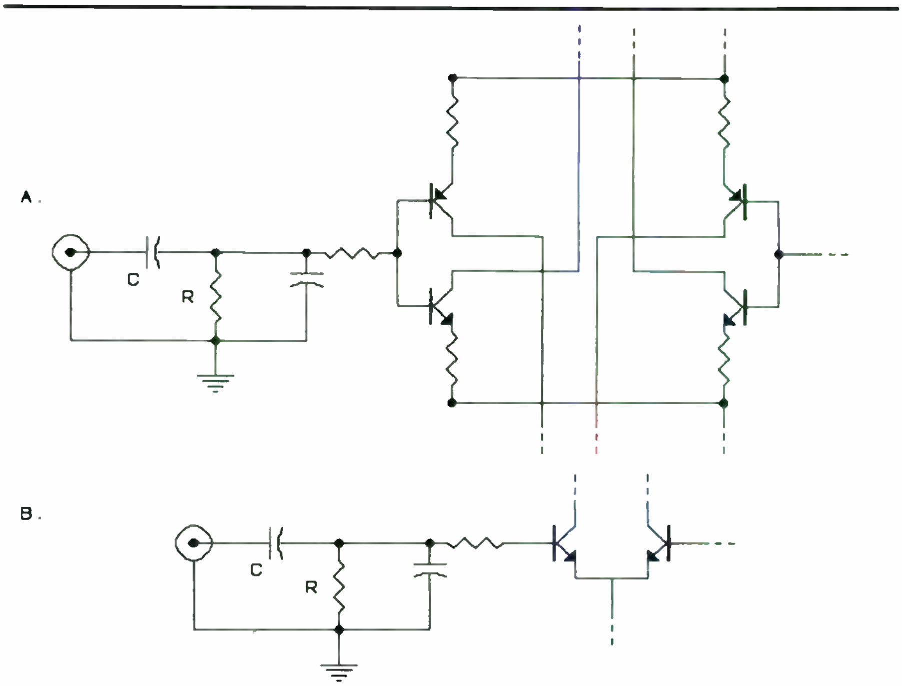


FIGURE 2: Typical power amp input stages. "A" is a full complementary differential input stage typical of higher priced amplifiers. "B" is the more common two-transistor differential input found in most cost-effective designs. The time constant of the RC input network determines the low-frequency rolloff.

DUPLICATE THE FILTER. Bi-amping normally requires an active (electronic) crossover between your preamp and power amplifiers. Even a simple active crossover will be somewhat complex if it is to enhance, rather than degrade, the performance of your loudspeaker system. A high-quality, low-impedance power supply is essential. High slew rate, low DC offset op amps must be complemented with high-quality metal-film resistors and polypropylene capacitors. You'll also need high quality internal wiring, connectors, and level-adjustment controls. Fourth-order Linkwitz-Riley crossovers require twice the number of active components needed for lower order filters. Over the years, I've built several active crossovers with excellent performance characteristics, but they weren't simple or inexpensive. Build a cheap active crossover, and you may wish you'd left your system running off a single amplifier. If you do it right, the rewards are considerable.

The Sapphire II/Sub-1 combination is an extremely attractive system to bi-amplify, since the complexities of an electronic crossover are not needed. The Sub-1 subwoofer is an unusual design. You'll probably degrade the performance of the Sub-1 if you defeat its passive low-pass filter. The impedances of the crossover components have been taken into consideration in deriving the Q of the system. This is part of what was necessary to extract such clean and extended low bass from a reasonably sized enclosure. So, we don't need to worry about a low-pass filter between the preamp and power amplifier.

That leaves us with the high-pass filter. Audio Concepts has made life extremely easy for us. The high-pass crossover they built into the Sub-1 is a simple first-order (6dB/octave) filter. In many cases, this wouldn't result in a first-order crossover characteristic due to the changing impedance of the satellite system. But Audio Concepts' aperiodic loading has resulted in an unusually smooth impedance curve for the Sapphire II mid-bass driver. When the Sapphire II is connected to the Sub-1, the result is a nearly ideal first-order crossover characteristic. The solid line in *Fig. 1* shows this. All we need to do is duplicate this filter ahead of the power amplifier.

DETERMINE IMPEDANCE. You could build a 6dB/octave active crossover for the satellites, but this is totally unnecessary. If your power amplifier is typical of most, it already contains a *passive* first-order high-pass filter. All we need to do is modify it to suit our purposes. *Figure 2* shows the input circuits of typical modern, solid-state power amplifiers. The input coupling capacitor C blocks DC offset from the preamp. The input termination resistor R determines the input impedance of the amplifier. The time constant which results from this R/C combination determines the low frequency roll-off point of the amplifier.

The manufacturer of your power amp will probably have set this R/C combination to produce a −3dB point somewhere between 1 and 5Hz. In modern high-performance solid-state amplifiers, this point will often be below 1Hz to minimize phase shift in the lowest audible frequencies. All we need to do is change the input coupling capacitor to a value which will produce a −3dB point of 160Hz. This is incredibly simple. Any first-order filter can be calculated using the formula:

$$C = 159{,}000 / RF$$

where C is the capacitance in microfarads, R is the input impedance in ohms, and F is the −3dB frequency in hertz.

The power amplifier I use for the Sapphire IIs is an Adcom GFA-585, which has an input impedance of 50k. The ex-

Damping Factor Reviewed

by Gary A. Galo
Contributing Editor

Damping factor, practically speaking, is the amplifier's ability to control the cone motion of the loudspeaker. More technically, it is the ratio between the speaker impedance and the amplifier source impedance.

THE BEST DESIGNS. When your power amplifier drives your loudspeaker, two things happen. First, the amplifier delivers alternating current to the loudspeaker. The current drawn from the amplifier is determined by the loudspeaker impedance (usually between 4Ω and 8Ω) and the amplifier's output voltage.

But there's something else happening which is often overlooked. As the loudspeaker's voice coil moves back and forth in the magnetic field, it also behaves like an electric generator, feeding alternating current back to the power amplifier.

The load impedance seen by the loudspeaker is the source impedance of the amplifier. The source impedance of the amplifier is usually very small, especially in modern solid-state amplifiers. It will normally be less than 0.1Ω in even the most modest power amplifiers, and can approach 0.01Ω in the best solid-state designs.

EXTRANEOUS MOTION. Although amplifier manufacturers usually give a damping-factor specification, they rarely tell you the source impedance. You can figure this out yourself quite easily:

$$Z_{SOURCE} = Z_{LOUDSPEAKER} / DF$$

where DF is the damping factor specified by the amplifier manufacturer and $Z_{LOUDSPEAKER}$ is the speaker impedance given in the DF specification. Amplifier manufacturers should give a load impedance figure with the damping factor, since the damping factor changes with loudspeaker impedance. If they don't, assume 8Ω. If the amplifier has a damping factor of 400 at 8Ω:

$$Z_{SOURCE} = 8 / 400 = 0.02\Omega$$

If the cone motion of the loudspeaker is well-controlled, the speaker cone movement will closely duplicate the signal fed by the amplifier. The loudspeaker will make very little extraneous motion on its own (I'd like to tell you that life is perfect, but it isn't, even with low impedance amplifiers). Extraneous motion can take the form of overshoot in either direction, ringing at the ends of transients, and so on. If the loudspeaker is connected to a high-source impedance, the cone will make a great deal of extraneous motion. If the source impedance is low, the loudspeaker motion will more closely mirror the signal from the amplifier, with minimal extraneous movement.

AN ANALOGY. Why is this so? If an electric generator is connected to a high-impedance load, it will move very easily. If you operated a hand-cranked generator, you could verify this yourself. In fact, once you got the generator started, you could let go of the crank, and it would continue to turn on its own for a short time. On the other hand, if a very low impedance load is connected to the generator, it becomes difficult to turn. As more current is pulled from the generator, the magnetic fields surrounding the generator's coils increase. The magnetic interactions in the generator make the generator harder to crank, and it will stop quickly when you release the handle.

Nearly all of us own at least one device which demonstrates this effect quite nicely—our automobiles. Every car contains an alternator which charges the battery. The term "alternator" is used because it actually generates three-phase alternating current which is converted to DC with a rectifier bridge, regulated, and then fed to the battery. Years ago, many cars contained generators which produced DC directly.

When you start your engine, you can hear the engine turning at its normal idling speed. If you turn on the lights, you can hear the engine slow down. The engine's crankshaft is turning the alternator via a belt. The lights are a very low-impedance load, and draw considerable current from the battery and alternator. This makes the alternator more difficult to turn, which is why the engine speed is reduced. The fact that the speed of a high-powered automobile engine can be reduced by pulling additional current from the alternator demonstrates the magnitude of this effect.

AWG CHART. When the loudspeaker is connected to a very low source impedance, extraneous motion becomes very difficult. If it's connected to a high source impedance, the cone can make all sorts of back and forth movements unrelated to the signal supplied by the amplifier. Unfortunately, loudspeaker cables raise the source impedance of the amplifier, as seen by the loudspeaker.

Figure 1 shows a power amplifier connected to a loudspeaker with 2-conductor speaker cable. If the loudspeaker could be directly connected to the power amp, the loudspeaker would see

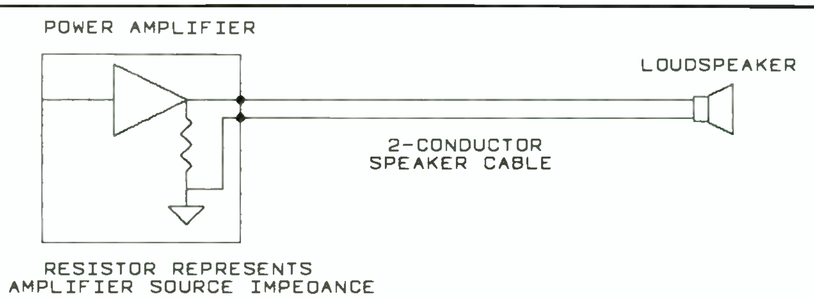


FIGURE 1: An amplifier is connected to a loudspeaker with 2-conductor cable. The lengths of *both* conductors must be considered when determining the wire resistance. The resistor in the power amplifier represents the amplifier's source impedance.

act capacitor value required is 0.0199μF, but 0.02μF is certainly close enough. Many amplifiers have 100k input impedances, in which case 0.01μF is fine. Tube amps are often higher. You'll have to determine the input impedance of your own power amplifier and plug that value into the formula.

CLOSE CURVES. Capacitor selection is extremely important. To obtain the finest possible sound from the bi-amplified system, it's important to use a premium quality capacitor. Since the value required is so small, cost and physical size aren't limiting factors. I chose MIT Multi-Caps designed by Richard Marsh. Since MIT doesn't manufacture an 0.02μF cap, the actual source impedance of the amplifier. In practical situations, however, we must use cable to connect the amp to the speaker. When we do this, the source impedance seen by the loudspeaker is actually the amplifier source Z *plus* the total resistance of the cable. If you have 15′ of speaker cable, you actually have 30′ of wire adding resistance to the source Z, since the cable has two conductors. The effective damping your amplifier can provide for the loudspeaker will be greatly affected by the resistance of your speaker cables.

You can compute the effective damping factor quite easily. *Table I* is a Copper Wire Resistance Chart which gives the resistance per foot of various wire gauges encountered in cable used for loudspeakers. Rather than list every American Wire Gauge (AWG), I've limited the chart to gauges from 8 to 20. I can't imagine any *SB* reader using cable as thin as 20AWG, but if you have cables made up of small strands in parallel, the chart can help you compute their equivalent resistance. Cables heavier than 8AWG are more likely to be found at your local welding supply shop rather than an audio dealer. Remember that the smaller the number, the heavier the wire. *The ARRL Handbook* (formerly *The Radio Amateur's Handbook*) contains a Copper Wire Table in every edition, and many other electronics reference books list this data. *The ARRL Handbook* lists resistance in ohms per thousand feet, along with a great deal of other wire data for all AWG sizes from 1 through 44.

TABLE 1

COPPER WIRE RESISTANCE CHART

WIRE GAUGE (AWG)	OHMS PER FOOT
8	0.0006405
9	0.0008077
10	0.001018
11	0.001284
12	0.001619
13	0.002042
14	0.002575
15	0.003247
16	0.004094
17	0.005163
18	0.006510
19	0.008210
20	0.01035

(Courtesy of The ARRL Handbook, *66th Edition, 1989; Copyright 1988 by the American Radio Relay League, Newington, CT 06111. Updated annually.)*

COMPUTE THE DF. Start by finding the amplifier source impedance, as shown in the formula above. We'll use an imaginary power amp with a damping factor of 400 at 8Ω, so the source Z is 0.02Ω. Let's say that we're using 15′ of 2-conductor, 16AWG wire. To find the total resistance of the wire, look up the resistance per foot in *Table I*, and multiply this figure by 30. Remember that there are two conductors adding resistance, so you must multiply the cable length by two.

$$30' \times 0.004094\Omega/\text{ft.} = 0.12282\Omega$$

Now add this resistance to the amplifier source impedance, to find the effective source impedance seen by the loudspeaker.

$$0.02\Omega + 0.12282\Omega = 0.14282\Omega$$

Finally, compute the damping factor using a transposition of the formula given above.

$$DF = Z_{LOUDSPEAKER} / Z_{SOURCE}$$

or

$$DF = 8 / 0.14282 = 56.015$$

As you can see, a high performance power amplifier has been reduced to rubble by flimsy loudspeaker cable. A damping factor of 400, when the cable is added, becomes a mere 56.

SHORT CABLES. Now let's see what happens when we switch to 15′ of 12AWG cable. Remember that the total wire length is 15′ times two.

Step 1.) $30' \times 0.001619\Omega/\text{ft.} = 0.04857\Omega$

Step 2.) $0.02\Omega + 0.04857\Omega = 0.06857\Omega$

Step 3.) $DF = 8 / 0.06857 = 116.67$

That's more like it. Now we have an effective damping factor of 117. Ideally, I'd like to see an effective damping factor of at least 100, but this is hardly an unbreakable rule. Many things beyond our control affect damping factor, including the series impedances of the wiring and crossover components inside the loudspeaker. However, many dynamic loudspeakers are designed to perform best if the crossover network is fed from a low source impedance. Speaker fuses can have resistance close to 0.01Ω if they have low current ratings. This negates any effort you've made to optimize loudspeaker performance with heavy cables. I don't recommend using speaker fuses, and most high-end power amplifiers don't have them.

If we want our speakers to perform to their potential, we must ensure that the resistance between the loudspeaker terminals and the amplifier is kept to a practical minimum. I believe this requires the use of wire with an AWG equivalent of 12 to 13 unless the cables are extremely short.

CABLE MERITS. Other factors also affect the sound of cables, and I do believe that cables sound different from each other for reasons beyond just wire gauge. The purity of the copper, type of insulation (dielectric) used, and the physical configuration can also make a difference.

The AudioQuest Indigo I use has ten individually insulated, solid, oxygen-free copper conductors, five for positive and five for negative. The individual conductors are 20AWG. When five are connected in parallel, the resistance is equivalent to a 13AWG cable. The ten conductors are wound spirally around a nylon core. At around $2 per foot, this cable offers genuine sonic value.

Debate will continue on the merits of "esoteric" cable configurations, but one thing is fairly certain—skinny wire is not acceptable for most applications. The negative effect of cable resistance on damping factor also makes a strong case for shorter loudspeaker cables and longer interconnects, if your preamp can drive longer cables without problems. Some preamps don't perform well when driving the extra capacitance of longer interconnects. Contact the manufacturer of your preamp if you have any doubts.

I hope this tutorial will help make choosing your gauges and loudspeaker cable lengths easier.

I put two 0.01μF in parallel for each stereo channel. A total of four 0.01μF capacitors are required for the two stereo channels. These capacitors cost $4.95 each, but I believe they're worth every penny. If you had to build or purchase an active crossover for bi-amplification, you could spend several hundred dollars. By comparison, a total expense of less than $20 seems like a bargain.

Replacing your amp's input coupling capacitor should be a very simple task. If you have any doubts about the location of the capacitor, obtain a service manual from the manufacturer. A few amplifiers are DC-coupled—they don't have an input coupling capacitor. In these rare cases, you can insert the capacitors between the input jacks and the PC boards. Again, consult the service manual if you have any doubts.

Figure 1 shows how close my modified GFA-585 power amp's high-pass characteristic compares to Audio Concepts' original passive filter. The solid line is the electrical response at the Sapphire II terminals, when fed from the Sub-1's high-pass filter. The dotted line is the modified power amp's low frequency roll-off. I made these measurements with my Loftech digital meter, which has resolution to 0.1dB. As you can see, the two curves are extremely close.

SHIELDED CABLES. Since we're not going to need an active crossover, we'll have to feed both power amplifiers from the preamplifier's outputs. Each channel of the preamp will be driving two power amplifier inputs with two cables. Most high-quality preamps, tube or solid state, should be able to do this without difficulty. You'll need to purchase, or make up, high-quality interconnect cables to go between your preamp and the two power amplifiers.

I've made cables with the Mogami 7551 RCA plugs and Joseph Grado Signature Laboratory Standard Audio Cable. Grado's cable is $5/foot. You can also use Mogami Neglex 2534 cable with the 7551 connectors, but you'll hear the superiority of the Grado cable in a high-resolution system. You can pay hundreds, or even thousands of dollars for interconnects, but that's not very practical (at least not for me). Grado's wire may seem a little expensive compared to the Mogami, but I believe it offers genuine value if you can make the investment. Cables are often system-sensitive. What produces good results in my system may not satisfy you.

Before you make your cables, you'll have to decide how you're going to feed the two power amps from your preamp, since most preamps only have one set of output jacks. An easy way to implement this is to solder the two cables feeding the power amp to a single RCA connector at the preamp end, making a giant, high performance "Y" connector. The Grado and Mogami cables are both four conductor shielded designs. When using either of these cables, it's best to float the shield on one end. With Mogami 2534, use the white pair for signal and blue pair for ground.

BETTER SOLUTION. On the preamp end, only the blue wires are connected to ground—the shield floats. Cut the outer insulation and shield back about 2½" on each cable and put a piece of heat-shrink tubing (available from Radio Shack) over the outer insulation to cover loose strands from the shield. The heat shrink should extend over the center conductors about ½". You should now be able to fit all eight center conductors from the two cables through the RCA connector's metal sleeve. Solder all four blue wires to ground and all four white wires to the center pin. On the other ends of the cables, the blue wires and the shield are both soldered to the ground of the RCA plug.

If you're using Grado cable, red and green are signal, blue and white are ground. The shield is floated on one end just like with the 2534. The Grado center conductors are a little tricky. Each one consists of fine copper strands wound around a thin nylon core. After you strip off the insulation, carefully separate the fine copper strands from the nylon core and then cut off the core. Now you can twist the strands together and solder them.

Opinions differ on the orientation of cables with floating shields. Mogami recommends connecting the shield where the cable terminates, which means at the

Continued on page 30

SCR
Metallized Polypropylene
Precision Capacitors
400VAC/630VDC
Now in stock at
Advanced Audio Products
(206) 527 3393
Big Cove Research
(205) 539 1840
Speakers Etc . . .
(602) 944 1878
Zalytron Industries
(516) 747 3515

You may wonder how good a capacitor can be, and you are right about that. Most people don't have access to the sophisticated instruments needed to test a capacitor. Chances are that you will have to make some sense of the numbers, jargon and figures that are advertised, very often with not much details on how it relates to sound quality. Quite frankly we don't know what to make of such a claim as: "the lowest phase deviations." Compared to what? At what frequencies? Believe it or not, some of these people are talking about Megahertz (millions Hz) or even Gigahertz (trillions Hz)! How about "the lowest ESR" (equivalent series resistance)? Compared to what? If at all true, we are talking milliohm here!

Much more concerned about the audible range, we, at ORCA, worked with SCR engineers to come up with a capacitor incorporating mechanical integrity and high internal damping. Why? Well the hard fact is that a capacitor is in many ways a transducer by construction, just like a woofer or a microphone. If you submit it to vibration, it will generate a current which will modulate the original signal. And capacitors are definitely submitted to vibrations in or near a speaker! That's why we make the SCR cap very solid:

– First we use a much thicker (higher voltage) film than most other manufacturers: 400VAC/630VDC. This allows us to wind our caps with a much higher tension force on our Swiss-made precision machines (a thinner film would break). Hence a much tighter, ultra compact cap that won't vibrate.

– Second we fill up the inner core and both ends with polyurethane resin (high structural strength and damping).

– Third we use heat-shrunk PVC tubing (for an even tighter construction), not tape-wrap.

Now let's talk about precision. You have heard about "self-healing" caps. What it truly means is that when a hole pokes through the film (it does happen) the whole capacitor does not go bad, it simple looses one turn (one layer). When your cap looses a turn of its winding, its value decreases. So the true capacitance of your cap decreases with time. A 5% cap can become a 10% cap. No need to be a rocket scientist to realize the thicker the film, the less chance to have holes poking through it and the more accurate the true cap value over a longer period of time. So now you know why we use the 400VAC/630VDC film. This is of real consequence in high order X-over designs.

Most of our customers don't know about all that (and they don't really need to), they prefer the SCR caps because to them they sound better. That should be the only reason for you to select them. As they say, if you don't hear the difference, don't pay for it. And we make your decision easy: we practice fair pricing. As an indication our 1 microF 400VAC SCR cap retails for less than $2.00.

OEMs & EXPORT, please contact directly ORCA, Design & Manufacturing Corp., 1531 Lookout Drive, Agoura, CA 91301–USA–TEL (818) 707 1629 FAX (818) 991 3072

It's no secret that we like to work with Joe. It is always enlightening and innovative. And a lot of fun too!

Besides his famous MTM geometry, Joe has designed some wonderful bandpass subwoofers for us. You know Joe, there was no available software that he could find which was good enough for him. Next thing we heard was he was busy designing his own program for "optimum" bandpass designs. And while at it, he added all the conventional designs and even equalizers too.

When we built the first speakers based on this new software, we were stunned by how close they came to predicted performance. Even in its early stage, this software was so unique and powerful that we thought it would be a wonderfull tool to share with speaker builders. We agree with Joe that the more people know about speaker design, the greater chance we all have to grow together as a large community of amateur speaker builders.

Joe went back to his permanent state of effervescence, and, assisted by his good friend Ron Warren, improved, polished and expanded the new born TOP BOX program. Now it is a start-to-finish design tool. When it came to price, we all agreed: it will be a full featured, densely packed high value package for a two digit price. After all the idea was to share it with as many of you as possible. At $99.00, it's a steal. And that's the way we want it.

TOP BOX is now available through:
BIG COVE RESEARCH TEL (205) 539 1840 FAX (205) 430 3961
JUST SPEAKERS TEL (415) 641 9228 FAX (415) 648 5306
ZALYTRON INDUSTRIES TEL (516) 747 3515 FAX (205) 294 1943

TOP BOX
SPEAKERWARE

Speaker software by Joe D'Appolito and Ron Warren

- Optimum open/close box systems.
- Automatic Optimum bandpass.
- Accurate port designs.
- Modified Thiele/Small for on target designs & simulations.
- Air Mass Loading Correction (AMLC) for accurate in-box predictions.

FREE 8-PAGE DESCRIPTION UPON REQUEST FROM ORCA.

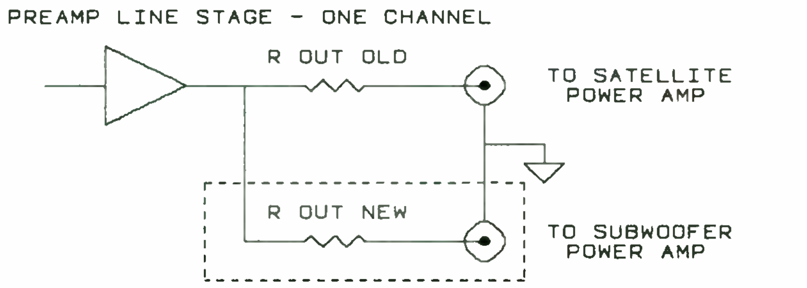


FIGURE 3: Modified preamp output stage (only one channel is shown). The parts inside the dashed line are added to feed the subwoofer amplifier. The new output resistor should be the same value used by the preamp manufacturer for R_{OUT} old.

Continued from page 28

power amp. Joe Grado says just the opposite—connect the shield at the preamp. To be honest, I've tried the cables both ways and I'm hard pressed to tell the difference. If you connect a pair of cables to a single RCA plug, as I've just described, you have little choice. You can't possibly fit the shields from two cables through a single connector sleeve, so they must be connected on the other end.

If you're willing to make a simple preamp modification, there's an even better solution. Simply add a second pair of RCA output jacks to your preamplifier. You could drill a new pair of holes in the rear panel for the new pair of jacks, or you could rewire an existing pair, if you have any that are unused. Most solid state preamps have a resistor in series with the output. This resistor determines the output impedance of the preamp, and protects the line level circuitry from the affects of capacitive loading. You could simply connect the second pair of RCA jacks in parallel with the first. I prefer feeding the new jacks from an additional pair of series resistors, which provides separate isolation for each output.

CABLE CAPACITANCE. *Figure 3* shows the circuit diagram. The new output resistors should be the same values as the originals. Be sure that the new jacks are insulated from the chassis, and that the shields connect to the same points as the original output jacks. I recommend Mogami 7557-G gold-plated RCA jacks, which are supplied with insulated mounting hardware. If you add a second pair of jacks to your preamp, you avoid the need for the special "Y" interconnects described above. If you make normal interconnects with Grado or Mogami cable, you should still float the shields on one end. But, you now have the flexibility of experimenting with cable orientation. In my own system, I added the second pair of RCA jacks to the preamp, and made four interconnects with Grado cable.

Once you've decided on interconnects, you'll need to select speaker cables. Again, you can spend thousands of dollars, but there has to be a practical limit. I'm using AudioQuest Indigo which you can buy from Audio Concepts for $1.99/ foot at this time. I prefer shorter speaker cables and longer interconnects, rather than the other way around. I have my power amplifiers located along the same wall with the loudspeakers. The interconnects going to my woofer amplifier are around 8′ long, while those to the satellite amp are 12′.

Your room, installation, and preamp will determine how you optimize the trade-off between the lengths of your interconnects and speaker cables. Your preamp must have a low enough output impedance to drive a higher capacitive load if you choose long interconnects. If you have any doubts, contact the manufacturer of your preamp and tell them the total lengths of the cables the preamp will be driving. They'll need to know what cables you are using and the total capacitance. The manufacturer should be able to tell you whether the preamp can drive the load without problems. The Grado cable has a capacitance of around 85pF/foot; Mogami 2534s is around 60pF/foot.

IMPERATIVE INSTALLATION. *Figure 4* shows the system connection diagram, which assumes that you've added a second pair of output jacks to your pre-

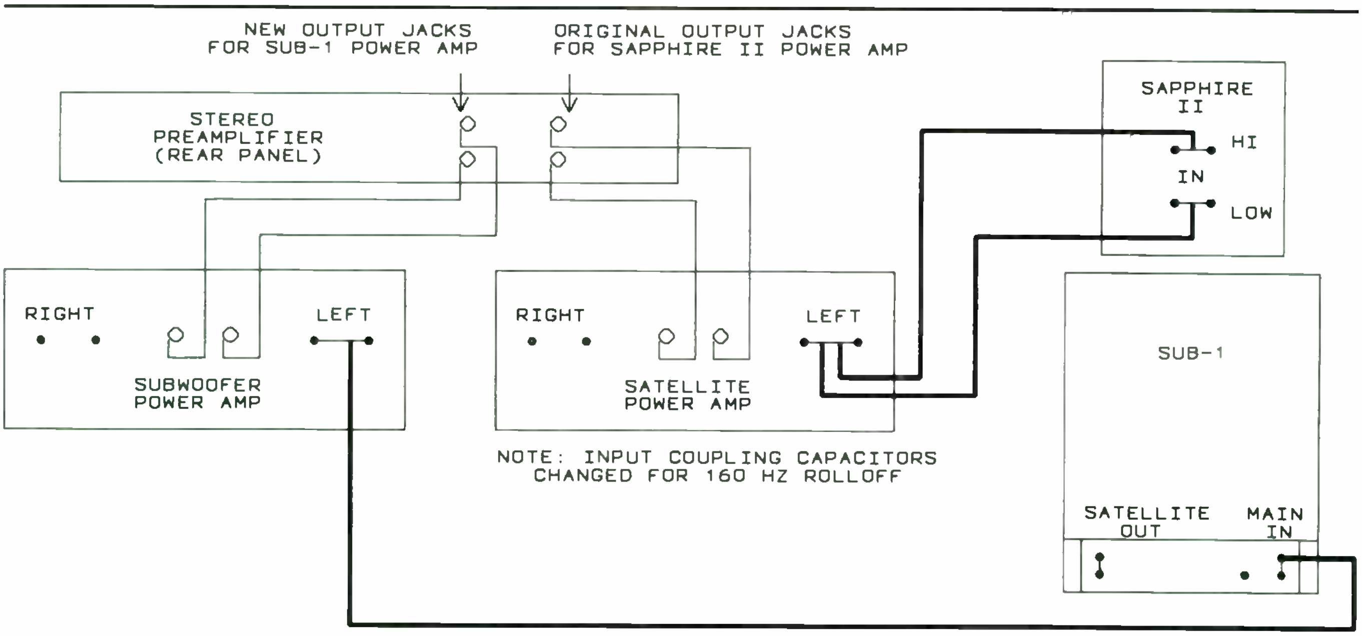


FIGURE 4: System connection diagram. Only the left channel loudspeaker connections are shown; the right channel is identical. The Sapphire IIs are bi-wired, and the Sub-1's satellite outputs are not used.

amp. To get the best performance from the Sapphire IIs, they should be bi-wired to the satellite amplifier as shown. If you're using identical power amplifiers, you don't need to worry about level matching between the satellites and subwoofers. I'm using Adcom GFA-585 power amps on both, so level matching isn't a problem.

It's worth noting that two power amplifiers can have identical voltage gains without having the same maximum power output. You may have a 250W/channel amplifier on the subwoofers and a 100W/channel amplifier on the satellites. As long as these amplifiers have the same voltage gain, you don't need to worry about level matching. Their outputs will be identical for a given input signal until the 100W/channel amp reaches its limit and clips. You don't really need as much power on the satellites as you do on the subwoofers. I chose to power the Sapphire IIs with a GFA-585 because I found the amplifier to be among the most sonically refined in its price class, not because I needed 400W/channel into the 4Ω load.

If your amplifiers aren't matched, you'll need to install a pot or a fixed attenuator at the input of the more sensitive amplifier. Don't install the attenuator at the preamp's output. You'll raise the output impedance of the preamp and substantially degrade its ability to drive long cables. This is a major weakness of completely passive preamplifiers. They work fine as long as you don't connect any cables to them. You could mount level controls in a small box close to the power amp. If the cable lengths between the controls and the power amp's inputs are kept to less than 1', you won't have any problems. It's imperative that you install the pot ahead of the input coupling capacitor. If you don't, you may affect the DC offset characteristics of the amplifier, and you'll change the input impedance.

EFFECTIVE PERFORMANCE. Bi-amplifying the Audio Concepts Sapphire II/Sub-1 combination raises this already excellent system to a new level of performance. The most striking improvement is in the detail and transparency in the midrange. The upper midrange is smoother and the entire midrange has greater clarity and definition. Eliminating the passive electrolytic crossover capacitor accounts for both of these differences. The soundstage is now larger, both left to right and front to back. And with my pair of high current Adcom

SOURCES

Audio Concepts, Inc.
901 South 4th St.
La Crosse, WI 54601
(800) 346-9183
Sapphire II, Sub-1, and AudioQuest Indigo Cable

Joseph Grado Signature Products
921 Tice Place
Westfield, NJ 07090
(908) 233-5893
Grado Signature Laboratory Standard Audio Cable

Michael Percy
PO Box 526
Inverness, CA 94937
(415) 669-7181

Old Colony Sound Lab
PO Box 243
Peterborough, NH 03458-0243
(603) 924-6371, FAX (603) 924-9467
Mogami connectors and 2534 cable—small quantities available on the 2534 cable.

Marshall Electronics, Inc.
PO Box 2027
Culver City, CA 90230
(212) 390-6608
Mogami connectors and 2534 cable—minimum order on 2534 cable is 164').

GFA-585 power amplifiers, the system has dynamics to burn.

In my review of the Sub-1s, I recommended using the passive crossover to feed just the mid-bass driver in the Sapphire IIs. The bi-wiring capability of the Sapphire II allows the tweeter to be connected directly to the power amp. The Sub-1's high-pass filter is unnecessary for the tweeter, and the electrolytic capacitors degrade the sound by a small degree. If you've been feeding your tweeter from the passive crossover, bi-amping will provide improvements in the treble region similar to those I've described in the upper midrange.

Finally, Audio Concepts has asked my opinion on a new woofer they're now using in the Sub-1s. The new driver is called the DV12, and it replaces the older, modified JC12 woofer. The new woofers have relatively thick, long-fiber pulp cones, rear-vented pole pieces, and rubber surrounds. These drivers provide a worthwhile improvement in performance over the original Sub-1 woofers. They are incredibly clean. The most noticeable improvement is in lower midrange/upper bass detail. The electrostatic-like transparency of the Sapphire IIs is further improved. Bear in mind that the Sub-1s reproduce information well above the crossover frequency, so they have a substantial effect on the performance of the Sapphire IIs.

At first I thought the new woofers had less weight in the low frequencies. When I played familiar recordings with extended low end, however, I realized that none of the Sub-1's bass extension had been sacrificed. I ran a warble tone analysis, and confirmed that the Sub-1s were still 1dB down at 22Hz in my room, exactly as they were with the old JC12 woofers. The improved transparency in the upper bass/lower midrange region makes the system sound a bit leaner in this region, but I quickly adjusted to the change, realizing that the old woofers were actually adding colorations now removed. The DV12 woofers are an excellent improvement for an already fine subwoofer.

(UNCERTAINTIES / Capacitors) = (GUESSWORK / Design)

The formula for the best design begins with the best components. And for critical applications in all audio circuits, you can't find better components than MIT's MultiCap capacitors.

- ***More Usable Bandwidth*** Each **MultiCap** gives you up to 10 precision, paralleled capacitors in one compact unit that extends the usable bandwidth.
- ***Highest Resolution*** Each **MultiCap**, with hand-soldered, large-gauge leads necessary for high current conditions, gives the highest sonic resolution possible. Closest to no capacitor at all!
- ***High Reliability*** The combination of care and innovation in the **MultiCap** gives you extraordinarily high quality and reliability. A cost-effective increase in the performance of your audio designs.

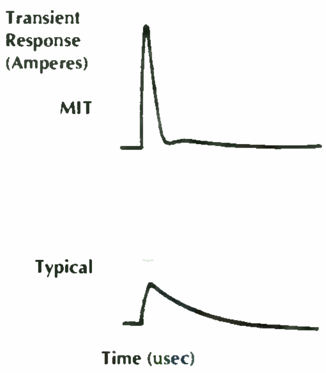
MIT **MultiCap** offers three series of capacitors: polystyrene film & foil; polypropylene film and foil; and metallized polypropylene film.

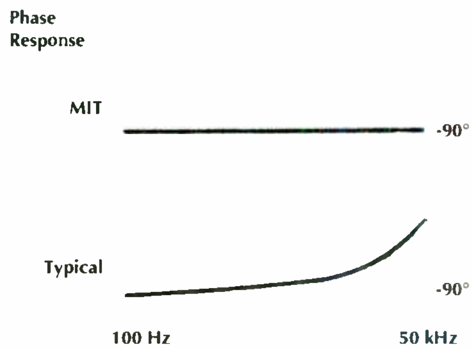
The Film & Foil MultiCaps

- Especially designed for all high-performance filter networks, coupling, supply by-pass, and servo circuits.
- Consistent and reliable for high performance under demanding high-current applications.
- Exceptional phase stability and superior pulse handling.
- Polystyrene, for extraordinary sonic performance, where high ambient temperature is not an issue.
- Polypropylene, for higher temperatures and lower costs.

The Metallized MultiCaps

- Same patented parallel design for ultra-low ESR and inductance losses.
- Smaller in size, less costly to manufacture. Lower prices for the consumer.
- Recommended for application wherever metallized capacitors are used.





Custom values, special low prices, and full technical support available to OEMs and modifiers from Music Interface Technologies. Ask for our current literature on capacitors and the CVTL laboratory. In the U.S., call CVTL/MIT 916.823.1186 or fax 916.823.0810. Outside the U.S., call MIT Components/Sonic Frontiers 416.847.3245.

MIT MultiCap ***"The most advanced capacitor design in the world."***

Fast Reply #GG703

CAPS FOR PASSIVE CROSSOVERS

BY RICHARD HONEYCUTT

"Hey, here's the man who can tell you." These words from the owner greeted me as I entered his electronics shop, and I knew I was in trouble again. The customer query, prompting this outbreak of confidence in me, was about loudspeaker crossovers. "What kind of capacitor do I need to use in this speaker system I'm building? I've got some old capacitors I took out of TVs and things, and I was wondering if any of them would do."

Karma, that's what it was. Having too often teased my wife about her tendency to describe the building of a watch when only the time was wanted, I was the one now actually being asked to describe how the watch should be built. Unwittingly, my questioner had hooked the professor in me and now, lack of blackboard notwithstanding, I'd like to convey to you what I explained to him that day.

SOME BACKGROUND. A capacitor is just two conductors (plates) separated by an insulator (dielectric), as Ben Franklin could have told you. (Of course, he'd have called the thing a Leyden jar, but no matter.) Its action can be described in any of several ways. The most fundamental description is that it stores charge. When one plate is connected to the positive terminal of a voltage source, and the other plate is connected to the negative terminal, charge flows briefly because of the electrostatic attraction of the opposite charges; like normal Girl and Boy Scouts with camps separated by a river, the opposite charges are attracted to each other. Thus we wind up with an excess of electrons on one plate and a deficiency of electrons on the other. Since charge cannot flow in the insulator, a point of equilibrium is soon reached, and no further charge flows. Now if the insulator is perfect, the capacitor's leads can be disconnected from the source and the charge will remain on the plates indefinitely. We define the capacitance of a given capacitor as the amount of charge it can store per volt of potential difference between the plates:

FIGURE 1: Circuit model for capacitor with ESR.

(1) $$C = Q/V$$

where Q is the electric charge in coulombs and V is the voltage.

If, however, after disconnecting the voltage source from the capacitor we reconnect it with the opposite polarity, the plate with excess electrons will dump them into the positive terminal of the voltage source. Electrons will then flow from the negative terminal onto the other plate, until equilibrium is established with the opposite polarity. Notice what has happened: there has been current in the connecting wires, even though there was at no time any current through the insulator. If we were to use a source of AC rather than DC, we could maintain a continuous current in the wires connecting the source to the capa-

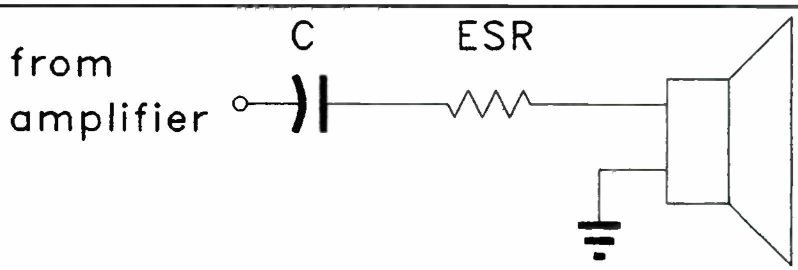


FIGURE 2: first-order H-F section with ESR.

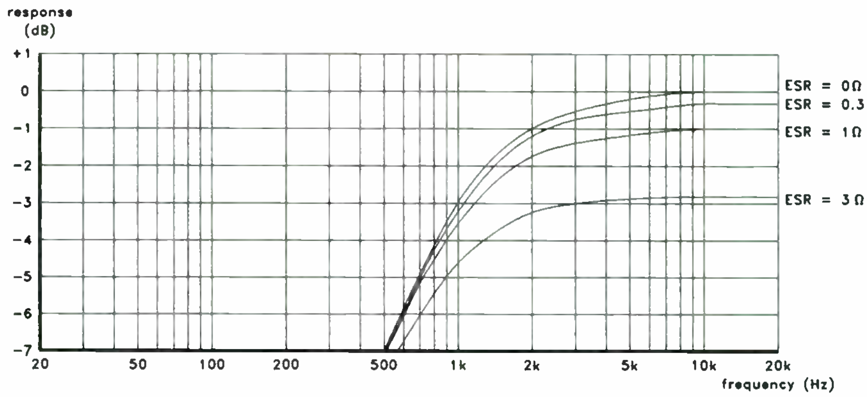


FIGURE 3: Effect of nonzero ESR upon first-order H-F section.

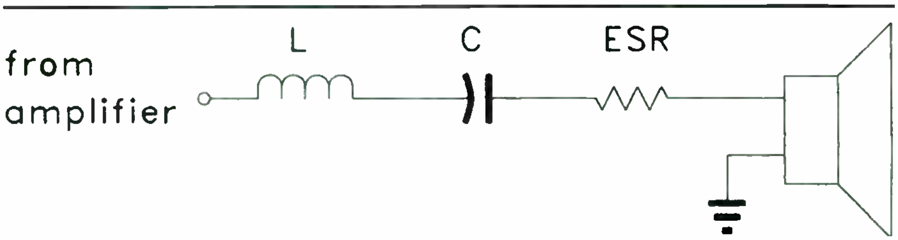


FIGURE 4: First-order M-F section with ESR.

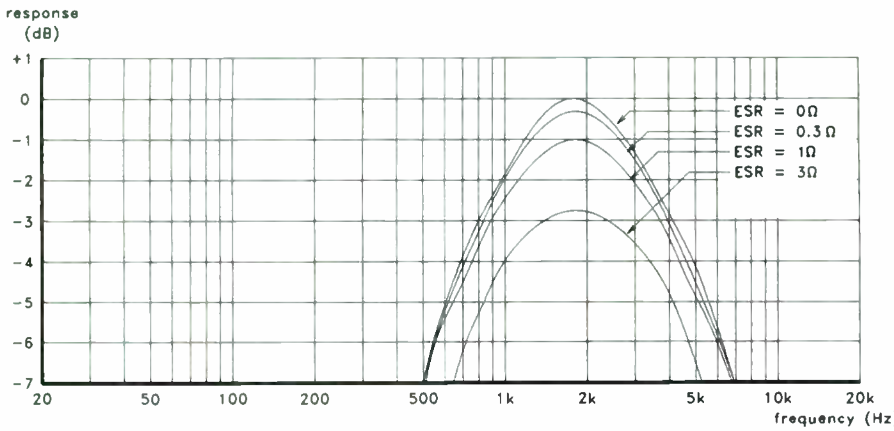


FIGURE 5: Effect of nonzero ESR upon first-order M-F section.

citor. Thus we often say that a capacitor blocks DC and passes AC. Any good physics professor would be quick to point out that it doesn't really pass AC—it just permits AC to flow in the remainder of the circuit.

THE DIELECTRIC. One of the valuable things discovered by Ben Franklin is that the storage of charge in a capacitor actually involves the innards of the dielectric. Molecules of some materials can be polarized, so that one end of each molecule is more negative than the other. When this polarization occurs in an insulating material, electrical energy is stored. A standard experiment in college physics begins by separating two large metal plates with a sulfur disc, and then connecting a battery to the plates. The battery is then disconnected and the sulfur disc is placed between two metal plates. A voltmeter connected between the two new plates will show a voltage. Thus the dielectric itself was charged.

With this knowledge, we should not be surprised to find that the capacitance of any given capacitor can be predicted from:

$$C = \epsilon A/t \quad (2)$$

where ϵ represents a characteristic of the dielectric, A is the area of the plates, and t is the thickness of the dielectric. Now ϵ, which is called the permittivity, can be broken down into two parts: ϵ_O and ϵ_R. These are the so-called permittivity of free space and relative permittivity, respectively. Relative permittivity is more often called dielectric constant, and varies from one material to another. So we can easily see from equation (2) three ways of controlling the capacitance of a capacitor: by choosing the plate area, by choosing the dielectric thickness, and by choosing the type of dielectric.

CAPACITOR TYPES. Capacitors made today fall into three general groups when classified by dielectric. The first is electrostatic capacitors; this family includes those whose dielectrics are made of mica, plastic film, ceramic, or oiled paper. The second is aluminum electrolytics. The third is tantalums. Both the second and third kinds of capacitors utilize a chemical process to form a metal oxide that acts as the dielectric. Because this metal oxide is very thin but has both a high breakdown voltage and a high dielectric constant, these types of capacitors provide large capacitance values in physically small packages. Thus engineers usually choose electrolytics or tantalums when they need capacitors over about $1\mu F$.

DIELECTRIC HYSTERESIS. Years ago, when I was a fledgling electronics buff, I too got many of my components by scavenging old TVs. Naturally, I wondered why there were some paper capacitors, some mica capacitors, and some ceramic capacitors. (No plastic film, back then!) Well, it turns out that each time our AC source reverses the polarization of the molecules in the dielectric, a little energy is required. Since energy can be neither created nor destroyed, any alert physicist, upon hearing the word energy, always wants to know where it came from and where it went. In this case, the energy came from the AC source, and it went into the surrounding air as heat. So we're losing a bit of energy from our circuit with each new cycle of AC. This process is called dielectric hysteresis loss.

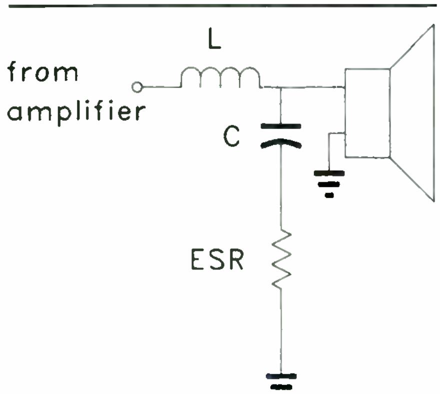


FIGURE 6: Second-order L-F section with ESR.

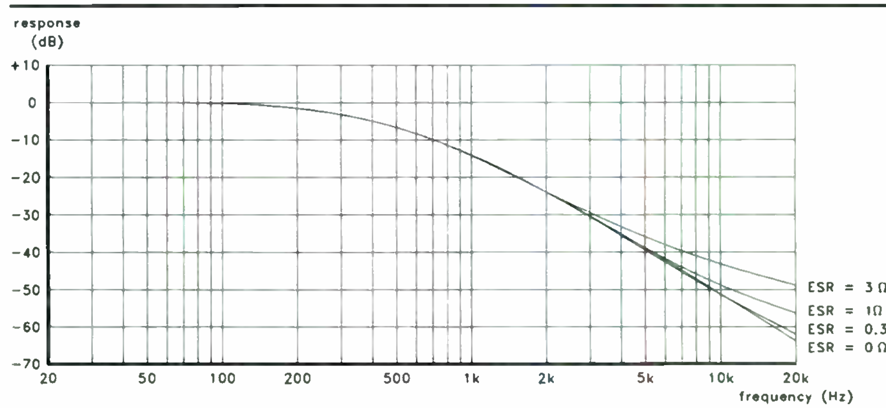


FIGURE 7: Effect of nonzero ESR upon second-order L-F section.

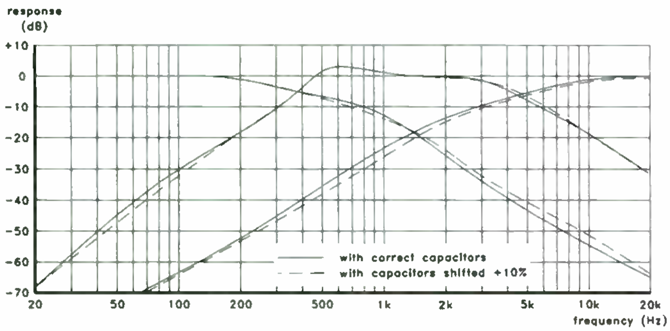


FIGURE 8: Effect of capacitance shift upon performance of three-way, second-order crossover network.

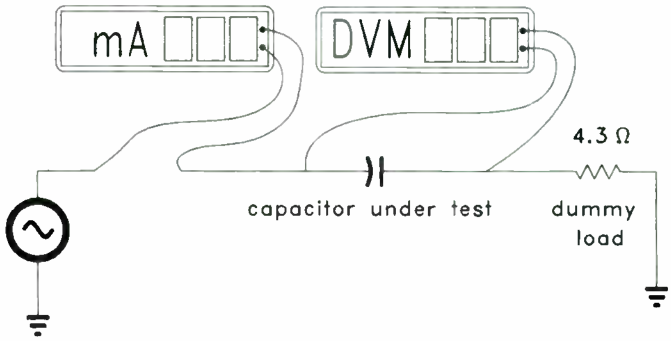


FIGURE 9: Setup for V/I test.

Just how much energy are we losing? It depends on the type of dielectric and how fast we're reversing polarity (that is, the frequency of the AC). The main reason for the use of different types of capacitors is that the dielectrics which provide the least dielectric hysteresis loss make for capacitors that are more expensive or larger or both. So engineers wisely choose the least expensive type of capacitor that will perform well at the highest frequency required.

ESR. Whenever we lose energy from a circuit, we will find a resistance that accounts for the energy loss. The energy lost because of dielectric hysteresis, plus some small losses due to lead and plate resistances in the capacitor, are lumped together into an Equivalent Series Resistance, or ESR. *Figure 1* shows an equivalent circuit for a capacitor that includes the effects of ESR. Now we can look at the effect of a capacitor with ESR being used in a loudspeaker crossover network. *Figure 2* shows the circuit of the high-frequency section of a first-order crossover using a capacitor with ESR. *Figure 3* shows the frequency response. The decrease in level is immediately obvious: it drops almost 3dB for a 3Ω resistor. (An 8Ω speaker is assumed.) Less obvious is the change in crossover frequency that results. With a 3Ω ESR, the −3dB frequency is about 700Hz, rather than 1kHz as it is with no ESR.

Figures 4 and *5* show the effect of ESR upon the performance of a first-order midrange section. As with the HF section, we find that ESR lowers the level and changes the crossover frequency.

A first-order low-frequency section does not contain any capacitors, so it would be unaffected by ESR. *Figures 6* and *7* show the effect of ESR upon a second-order LF section. Notice that there is, in fact, very little effect at all, except at frequencies above ten times the crossover frequency. There, the ultimate attenuation is less than it would be with a perfect capacitor. From these figures we can see the not-too-surprising result that ESR is mainly a problem in capacitors in series with the signal from amplifier to speakers.

Now, back to the question of which capacitor to use. Going from best to worst, we find that the ESR is lowest in mica and standard ceramic disc capacitors. It is almost as low in paper and film capacitors of all kinds, somewhat higher in tantalums, and highest in aluminum electrolytics. Unfortunately, it is impractical to build mica or ceramic disc capacitors large enough for use in crossovers. Thus we have to use film capacitors or electrolytics. Film capacitors are always better, but they may become impractically large and expensive for values over about 5μF. Tantalums are hard to find with a high enough voltage rating. Therefore, aluminum electrolytics are often chosen for crossovers.

But there's another consideration. The chemical process that forms the dielectric in an electrolytic capacitor requires a DC polarizing voltage. In order to successfully use an electrolytic in a crossover network (which has no DC present unless your amplifier is dying), we have to use two electrolytics connected back-to-back (+ to + or − to −) to form a nonpolar electrolytic. Commercial non-

Continued on page 38

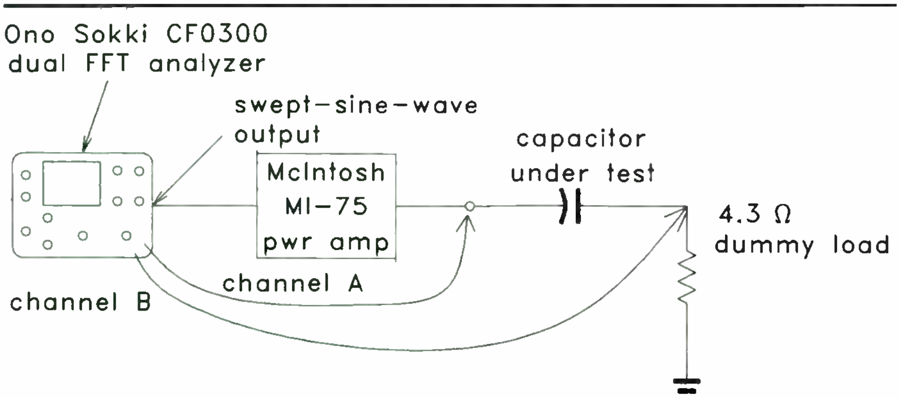


FIGURE 10: Test setup for transfer functions.

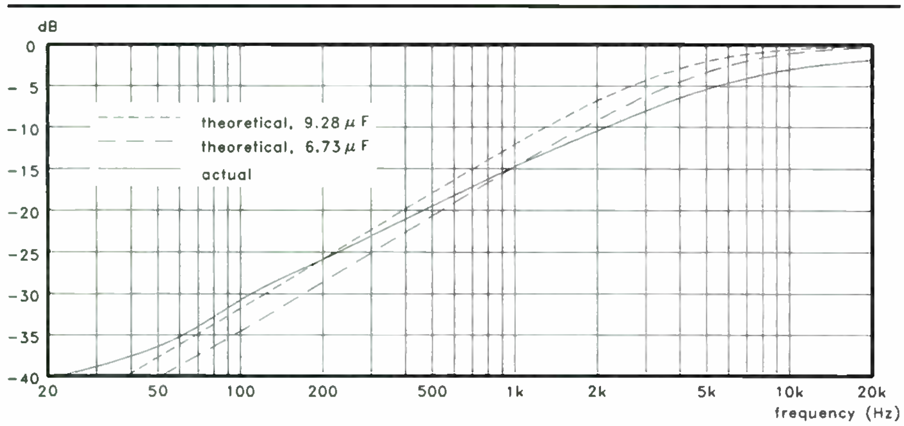
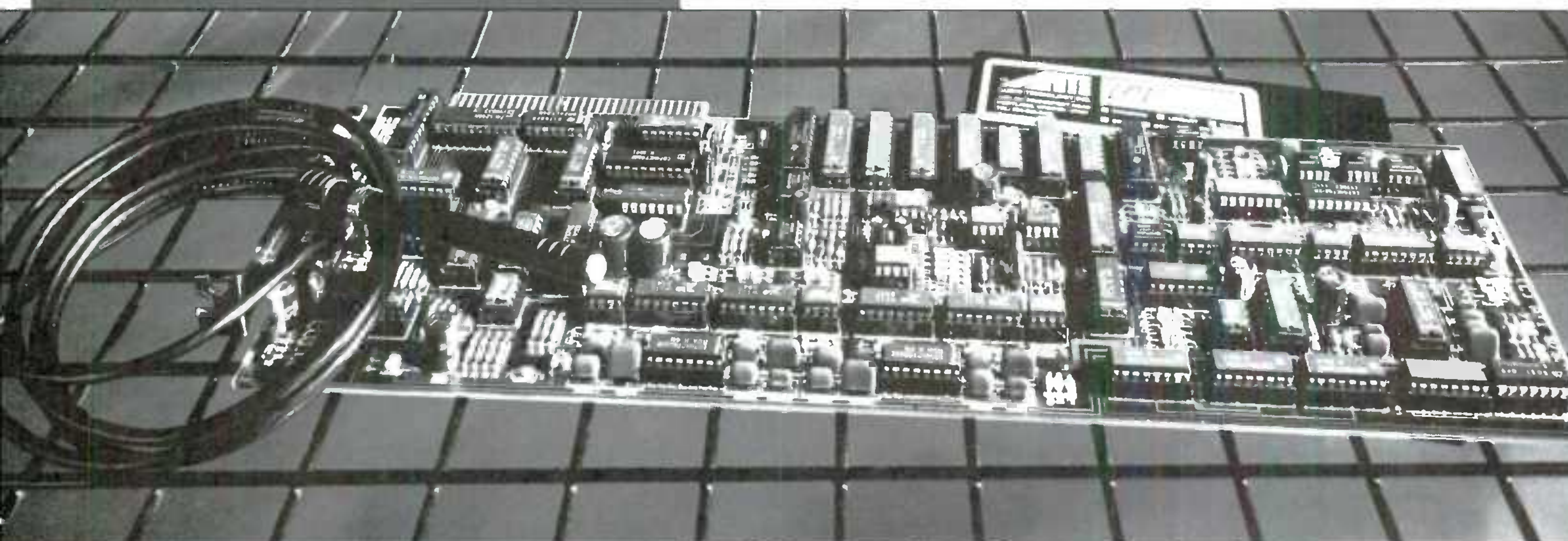


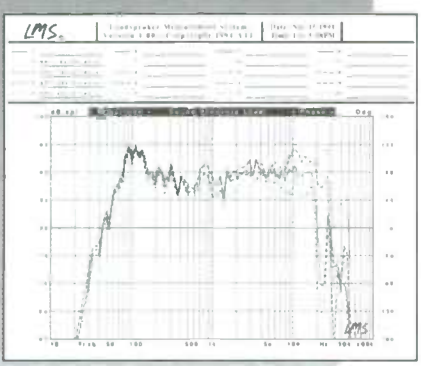
FIGURE 11: 6.6μF/50V, 17-yr.-old nonpolar electrolytic.

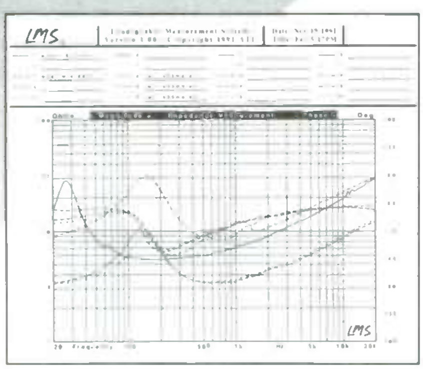
LMS

Loudspeaker Measurement ~~System~~ Solution!



Quick, Easy, Accurate





A laboratory-quality Loudspeaker Measurement System tool at an unbelievable price!

The ATI LMS is a complete electro-acoustic measurement system for your test/analysis requirements. You get a calibrated precision condenser measurement microphone, a PC-compatible card and LMS system control software. The LMS feautres a programmable sine wave oscillator, dual tracking filters with LP/HP/BP/BR modes, and a high-speed gating system for quasi-anechoic measurments. You can take SPL and Impedance measurements at the touch of a key, and print the graphs to a wide variety of output devices. Moreover, other system utilities are provided for scaling, smoothing, subtracting, dividing, and PASS/FAIL curve comparisons. For serious production testing, the LMS software provides macro programming capability for automatic test operation. And this is only the start. The LMS system provides features and capabilities usually found (if at all!) on systems costing many times more.

By utilizing the power of a PC, our own precision calibrated microphone (with greater than 20kHz bandwidth), and complete test/control software, we are able to offer a system with more features, precision, flexibility, and capabilities. Up to now, anything with this much performance had to be assembled out of high-priced stand-alone test instruments, expensive measurement microphones, and non-existent software!

Still not convinced? Give us a call and take our demo software package for a test drive.

Breakthrough Technology from ATI

Audio Teknology Incorporated
7556 SW Bridgeport Road
Portland, Oregon 97224
TEL: 503-624-0405
FAX: 503-624-0194

Fast Reply #GG1428

Continued from page 36
polars are available, but they're just two separate electrolytics internally connected back-to-back.

The capacitors in a TV generally fall into two categories: mica, ceramic, and film capacitors too small for crossovers, and polar electrolytics. So the final answer to the customer's inquiry about TV capacitors was: "Probably not. You'll need to buy film or nonpolar capacitors of the correct value and with a voltage rating of around 50V for a 100W amplifier. And you might want to know that the voltage rating doubles for each quadrupling of power over 100W."

MORE QUESTIONS. I headed for the door, as the other inhabitants of the shop breathed a sigh of relief at the end of the lecture. As I drove home, though, several nagging questions began to arise from the unquiet graves into which I had nudged them over the years. Just how bad is the ESR in a modern electrolytic? Is this a real problem, or just a perfectionist's neurosis? How about other capacitive effects, such as dielectric absorption, and the change in capacitance with frequency? Do capacitance and/or ESR change with applied voltage? Is there any truth in the claim I occasionally hear that crossover capacitors can cause distortion? How much crossover drift results from the aging of electrolytics?

TABLE 1

CHARACTERISTICS OF CAPACITORS

Type	C @ 120 Hz (μF)	C @ 1 kHz (μF)	DF @ 120 Hz	DF @ 1 kHz
I. Polyester film				
a. 0.56 μF/100V (12 yrs. old)	0.574	0.571	0.001	0.003
b. 1 μF/600V (9 yrs. old)	1.004	0.998	0.002	0.004
c. 4 μF/300V (10 yrs. old)	4.18	4.17	0.004	0.016
II. Paper-oil				
a. 4 μF/1000V (26 yrs. old)	4.09	4.07	0.012	0.077
b. 8 μF/1000V (11 yrs. old)	7.83	7.80	0.005	0.022
c. 4 μF/600V (25 yrs. old)	4.57	4.55	0.005	0.009
III. Nonpolar 'lytic				
a. 6.6 μF/50V (17 yrs. old)	9.28	6.73	0.202	0.280
b. 6.8 μF/100V (1 yr. old)	12.8	12.09	0.048	0.059
c. 2.2 μF/50V (20 yrs. old)	2.49	2.2	0.083	0.107
d. 26 μF/200V (15 yrs. old)	32.8	30.7	0.049	0.093
e. 6.8 μF/100V (1 yr. old)	7.32	6.84	0.057	0.063
IV. Back-to-back 'lytics				
a. two 47 μF/400V (9 yrs. old)	23.9	22.2	0.097	0.558
b. two 47 μF/100V (1 yr. old)	20.9	20.3	0.047	0.104
c. two 47 μF/50V (4 yrs. old)	30.7	29.1	0.047	0.188
V. Back-to-back 80 μF/400V computer-grade 'lytics (18 yrs. old)	47.4	46.3	0.033	0.128
VI. "Motor-run" film 25 μF/240VAC (6 yrs. old)	24.6	24.7	0.003	0.017
VII. Back-to-back ancient can 'lytics				
a. two 80 μF/450V Western Electric (37 yrs. old)	43.4	41.6	0.056	0.180
b. dual 30 μF/450V (27 yrs. old)	16.07	15.22	0.045	0.128
c. dual 20 μF/400V (34 yrs. old)	11.92	10.7	0.083	0.251
d. dual 10 μF/400V (31 yrs. old)	7.62	7.4	0.097	0.317
X. Paralleled film and 'lytic (I c and VII d)	16.96	16.25	0.031	0.035

TESTING. Back at the house, my library gave little help on any of these questions. Not to worry! When library fails, lab steps in! First, I gathered 20 capacitors of different kinds: three polyester, three paper-oil, five commercial nonpolar electrolytics (ranging from new to 10 years old), three homemade nonpolars (formed from pairs of fairly new polar electrolytics), a nonpolar made from back-to-back computer-grade electrolytics, a high-current film one made for induction motor use, and four nonpolars made from very old but unused electrolytics.

Then, armed with my trusty H-P 4261A LCR meter, I set to work. The 4261A measures capacitance and dissipation factor (DF) at two frequencies: 120Hz and 1kHz. (Dissipation factor is ESR/capacitive reactance.) *Table 1* lists the capacitors and the measurement results, which seemed pretty much as expected. The film and paper-oil capacitors had very low dissipation factors, implying very low ESR, and their capacitance change with frequency was less than ½%. Electrolytics were much worse on both counts: DF ranged from 0.033 to 0.558 (really!) and capacitance shift, from under 3% to over 27%. (Just for reference, I ran some curves to investigate the effect of a 10% capacitance shift upon a three-way second-order crossover network. The results are shown in *Fig. 8*.) Also, dissipation factor increased with frequency, and capacitance decreased. These effects could be explained by changes in the behavior of the dielectrics as frequency varied.

These results prompted me to analyze in more detail. Most textbooks I had read implied that ESR increases with frequency. By writing a simple BASIC program to analyze the measured capacitance and dissipation factor, I could solve for the ESR of each capacitor at each frequency. To my amazement, the ESR did not increase with frequency—it decreased! But since it decreased at a rate less than proportionality, and capacitive reactance decreases at a rate proportional to frequency, the dissipation factor did increase with frequency.

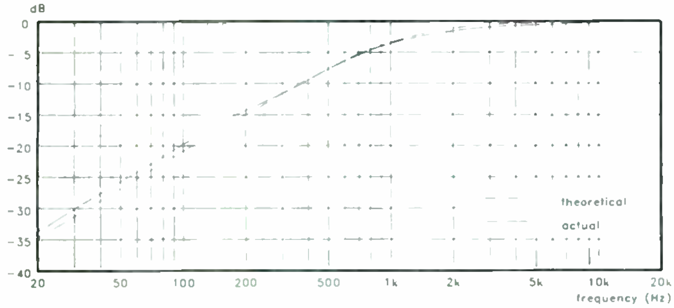


FIGURE 12: 26μF/200V, 15-yr.-old nonpolar electrolytic.

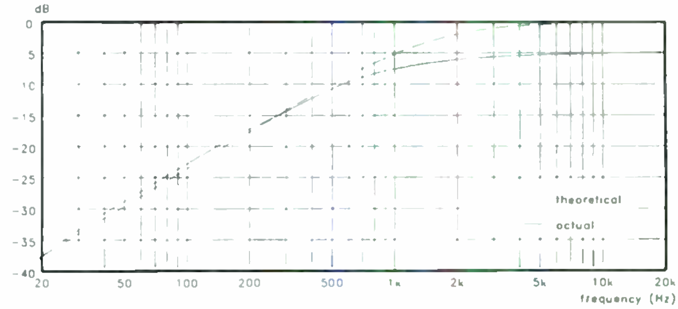


FIGURE 13: Two 47μF/400V, 9-yr.-old electrolytic back-to-back.

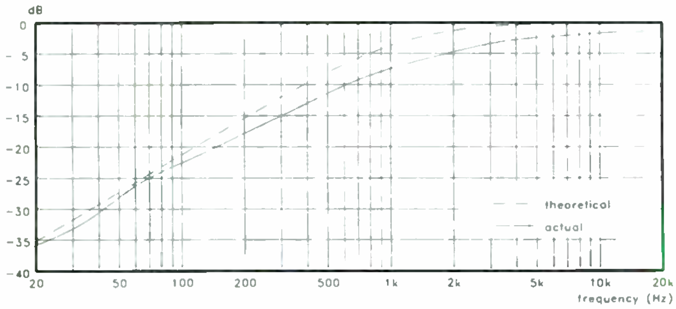


FIGURE 14: Two 47μF/100V, 1-yr.-old electrolytic back-to-back.

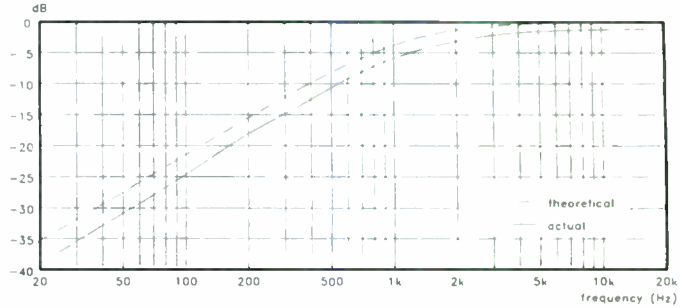


FIGURE 15: Two 47μF/50V, 4-yr.-old electrolytic back-to-back.

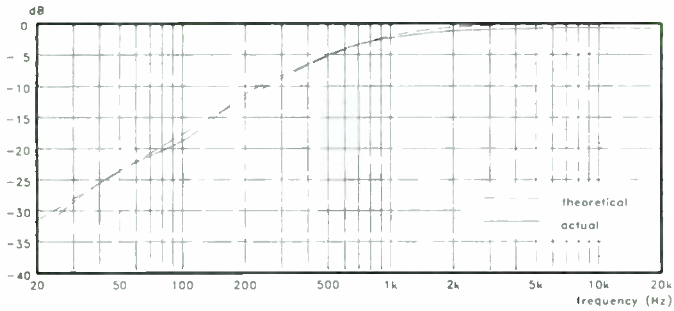


FIGURE 16: Two 80μF/400V, 18-yr.-old computer-grade electrolytic back-to-back.

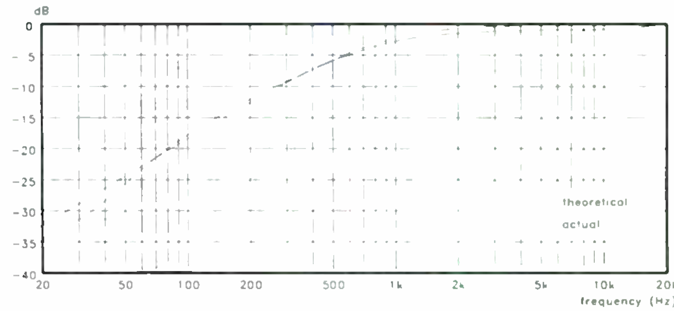


FIGURE 17: Two 80μF/450V, 37-yr.-old electrolytic back-to-back.

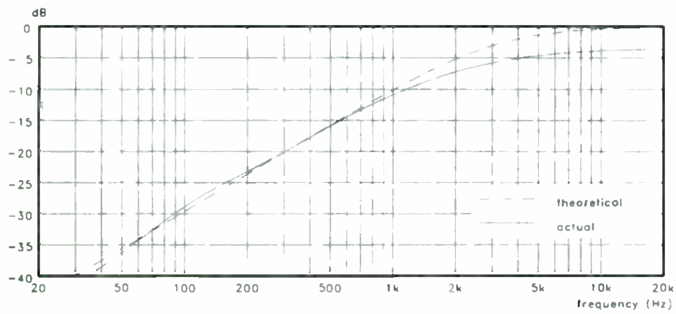


FIGURE 18: Dual 30μF/450V, 34-yr.-old electrolytic back-to-back.

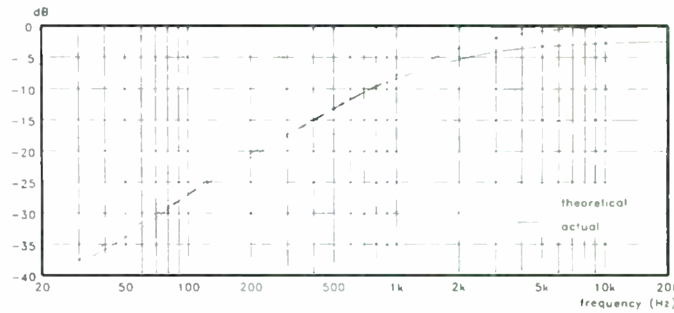


FIGURE 19: Dual 20μF/400V, 31-yr.-old electrolytic back-to-back.

MORE TESTING. Unsure what to make of this revelation, I set up another series of tests. Using a digital milliammeter in series with the capacitor and a digital voltmeter in parallel, I connected each capacitor in series with a 4.3Ω load and measured the current and voltage for the capacitor at each frequency. The test setup used is shown in *Fig. 9*. Another bit of BASIC programming allowed me to separate the effect of reactance from that of ESR. The results agreed with the earlier measurements. Still not satisfied, I connected a dual FFT analyzer as shown in *Fig. 10* and measured the transfer function of each capacitor at multiple frequencies.[1] After some more programming, again the same results! Thus my first conclusion differed from what I had been led to expect: even in electrolytic capacitors, ESR decreases as frequency increases—but it does so at a rate less than proportionality, so that dissipation factor increases somewhat with frequency.

Mulling over this newfound knowledge, and realizing I had come no nearer to answering the practical question about using modern electrolytics in crossover networks, I then decided to plot the magnitude of the theoretical and actual transfer functions of each capacitor for frequencies from 20Hz–20kHz (*Figs. 11–21*).

For the film capacitors, there was no discernible difference between the theoretical and actual values. In other words, the capacitors behaved essentially ideally, having almost constant capacitance and negligible ESR. The electrolytics showed effects of ESR and capacitance shift to

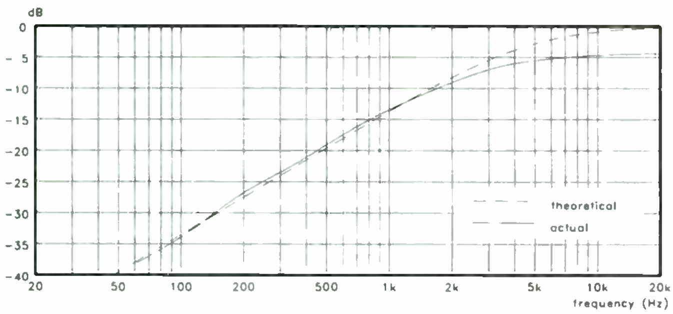


FIGURE 20: Dual 10μF/400V, 31-yr.-old electrolytic back-to-back.

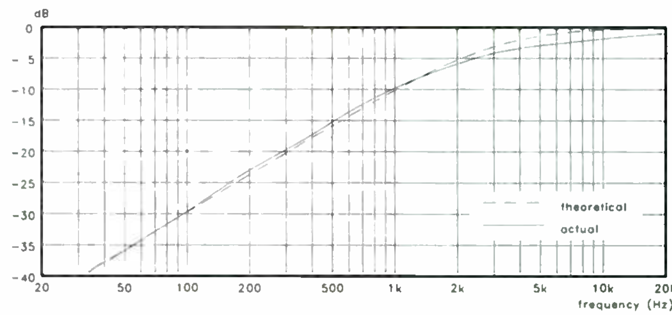


FIGURE 21: Parallel dual-10μF/400V electrolytic (*Fig. 19*) and 4μF/300V film.

a more significant extent. The two worst electrolytics for excess loss at 20kHz were numbers IVa (*Fig. 13*) and VIId (*Fig. 20*). These had excess losses of about 3dB at 20kHz. Some of the electrolytics had almost no excess loss at 20kHz, but did show several dB of excess loss about 1kHz. In fact, two of the electrolytics performed less ideally in the midrange than at the extreme top (*Figs. 14* and *15*). This is a result of the decrease in ESR with rising frequency discussed earlier.

You probably will notice that *Fig. 11* has two "ideal" curves. This capacitor showed enormous capacitance shift with frequency: as *Table 1* shows, it measured 9.28μF at 120Hz, and 6.73μF at 1kHz. This disparity was so great that I rechecked the measurement several times. Notice that at low frequencies the capacitor performed more like the 9.28μF curve, changing in the midrange to be more like the 6.73μF curve. While this particular capacitor was an extreme example, you can see from the table that capacitance shifts of 10% or so were not uncommon among the electrolytics. This would suggest careful testing when building a crossover using electrolytics, to make sure the crossover occurs at the intended frequency.

Figure 21 represents a parallel combination of capacitor Ic (4μF film) with VIId (10/10μF electrolytic can). Notice that although the actual curve approaches the theoretical one more nearly than does the curve in *Fig. 20* (VIId alone), it still differs by a couple of decibels at places. Thus the age-old attempt to compensate for using electrolytics by paralleling film capacitors does not seem to work perfectly.

You may have noticed that there are no curves for capacitor IIIb (6.8μF nonpolar) and IIIc (2.2μF nonpolar). These capacitors performed almost ideally. Therefore, we cannot necessarily assume that all electrolytics are bad. It's important to note, though, that these were relatively new capacitors.

AGING. Whenever electrolytic capacitors are discussed, the effects of aging come to mind. As an electrolytic ages, the electrolyte tends to dry out somewhat, reducing the capacitance. Leakage also typically increases with age. The oldest capacitors I tested were those in group VII, represented by *Figs. 17–20*. On the whole, these capacitors performed no worse than the other electrolytics. And as you can see from *Table 1*, the capacitance of each was higher than the rated value. (The "rated" capacitance for a nonpolar is half the value of each individual polar capacitor making it up.) Of course, electrolytics are typically produced with an actual capacitance that is higher than the stated value, in order to compensate for aging. Apparently this works, because no capacitor in this group showed low

REFERENCE

1. A transfer function is the equation describing what a device does with a signal as a function of frequency. In general, it contains both phase and magnitude information. The magnitude of the transfer function at a given frequency can be described as V_{OUT}/V_{IN}. So the magnitude of an amplifier's transfer function is just its gain as a function of frequency. In this case, the magnitude of the capacitor's transfer function is the dB decrease in voltage appearing across the load resistor, plotted as a function of frequency.

TABLE 2

VARIATION OF C AND D-F WITH APPLIED VOLTAGE (47μF/100V UNIPOLAR ELECTROLYTIC)

Bias Voltage	C @ 120 Hz	C @ 1kHz	DF @ 120 Hz	DF @ 1 kHz
0 VDC	47.2 µF	42.8 µF	0.099	0.392
1.5 VDC	47.2 µF	42.8 µF	0.105	0.436
6 VDC	47.8 µF	43.2 µF	0.108	0.441

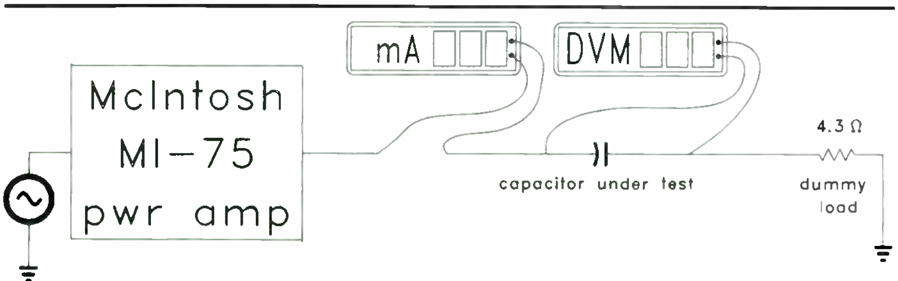


FIGURE 22: Test setup for effect of current on capacitance and D-F.

About the Author

Mr. Honeycutt resides in North Carolina with his wife, Betty Jane and three children. A physicist by training, he currently teaches Electronic Engineering Technology. Author of two textbooks, he also writes articles for several electronics magazines. His spare time is filled designing church, institutional and home sound systems.

capacitance, even though they ranged in age from 31-37 years.

DIELECTRIC ABSORPTION. Although touched upon earlier, perhaps some further words of explanation are in order. With some dielectric materials, especially tantalum dioxide, it is possible to charge a capacitor, discharge it through a short circuit, remove the short, and then draw some more charge from the capacitor by shorting it again. The refusal to completely discharge on the first try results from molecular polarization of the dielectric in ways that take some time after the initial discharge to return to normal. This phenomenon is called dielectric absorption. When a capacitor exhibits this effect, it will distort signals passing through it. (The effect is much like diagonal clipping in an AM detector, and occurs for much the same reason.)

Usually, dielectric absorption causes problems primarily in high-impedance circuits—a few k ohms or more. But since aluminum electrolytics are subject to the effect also, I decided to check for distortion of a sine wave caused by crossover capacitors. The test setup used was like the one shown in *Fig. 10*, but with a constant-frequency sine wave. With a test signal of 50W at 1kHz into a 4.3Ω load, the distortion of all the capacitors tested was below the residual distortion of the test equipment, which was –57dB. This translates into a residual THD of about 0.13%. Apparently, dielectric absorption is not a problem in crossover networks.

AND MORE TESTING. The last remaining question was whether capacitance and dissipation factor vary with the voltage across a capacitor. The answer is yes. Using the H-P 4261A, I measured two 47μF unipolar electrolytics. The results are given in *Table 2*. Notice that both capacitance and dissipation factor increase with applied DC voltage. Wondering whether this occurs in a nonpolar with AC applied, I measured the voltage at 120Hz and 1kHz for these two capacitors connected back-to-back. First with 10mA AC, and then with 250mA, I used the test setup shown in *Fig. 22*. The total impedance calculated from these measurements dropped from 54Ω to 48Ω at 120Hz, and rose from 7.00Ω to 7.24Ω at 1kHz. More puzzling behavior! When I tried pumping an ampere through the capacitor at 1kHz, the impedance dropped to 6.44Ω.

So the behavior of a nonpolar electrolytic does vary with the signal current, but not in a simple way. Further, since the reactance of this capacitor at 1kHz would be 6.74Ω, based upon the measured capacitance of 23.6μF at that frequency, the lower total impedance of 6.44Ω at 1A current indicates that the capacitance must be increasing with increased voltage, as for the DC case. The practical result of this in a crossover network would be a slight shift in crossover frequency and woofer/mid/tweeter balance as signal power changes. Since total impedance changed by over 11% within the current range tested, the effect could well be significant.

CONCLUSIONS. So what can we conclude from all this data? First, we are forced to admit that to use electrolytic capacitors in a passive crossover network is to compromise the product. Second, we can say that this compromise will show up as frequency-response irregularities for which we can compensate, and as signal-current-dependent variations for which we cannot. Third, we can conclude that film capacitors do not introduce measurable signal degradation in terms of either frequency-response anomalies or distortion. And finally, we cannot recommend paralleling film and electrolytic capacitors as a viable way of avoiding the problems that electrolytics introduce.

Now I'm going back to my friend's shop to wait for that sucker whose question got me into all this! Boy, have I got news for him!

THE DISAPPEARING LOUDSPEAKER: AN OPTIMIZATION ODYSSEY

BY VICTOR STAGGS

This pair of speakers is named for their sound, and I will explain later how they earned it. This loudspeaker design does exist, and two pairs have been built. They represent my effort to learn the art of loudspeaker system computer optimization, and were designed using analysis and optimization software I wrote for the research computer at U.C. San Diego. Since my thesis had nothing to do with loudspeakers, I owe *SB* readers some explanation.

The year I began working on my graduate thesis I had read a useful but particularly difficult paper and was digesting it when, for relaxation, I strolled through a swap meet one Saturday morning.

This was my downfall, because one seller was surrounded by crates of surplus (but brand new) Peerless drivers of all sizes. I couldn't resist picking up a pair of 6½-inch woofers for use in compact monitors, as such a design would be easier to heft around than a floor-standing behemoth.

The beauty of doing your own measurements, writing the software, and developing your own design is indulging your favorite engineering philosophy and techniques instead of following another's trail. This alone is worth much personal satisfaction, with the added pleasure of listening to the result.

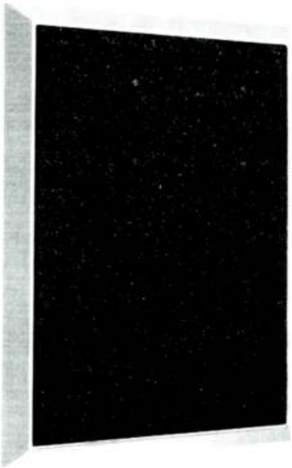


PHOTO 1: The prototypes have beveled edges and are covered with metallized Mylar film.

This speaker project will not be a simple one for beginners, but it will be rewarding for those who can handle a fairly exacting project and a lot of do-it-yourself labor. Let's start by understanding the engineering goals of this design.

ENGINEERING GOALS. Overall, my goal was to produce a pair of compact near-field monitors equalized for close-in listening with the speakers on stands, well clear of walls. I call this a near-field free-space design. The equalization was to account for box diffraction at low frequencies, woofer imperfections in the crossover range, and the tweeter's fundamental resonance.

As the design evolved, I found it practical to use the Linkwitz-Riley fourth-order acoustic response as the target function, for good directivity in the crossover region. The dividing network was to account for the phase response of the drivers all the way from DC to 10kHz, so as to minimize sources of interdriver time offset.

I wanted the network to present the drivers with a low source impedance, to preserve good electrodynamic voice-coil damping as an aid in suppressing the many "micro-resonances" that creep into real-world drivers. Deep bass from these little woofers was not something I demanded, since I wished to control their excursion and minimize harmonic distortions.

These speakers were to have a flat, smooth front panel with no offset to correct for time-alignment problems. I arranged the woofer network to correct as much as possible for any driver time offset, and the flat front panel would minimize scattering the tweeter sound.

Readers may recognize this kind of design as typical of small British mini-monitors, and atypical of the usual American acoustic suspension two-way speaker. However, an increasing number of domestic designs follow the European philosophy in our high-end marketplace.

TABLE 1

NEAR-FIELD MONITOR GENERAL SPECIFICATIONS

General Type:	Bass-reflex two-way, free-field equalized
Crossover Response:	Fourth-Order acoustic Linkwitz-Riley
Crossover Frequency:	2.2kHz
Frequency Range:	Approximately 60Hz–16kHz
Sensitivity:	84dB/W 8Ω at 1M on-axis, anechoic
Power Handling:	Approximately 50W per system mid-band
Application:	Free-field mounting, near-field monitoring
Drivers:	*Woofer*: Peerless TP165R, made in Denmark, 6½" nominal diameter, Butyl rubber surround, 15oz (0.43kg) magnet, 33mm voice coil diameter, graphite-loaded polypropylene cone, 8Ω nominal impedance *Tweeter*: Son Audax DTW12X9T25FFF, 1" dome diameter, ferrofluid cooled, doped fabric diaphragm, rectangular mounting flange, 8Ω nominal impedance

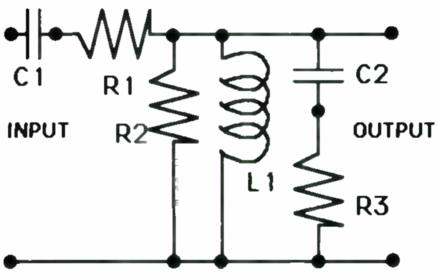


FIGURE 1: Tweeter dividing network.

Table 1 lists the general specifications of these near-field monitors.

THE TWEETER NETWORK. Several software packages were written to handle the measurement and design chores. The tweeter network design will make the need for them apparent. The tweeters chosen were the Audax DTW12X9T25FFF, which are rectangular-frame one-inch dome units with doped fabric diaphragms and ferrofluid cooling in the voice coils, which have a nominal 8Ω impedance. I believe in using inherently well-damped drivers, and I prefer these to the metal dome units with their ultrasonic ringing.

In order to perform computer optimization, we must know the complex input impedance of the driver's voice coil, so the dividing network will see its true load in the simulation. I measured the magnitude of the impedance and derived its equivalent circuit values using Thiele/Small theory and judicious guessing for the voice-coil inductance parameters. This turned out to be difficult for the tweeter because of the presence of the ferrofluid.

I wrote a software module that fits the impedance magnitude of the voice coil model to the actual measured value at all of the measured frequencies, using the Thiele/Small values as starting points. After the program adjusted the model parameters to achieve close agreement between the actual and predicted impedance magnitude, I had a complete list of the voice coil model values which Eugene Zaustinsky and I used in the equivalent circuit, which extends the Thiele/Small model to account for eddy-current effects in the pole pieces.

This voice-coil impedance didn't look very much like the one in the Audax data sheet, which called for further cross-check. I measured the tweeter's acoustic frequency response by close-miking it, and also predicted its response theoretically. The two agreed very well except at high frequencies, where dome resonances intruded. I decided my numbers were reliable and pressed on.

TABLE 2

TWEETER DIVIDING NETWORK PARTS LIST

ITEM	QTY.	REF.	PART	COMMENTS
1	1	C1	6.25μF	4.7 + 1.5μF
2	1	C2	0.85μF	0.82μF
3	1	L1	0.523MHz	
4	1	R1	3.17Ω, 5W	3.3Ω
5	1	R2	11.9Ω, 5W	12.0Ω
6	1	R3	6.29Ω, 2W	6.2Ω

Note: The items under comments are suggested values.

Figure 1 is the tweeter dividing network's schematic. Its values were optimized by software predicting the tweeter's output, using the measured acoustic frequency response, and loading the dividing network with the model voice coil parameters. The target function was the fourth-order Linkwitz-Riley high-pass response centered at 2.2kHz, using the tweeter's second-order high-pass acoustic response as one of the elements. This left the second-order network to complete the response and to pad down the tweeter's efficiency to match that of the woofer.

Readers will notice I haven't an L-pad in the traditional location in the network. This achieves better damping of the tweeter below its passband edge, by preserving a shunt across the voice coil at low frequencies. Thus, the tweeter sees a low source impedance except at frequencies near the crossover, where the ferrofluid damping is active anyway.

The Zobel network across the tweeter terminals has electrical values which were allowed to be free parameters and to participate in the optimization procedure, but even so they came close to those values that make the tweeter appear resistive at high frequencies.

TABLE 3

PEERLESS WOOFER TP1 65R PARAMETERS

QUANTITY	MEASURED VALUES	CATALOG VALUES
Nominal impedance	—	8Ω
DC Resistance	6.16Ω	6.0Ω
Voice-Coil Inductance	0.876mH @ 1kHz	0.8mH
Resonance Frequency	32.8Hz	38Hz
Total Q Factor	0.227	0.28
Acoustic Volume	40.29 liters	36 liters
Sensitivity, dB/WQ @ 1M re 8Ω	89.8dB	89.6dB

Note: The measured values are with the driver as loaded by the box air mass load. Sensitivity in the monitor enclosure will be 84dB due to diffraction equalization. The inductance varies with frequency.

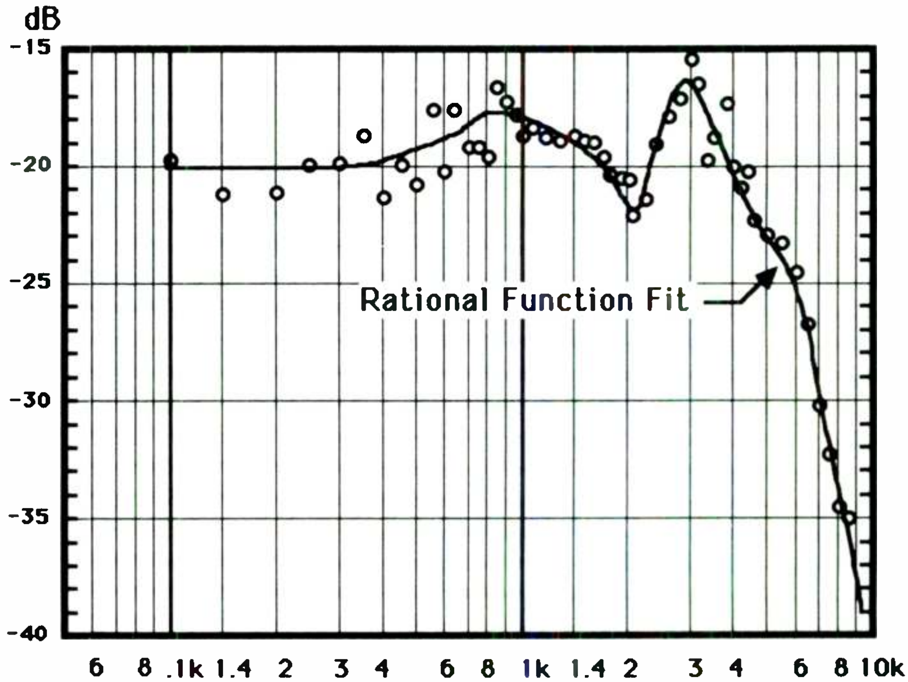


FIGURE 2: Measured woofer frequency response, corrected for low frequency box diffraction and woofer bass rolloff.

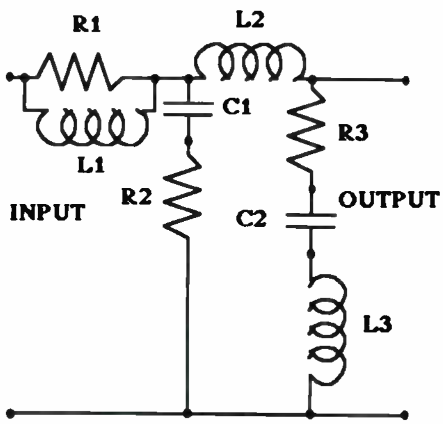


FIGURE 3: Woofer dividing network.

TABLE 4

WOOFER DIVIDING NETWORK PARTS LIST

ITEM	QTY.	REF.	PART	COMMENTS
1	1	C1	45.8μF	31 + 12 + 2.75μF *
2	1	C2	7.91μF	8.0μF *
3	1	L1	3.92MHz	
4	1	L2	1.43MHz	
5	1	L3	0.06MHz	
6	1	R1	4.3Ω, 5W	
7	1	R2	3.3Ω, 5W	
8	1	R3	1.5Ω, 2W	

*Note: These are suggested values.

With this network in place, tweeter phase matches the desired values all the way down to DC, and the diaphragm is well damped. The result is good transient behavior and articulation at all frequencies, which translates into transparency. A plot of the predicted tweeter response with the network in place looks so much like the target function that there is no point in presenting it in a figure.

THE WOOFER NETWORK. Most of the labor in this project went into measuring the woofer's impedance parameters and its acoustic response, and then optimizing its network. The reasons are the practical difficulties of measuring acoustic response in the real world, and because of the ambitious set of requirements and constraints placed on the woofer's network design. I'll try to make a long story short.

Step one with any woofer design begins with measuring its Thiele/Small parameters using a test box. This enables finding a practical bass alignment for it, and then constructing the chosen enclosure design. Now, we can make serious acoustic measurements of the woofer's response up to and beyond the crossover region, to provide data for optimal crossover network design.

The voice-coil impedance magnitude was measured and fit with my modeling software, so that I would know the complex impedance at all frequencies. The Thiele/Small parameters derived from this fit are shown in *Table 3*, alongside the factory data for the woofer, the Peerless TP165R.

Because small enclosures containing acoustic lining tend to be absorptive, the bass alignment is a lossy *ad hoc* tuning with the port resonance at 50Hz. The bass response rolls off below about 80Hz, so the port mostly controls cone excursion. Response is fifth-order, but because of the high damping, the rolloff looks like third-order near the cutoff frequency.

The outdoor acoustic measurement setup looked like a circus sideshow, and my friend (whose backyard and instruments I borrowed) threatened to take a picture of the whole affair. The speaker enclosure was mounted on the platform of a very tall, extensible photo tripod, with the microphone held at 1M on-axis by a plastic boom. Thus, the speaker could be positioned with its mike and then racked up as far off the ground as the tripod would take it.

Down on the ground were the measuring instruments and a Hewlett-Packard wave analyzer. This last instrument was used to track the oscillator frequency with a 4Hz wide bandpass filter in order to eliminate noise from jet planes, helicopters, lawn mowers, shouting children, and the next door neighbor's honking pet goose.

The frequency response was measured with a sine wave at many frequencies, including all the response peaks or dips. Both woofers were measured in their enclosures to check for driver consistency, which was excellent. Thus, crossover adjustments to suit the individual drivers were unnecessary. I also did a ground-plane type of measurement to get a feel for the effect of ground and building reflections on the free-space measurement.

The tweeter efficiency was measured at this time by feeding it a ⅓-octave band of pink noise at 1W re 8Ω input impedance, at a center frequency where the tweeter response, without its network, was flat.

No one could afford to optimize his woofer network using all of the measured frequencies, and so the response data were fit with a complex rational function. This averaged out the tiny ripples the network could not accommodate, and it helped to smooth through the distracting ground and building reflections. When optimizing the network, I chose a reduced set of 22 frequencies for efficiency.

The rational complex fit to the amplitude response had a second purpose. I could not measure the woofer's acoustic phase response, but, once I found the rational fit, the phase was simple to obtain by evaluating the rational function. Thus, I knew everything but the relative acoustic centers of the woofer and tweeter.

The rational function was a good fit to the data near and above the crossover frequency, as you can see in *Fig. 2*. In this figure, both the measurements and the fit have already had the box's low-frequency diffraction "spreading loss" subtracted out. We are left with what

Continued on page 46

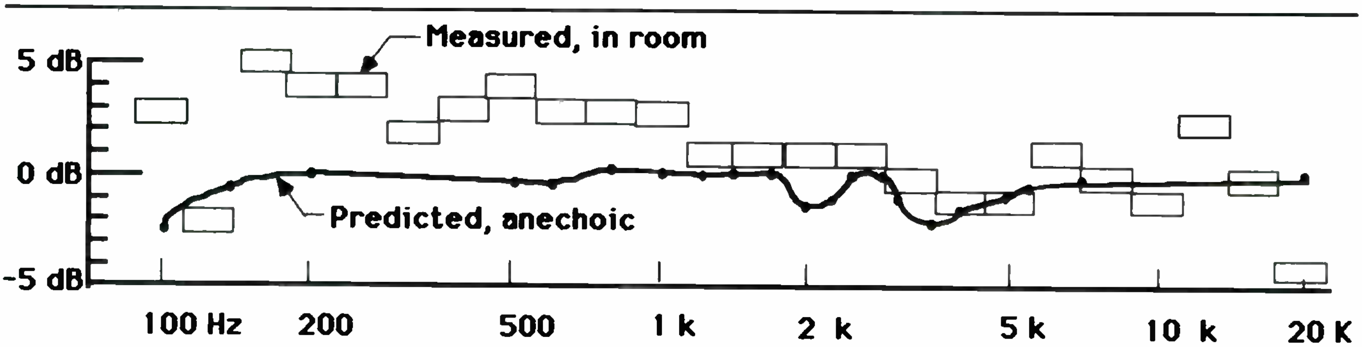


FIGURE 4: Predicted and measured frequency responses.

THE WILLIAMSON AMPLIFIER
D.T.N. Williamson

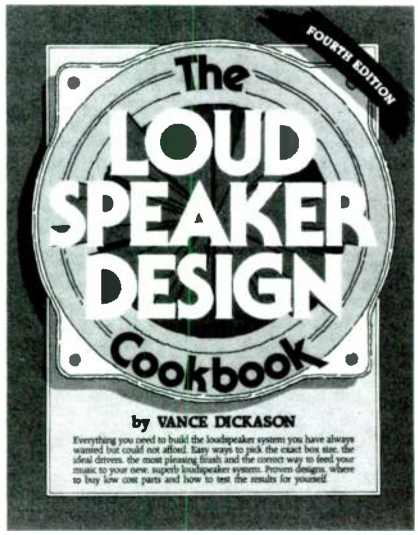
In April 1947, Williamson's power amplifier, using excellent-quality push/pull output tubes, a special output transformer, and a highly filtered power supply, became an overnight success. The author takes the reader deep into his design considerations, offering practical advice on how to build the units plus concise instructions on setting up the new amp. A cult classic.

Year 1947, 1990 — Pages 40
0-9624-1918-4 Softcover 8½ x 11 — $4.95

AUDIO AMATEUR LOUDSPEAKER PROJECTS
Audio Amateur Magazine

A collection of the 25 best speaker articles from Audio Amateur during the decade of the 1970s, proof that great designs are never out-of-date, a rich source of both theory and practice as well as design. The electrostatic and transmission-line pieces are particularly interesting and useful.

Year 1985 — Pages 135
0-8338-0193-7 Softcover 8½ x 11 — $20.00



THE LOUDSPEAKER DESIGN COOKBOOK
Vance Dickason

This new fourth edition of speaker designing's "bible" is twice as big as its predecessors and features brand new chapters on how loudspeakers work and loudspeaker design software. Other chapters cover closed and vented boxes, passive radiators, transmission-line systems, cabinet construction, drivers, crossovers, and loudspeaker testing, plus a complete resources listing. By far, the best-selling book in the field.

Year 1991 — Pages 152
0-9624-1917-6 Softcover 8½ x 11 — $29.95

THE AUDIO GLOSSARY
J. Gordon Holt

Authored by the founder of Stereophile magazine, this new best seller is a comprehensive overview of over 1,900 technical and subjective audio terms explained in precise yet at times humorous fashion. Three editions: Softcover (S); Hardcover with Dust Jacket (H); and Limited, Autographed Hardcover with Gold-Embossed Binding and Dust Jacket (L).

Year 1990 — Pages 152
0-9624-1914-1(S) Softcover 5¼ x 7½ — $9.95
0-9624-1914-1(H) Hardcover 5¾ x 8 — $17.95
0-9624-1914-1(L) Limited 5¾ x 8 — $30.00

BULLOCK ON BOXES
Robert M. Bullock III, with Robert White

An easy-to-read guide to designing and building vented-box systems based on Thiele/Small models, this book is a collection of Dr. Bullock's Speaker Builder magazine articles. Includes theory, model descriptions, plus related information from numerous sources, especially computer modeling from the authors' BOXRESPONSE and BOXMODEL programs.

Year 1991 — Pages 72
0-9624-1915-X Softcover 8½ x 11 — $10.95

LOUDSPEAKERS: THE WHY AND HOW OF GOOD REPRODUCTION
G.A. Briggs

This easy-to-read classic, last revised in 1949, introduces the reader to concepts such as impedance, phons and decibels, frequency response, response curves, volume and watts, resonance and vibration, cabinets and baffles, horns, room acoustics, transients, crossovers, negative feedback, Doppler and phase effects, and much more. A provocative survey of the right questions about sound reproduction.

Year 1949, 1990 — Pages 88
0-9624-1913-3 Softcover 5½ x 8½ — $6.95

KILLER CAR STEREO ON A BUDGET: AN EASY CURE FOR HO-HUM AUTO SOUND
Daniel L. Ferguson

Providing quick and tested methods for upgrading your car's sound, this manual provides an excellent, easy-to-understand, hands-on treatment of the cost-effective design and installation of high-quality vehicle sound systems. Included are Getting Started; Sedan, Hatchback, and Pickup Truck Systems; Speaker Cabinet Design; Filter Crossovers; and Installation.

Year1989 — Pages 118
0-9624-1910-9 Spiralbound 5½ x 8½ — $9.95

HARD-TO-FIND AUDIO BOOKS— AND MORE!

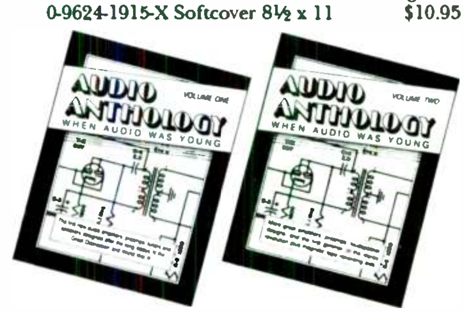
OLD COLONY SOUND LAB

❖IN USA, PLEASE ADD $2 SHIPPING FOR FIRST BOOK. IN CANADA, $4.50. OVERSEAS SURFACE, $6. OVERSEAS AIR, $9. ALL CUSTOMERS PLEASE ADD $.75 EACH ADDL BOOK

❖ASK TODAY FOR YOUR FREE CATALOG OF KITS, COMPONENTS, CABLE, SOFTWARE AND TEST CDs!

❖MC/VISA WELCOME!

OLD COLONY SOUND LAB
PO BOX 243, DEPT. B92
PETERBOROUGH, NH 03458
24-HOUR LINES:
TEL (603) 924-6371
FAX (603 924-9467



AUDIO ANTHOLOGIES, VOLS. 1-4: WHEN AUDIO WAS YOUNG
C.G. McProud, editor

These are the outstanding articles from Audio Engineering magazine during 1947-1957. Amps, preamps, tuners, speakers, and more. Vol. 1 covers 1947-50 and 38 articles. 2: 1950-1952, 45. 3: 1952-1955, 43. 4. 1955-1957, 34.

Year 1950, 1987 — Pages 124
0-8338-0195-3 Vol. 1 Softcover 8½ x 11 — $16.95

Year 1952, 1989 — Pages 124
0-8338-0197-X Vol. 2 Softcover 8½ x 11 — $16.95

Year 1955, 1990 — Pages 124
0-9624-1911-7 Vol. 3 Softcover 8½ x 11 — $16.95

Year 1957, 1991 — Pages 144
0-9624-1919-2 Vol. 4 Softcover 8½ x 11 — $16.95

Continued from page 44
would be measured with the speaker mounted in an infinite plane, except for small diffraction artifacts due to the box's front edges.

Figure 3 is the woofer dividing network schematic. It is not the first topology I tried, and I made several choices to improve the quality of the final optimization. At this time I started calling my choice the "last gasp" network. I had to relax the constraint of seeing a constant 8Ω impedance at the complete speaker system input terminals, so this design now is 8Ω below the crossover frequency and 4Ω above, with a smooth transition between the two ranges.

The dividing network software model was presented with a complex speaker impedance load that contained virtually all of the effects of the enclosure, including the port and the absorptive lining losses. This level of detail was required in order to produce a trustworthy set of optimal dividing network values.

I followed a two-step procedure to design the woofer-dividing network. First, the circuit values were adjusted to produce an optimal fit to the Linkwitz-Riley fourth-order low-pass acoustic response.

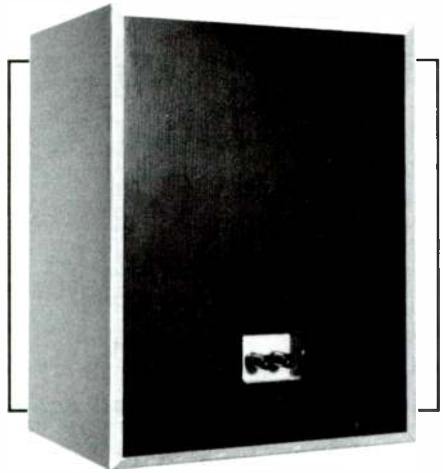


PHOTO 2: The prototype has only the terminals on the rear. The later version also has the port in the rear.

Then the tweeter response was added, and the circuit values were refined to produce a flat-summed response, including the effects of a driver time-offset value of one inch, based on a judicious guess about driver geometry measurements and a survey of the literature.

Actually, the network values did not change appreciably when I dialed in the time offset, because the Linkwitz-Riley response is already the one least sensitive to driver offset. Furthermore, the woofer network is minimum-phase, and it cannot act as a pure time delay.

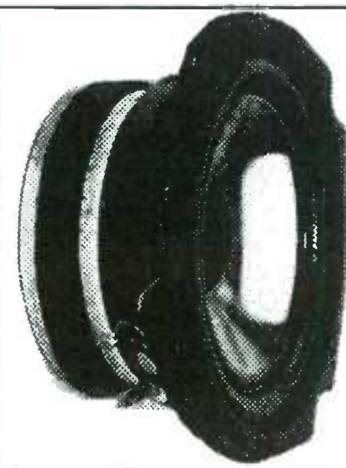
I found in my research that, for this woofer in this configuration, I got a better fit and a simpler circuit by omitting the Zobel network for the woofer voice coil. Therefore, the voice-coil inductance is an active part of this dividing network's optimal behavior.

A walk through the woofer network schematic of *Fig. 3* may be instructive. The first four elements form the box spreading loss correction circuit. They shelve down the woofer efficiency from the lower midrange on up in frequency. A two-element circuit would have sufficed here, but by making it into a generalized L-pad, we improve the low impedance presented to the successor elements in the network. Also, the use of C1 and R2 results in a smaller value of L1, which makes L1 cheaper and also less likely to experience core saturation with high input currents.

L2 and the following branch to ground are the elements of a classic second-
Continued on page 51

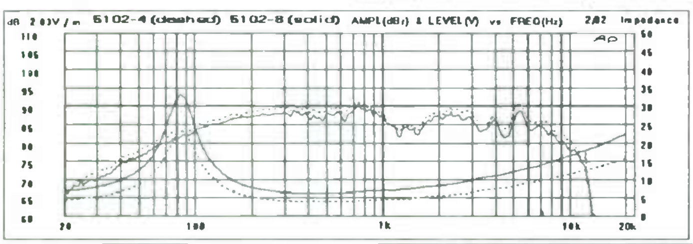
Madisound 5102
4.5" Polypropylene Bass-Mid 4 or 8 Ω

	5102-4	5102-8
Fs (Hz)	76	78
Rscc (Ω)	3.46	5.7
VcL (mH@1K)	0.09	.23mh
Qms	2.78	2.33
Qes	.43	.44
Qts	.37	.37
Mmd (g)	7.4	7.1
Cms (μm/N)	560.09	561.21
Vas (Ltrs)	3.57	3.57
Efficiency (db 1w/1m)	87.6	89.1
Xmax	3.5mm pk	
Power	50 w	
Magnet	12 oz	
Voice Coil	1" 2-Layer Kapton	
Cone	Black Poly	
Surround	Foam	
Cutout/Depth	4.25"/2"	
Price	**$20.00**	



	Vented		Sealed	
	4 Ω	8 Ω	4 W	8 Ω
VB ltrs	3	3.5	1.4	1.8
FB Hz	84	78	~	~
F3 Hz	84	79	145	136
Port Diameter	1"	1"	Qtc= .7	Qtc= .7
Port Length	2.4"	2"	~	~

**5102 4 / Dashed; 5102 8 / Solid

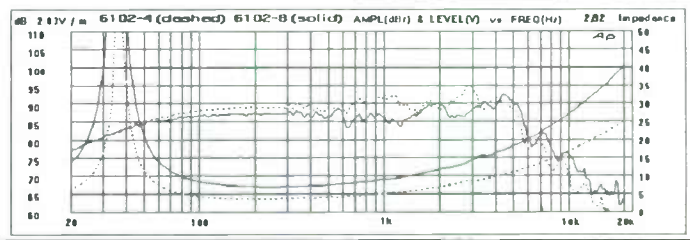


Madisound 6102
6.5" Polypropylene Woofer 4 or 8 Ω

	6102-4	6102-8
Fs (Hz)	30	30
Rscc (Ω)	3.35	6.6
VcL(mH@1K)	.087	0.18
Qms	6.6	7.5
Qes	.35	.45
Qts	.33	.42
Mmd (g)	14.5	11.2
Cms (μm/N)	1812.8	2312.9
Vas (Ltrs)	39	49.7
Efficiency (db 1w/1m)	87	87
Xmax	3.5mm pk	
Power	50 w	
Magnet	12 oz.	
Cone	Black Poly	
Surround	Foam	
Voice Coil	1" 2-Layer Kapton	
Cutout/Depth	5.62"/2.87"	
Price	**$24 00**	

	Vented		Sealed	
	4 Ω	8 Ω	4 W	8 Ω
VB ltrs	18	40	11.4	28
FB Hz	36	30	~	~
F3 Hz	41	33	58	47
Port Diameter	1.5"	2"	Qtc= .7	Qtc= .7
Length	4.6"	5.2"	Rg=.4	Rg=.4

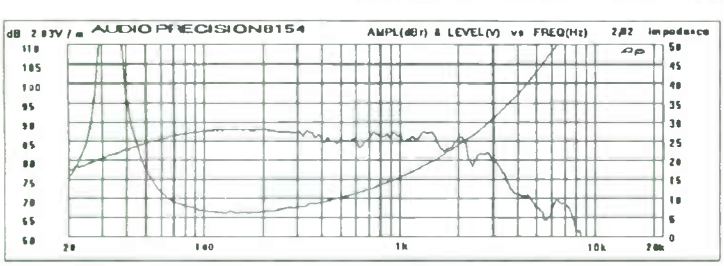
**6102 4 / Dashed 6102 8 / Solid



Madisound 8154—8" Polypropylene Woofer 8 Ω

Fs	30.6Hz
Rscc	4.55Ω
VcL @1K	.25mh
Qms	9.7
Qes	.28
Qts	.27
Mmd	34g
Cms (μm/N)	798.5
Vas	49.5 Liters
Efficiency	89db 1w/1m
Xmax	3.5mm pk
Power	75 w
Magnet	20 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 4-Layer Kapton
Cutout/Depth	7.12"/3.37"
Price	**$31.00**

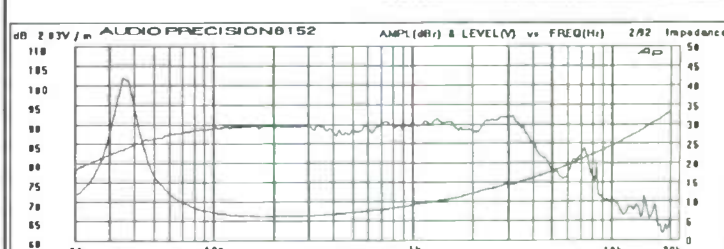
8154 B4 Alignments			
	Rg = 0	Rg = .4	Rg = .7
Vb Liters	15	19	23
F_3 Hz	51	45	42
Fb Hz	46	43	40
Port Dia	2"	2"	2"
Length	6.1"	5.3"	5.1"



Madisound 8152—8" Polypropylene Woofer 8 Ω

Fs	33Hz
Rscc	5.1Ω
VcL @1K	.13mh
Qms	3.5
Qes	.45
Qts	.4
Mmd	23g
Cms (μm/N)	889
Vas	55 Liters
Efficiency	89db 1w/1m
Xmax	2.5mm pk
Power	75 w
Magnet	20 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 2-Layer Kapton
Cutout/Depth	7.12"/3.37"
Price	**$31.00**

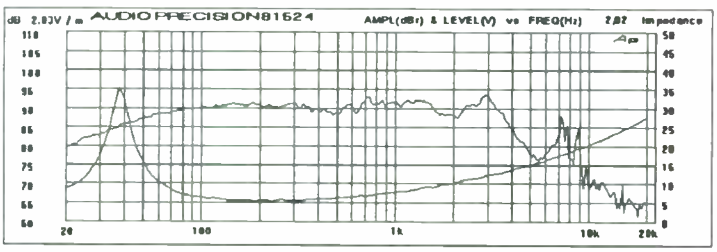
8152 B4 Alignments			
	Rg = 0	Rg = 0	Rg = .4
Vb Liters	30	50	63
F_3 Hz	43	35	33
Fb Hz	34.8	34.8	32.5
Port Dia	2"	2"	2"
Length	5.1"	2.5"	2.2"



Madisound 81524—8" Polypropylene Woofer 4Ω

Fs	36Hz
Rscc	3.7Ω
VcL @ 1K	.1mh
Qms	3.9
Qes	.44
Qts	.4
Mmd	22g
Cms (μm/N)	819.59
Vas	51 Liters
Efficiency	89db 1w/1m
Xmax	2mm pk
Power	75 w
Magnet	20 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 2-Layer Kapton
Cutout/Depth	7.12"/3.37"
Price	**$31.00**

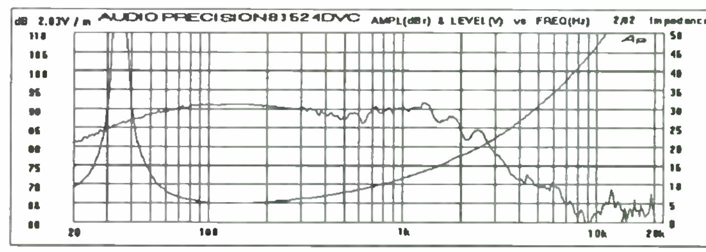
81524 B4 Alignments			
	Rg = 0	Rg = 0	Rg = .3
Vb liters	28	45	56
F3 hz	45	38	33
Fb hz	37	37	34.8
Port Dia	2"	2"	2"
Length	4.7"	2.3"	2"



Madisound 81524DVC—8" Dual Voice Coil Polypropylene Woofer 4Ω/4 Ω

Fs	31.7Hz
Rscc	3.5Ω
VcL @ 1K	.34mh
Qms	9.2
Qes	.32
Qts	.31
Mmd	38g
Cms (μm/N)	631.44
Vas	39.2 Liters
Efficiency	87.5db 1w/1m
Xmax	5mm pk
Power	80 w 40/40
Magnet	20 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1" 2-Layer Kapton
Cutout/Depth	7.12"/3.37"
Price	**$34.00**

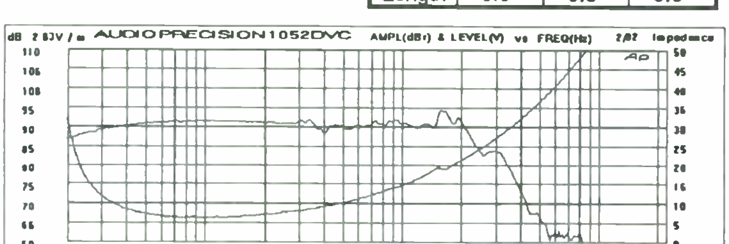
81524DVC B4 Alignments			
	Rg = 0	Rg = .5	Rg = 1
Vb liters	22	33	46
F3 hz	39	34	30
Fb hz	34	33.5	30
Port Dia	2"	2"	2"
Length	6"	5"	4.3"



Madisound 1052DVC—10" Dual Voice Coil Polypropylene Woofer 8Ω/8Ω

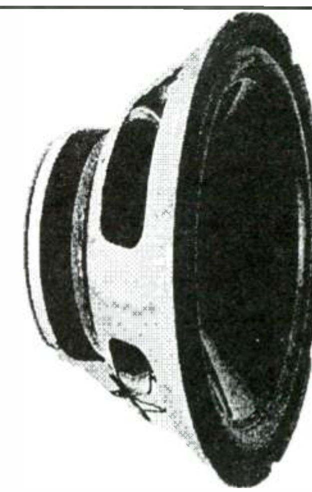
Fs	20.4Hz
Rscc	6.1Ω
VcL @1K	.46mh
Qms	3.68
Qes	.28
Qts	.26
Mmd	46g
Cms (μm/N)	1220.1
Vas	197 Liters
Xmax	6mm pk
Efficiency	90db 1w/1m
Power	50/50 w
Magnet	30 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 2/2-Layer Kapton
Cutout/Depth	9.12"/4.45"
Price	**$41.00**

1052DVC QB3 Alignments			
	Rg = 0	Rg = .5	Rg = 1
Vb Liters	47	57	69
F_3 Hz	38	35	33
Fb Hz	31	28.6	26.8
Port Dia	3"	3"	3"
Length	9.9"	9.3"	8.6"

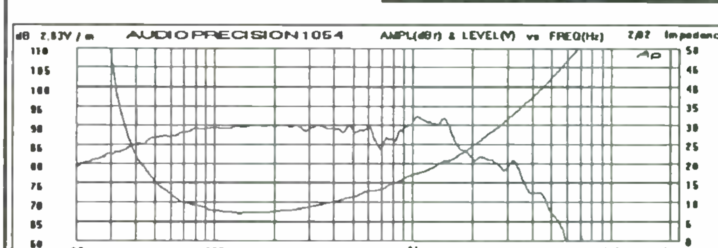


Madisound 1054—10" Polypropylene Woofer 8 Ω

Fs	24.6Hz
Rscc	6Ω
VcL @1K	.24mh
Qms	4.07
Qes	.25
Qts	.237
Mmd	42g
Cms (μm/N)	997.57
Vas	160 Liters
Xmax	3.5mm Pk
Efficiency	92db 1w/1m
Power	125 w
Magnet	30 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 4-Layer Kapton
Cutout/Depth	9.12"/4.45"
Price	**$38.00**



1054 QB3 Alignments			
	Rg = 0	Rg = .5	Rg = .9
Vb Liters	29	35	42
F_3 Hz	52	46	43
Fb Hz	41.7	38	35.7
Port Dia	3"	3"	3"
Length	8.7"	8.3"	7.8"

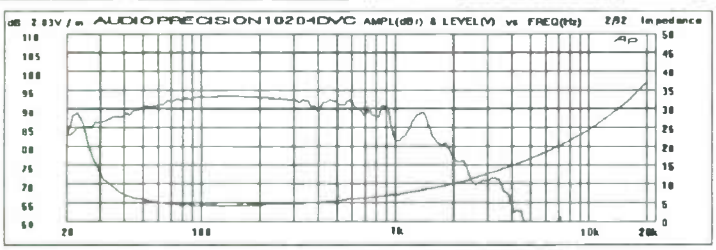


Madisound 10204DVC—10" Dual Voice Coil Woofer 4Ω/4Ω

Fs	21.2Hz
Rscc	3.6Ω
VcL @1K	.35mh
Qms	3.5
Qes	.21
Qts	.2
Mmd	50.4g
Cms (µm/N)	1045.4
Vas	168 Liters
Xmax	5 mm Pk
Efficiency	90.7db 1w/1m
Power	100/100 w
Magnet	40 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2" 2/2-Layer Alum.
Cutout/Depth	9.12"/4.45"
Price	**$49.00**

Can be used with both voice coils in series for 8Ω or in parallel for 2Ω

10204DVC B4 Alignments			
	Rg = 0	Rg = .5	Rg = 1
Vb liters	21	30	42
F3 hz	52	45	39
Fb hz	41.5	36.8	33
Port Dia	2.5"	2.5"	2.5"
Length	8.7"	7.4"	6.3"

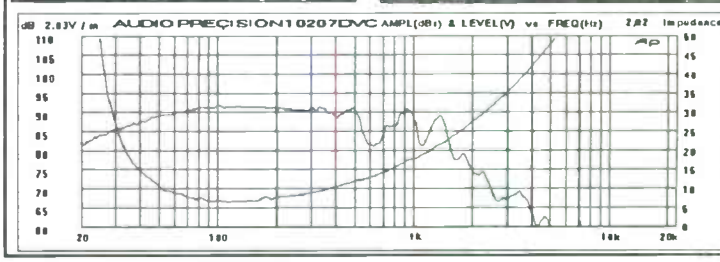


Madisound 10207—10" Dual Voice Coil Woofer 8Ω/8 Ω

Fs	16.2Hz
Rscc	5.7Ω
VcL @1K	.51mh
Qms	3.43
Qes	.23
Qts	.22
Mmd	57g
Cms (µm/N)	1138.6
Vas	184 Liters
Xmax	5 mm Pk
Efficiency	89.4db 1w/1m
Power	200 w 100/100
Magnet	40 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2" 2/2-Layer Kapton
Cutout/Depth	9.12"/4.45"
Price	**$49.00**

Can be used with both voice coils in series for 16Ω or in parallel for 4Ω

10207DVC QB3 Alignments			
	Rg = 0	Rg = .5	Rg = 1
Vb liters	26	33	40
F3 hz	45	41	37
Fb hz	35.5	32.5	30
Port Dia	2.5"	2.5"	2.5"
Length	9.5"	9"	8.6"

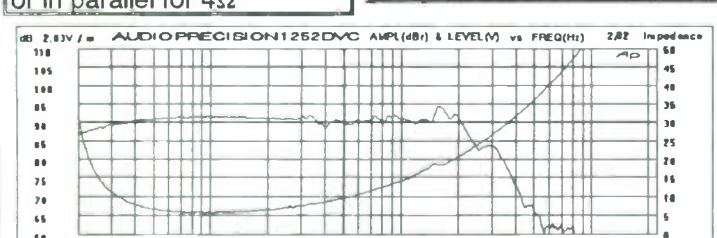


Madisound 1252DVC—12" Dual Voice Coil Woofer 8Ω/8Ω

Fs	15Hz
Rscc	5.6Ω
VcL @1K	.3mh
Qms	4.1
Qes	.39
Qts	.36
Mmd	78g
Cms (µm/N)	1331.4
Vas	533 Liters
Xmax	6mm pk
Efficiency	88.5db 1w/1m
Power	100 50/50w
Magnet	30 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	1.5" 2/2-Layer Kapton
Cutout/Depth	11.12"/5.0"
Price	**$42.00**

Can be used with both voice coils in series for 16Ω or in parallel for 4Ω

1252DVC B4 Alignments				
	Rg=0	Rg=0	Rg=0	Rg=0
Vb Ltrs	85	100	130	142
F_3 Hz	32.2	31	30	26
Fb Hz	QTC	QTC	QTC	17
Port Dia	.96	.9	.8	3"
Length	Sealed			11"

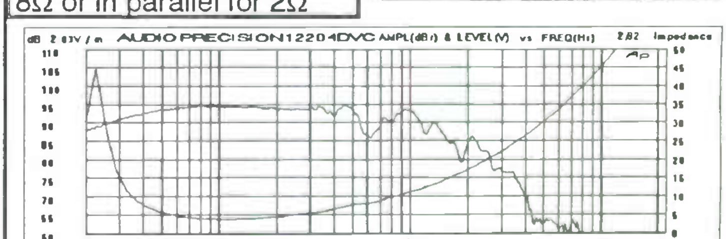


Madisound 12204—12" Dual Voice Coil Woofer 4Ω/4Ω

Fs	22.8Hz
Rscc	3.6Ω
VcL @ 1K	.26mh
Qms	4.58
Qes	.42
Qts	.38
Mmd	68.8g
Cms (µm/N)	550.6
Vas	220 Liters
Xmax	5 mm Pk
Efficiency	90.3db 1w/1m
Power	200 100/100 w
Magnet	40 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2" 2/2-Layer Kapton
Cutout/Depth	11.12"/5.0"
Price	**$53.00**

Can be used with both voice coils in series for 8Ω or in parallel for 2Ω

12204DVC B4 Alignments					
	Rg=0	Rg=.5	Rg=.5	Rg=0	Rg=.5
Vb Liters	85	85	100	113	142
F_3 Hz	42	38	37.5	31	28
Fb Hz	QTC			24	21.6
Port Dia	.75	.85	.8	3"	3"

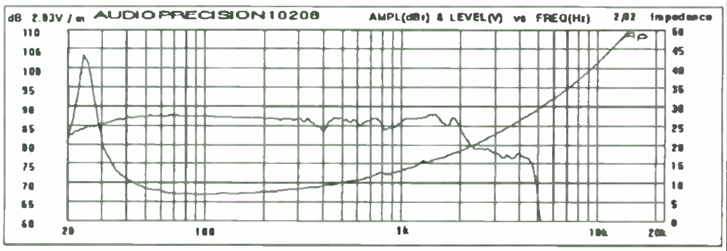


Madisound 10208—10" Sealed Box Woofer 8Ω

Fs	24Hz
Rscc	5.7Ω
VcL @1K	.13mh
Qms	4.62
Qes	.62
Qts	.54
Mmd	45g
Cms (μm/N)	900.5
Vas	145 Liters
Efficiency	87.5db 1w/1m
Xmax	6.5 mm pk
Power	100 w
Magnet	30 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2" Kapton
Cutout/Depth	9.12"/4.45"
Price	**$49.00**

10208 Sealed Box Alignments				
	Rg=0	Rg=.5	Rg=0	Rg=.5
Vb Ltr	99	99	142	142
F3 hz	32	31	31.3	29.4
Qtc	.86	.93	.78	.84

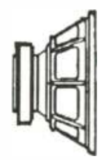
The use of fill will reduce the Qtc. This driver may be okay for free air applications to 45 Hz.



Madisound Driver Measurements

- All measurements made in a 37 m^3 anechoic chamber equalized to give response for an infinite baffle.
- All frequency responses measured corresponding to 2.83Vrms @ 1 meter, same voltage for 4Ω and 8Ω drivers.
- Dual voice coils are measured at 2.83Vrms per coil.
- Dual voice coil Theil-Small parameters are measured with voice coils in series using Delta Mass method with Audio Precision and Leap.
- Suggested box alignments are sometimes given with an (Rg) value, which is added resistance from inductors in series with the woofer. If you need specific box alignments, please call.

Ordering Information: All speaker orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment, and personal checks must clear before shipment. Add 10% for shipping charges. Residents of Alaska, Canada and Hawaii, and those who require Blue Label air service, please add 25%. There is no fee for packaging or handling, and we will refund to the exact shipping charge. We accept Mastercharge or Visa on mail and phone orders.



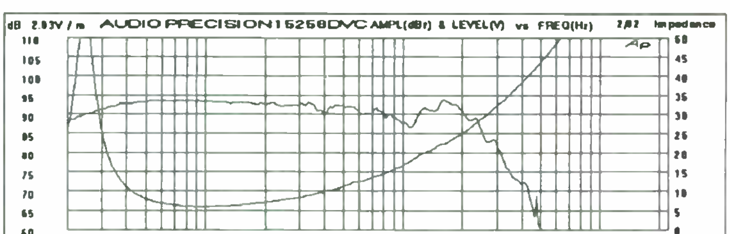
Madisound Speaker Components
Box 44283 (8608 University Green)
Madison, WI 53744-4283
Voice: 608-831-3433
Fax: 608-831-3771

Madisound 15258DVC—15" Dual Voice Coil Woofer 8Ω/8Ω

Fs	22.5Hz
Rscc	5.5Ω
VcL @1K	.36mh
Qms	5.35
Qes	.52
Qts	.47
Mmd	121.5g
Cms (μm/N)	367.38
Vas	220 Liters
Xmax	5.5 mm pk
Efficiency	91db 1w/1m
Power	200 100/100 w
Magnet	60 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2 (2") Kapton
Cutout/Depth	13.87"/6.0"
Price	**$73.00**

15258DVC Sealed Box Alignments				
	Rg=0	Rg=.5	Rg=0	Rg=.5
Vb Liters	100	100	142	142
F_3 Hz	37.8	36.6	35.4	33.8
Qtc	1.03	1.12	.9	.98

It is recommended to use fill and flow resistive vents with this driver.

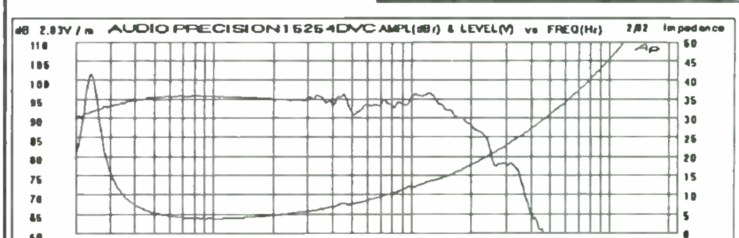


Madisound 15254DVC—15" Dual Voice Coil Woofer 4Ω/4Ω

Fs	23Hz
Rscc	3.7Ω
VcL @1K	.25mh
Qms	5.71
Qes	.47
Qts	.44
Mmd	122g
Cms (μm/N)	346.1
Vas	347 Liters
Xmax	5.5 mm pk
Efficiency	91.5db 1w/1m
Power	200 100/100 w
Magnet	60 oz.
Cone	Black Poly
Surround	Foam
Voice Coil	2 (2") Kapton
Cutout/Depth	13.87"/6.0"
Price	**$73.00**

15254DVC Sealed Box Alignments				
	Rg=0	Rg=.5	Rg=0	Rg=.5
Vb Liters	100	100	142	142
F_3 Hz	40	37.7	38	35
Qtc	.92	1.04	.8	.91

It is recommended to use filling and flow resistive vents with this driver



Continued from page 46
order low-pass network, except for the presence of L3 and R3. These extra parts cause the network response to roll up above 4kHz to compensate for the woofer's own rolloff, preserving a constant 24dB/octave slope in the acoustic response.

This crossover achieves the desired woofer acoustic amplitude and phase all the way up to 10kHz, the limit of measurement. This satisfies one of my design goals, which was to test the audible effect of phase control far beyond the crossover frequency region.

FINAL MEASUREMENTS. *Figure 4* shows the predicted frequency response and its measured counterpart. The measurement used the ⅓-octave pink noise technique in a large listening room with good wall damping, using the orbital method to average through room modes. The instrument had an LED readout with a resolution of 1dB, and so all of the measurement uncertainties are denoted by drawing rectangles to represent the data.

The orbital technique consisted of holding the microphone on-axis at 1M with its boom, and then rotating the speaker/microphone array through six aiming angles and averaging the frequency responses within the instrument. This yields an on-axis measurement with the reverberant field averaged at six room locations. Therefore, reverberation is present in the measurements, but smoothed.

I did no averaging through room height, so a floor (and ceiling) dip is evident in the vicinity of 125Hz. In spite of this, the measured curve shows the predicted dip just above the crossover frequency, plus the addition of reverberant amplitude at low frequencies due to the combination of box diffraction spreading and equalization. This is a good real-world result, and it shows that one can expect to tune the bass response to some extent by careful positioning of the speakers not too close to the walls.

The measurements were made on a completed loudspeaker with its grille in place. The grille was designed to integrate into the front panel anti-diffraction treatment, and so no measurements were made with it off. The ripples in the tweeter response are caused by the imperfections in the grille treatment.

As promised, the measured modulus of input impedance for the completed loudspeaker system, shown in *Fig. 5*, varies from a nominal 8Ω to a nominal 4Ω as frequency rises. Except for the usual bass-reflex double hump, the impedance is a benign load that will not be sensitive to the amplifier's output impedance nor to cable effects. This speaker is a good match for a vacuum-tube amplifier.

Apparently the port tuning is a bit low in frequency. The vent I used was of 1½″ I.D. plastic pipe cut to a length of 3⅜″, which is probably a good starting point for this design.

LISTENING IMPRESSIONS. After these systems were working as intended, I invited an audiophile friend to hear them. They performed according to design, delivering transparent sound with adequate, though not very extended, bass. In addition to see-through transparency, a near-holographic image between the speakers had depth, width, and solidity. That blob of air seemed to be occupied by people playing music, and if the listener closed his eyes, he felt enveloped within the recording environment rather than just listening to it.

A few days after my friend departed, I got a phone call from another friend, asking to hear my "disappearing loudspeakers." "What?" I asked. The caller explained that my friends were referring to my speakers this way because they seemed to disappear into the soundfield. So my audiophile friends trekked through my apartment to undergo the near-field monitoring experience.

Another quality evident in these speakers is their excellent low-level information recovery. This is due to the careful front panel layout and the treatment of the panel edges, in addition to the inherent driver damping preserved by the network design. I was surprised that these things could be so audibly effective, and that there could be such differences in low-level detailing between this design and many others.

I would not be so impressed with this design had I not already acquired plenty of experience with diffraction control during the design of my larger main loudspeakers. No matter how much I enjoy my large, full-range speakers, I am always rather sobered when I put up the little ones and hear what they can do.

These speakers are limited due to their size. At very high playback levels the midrange starts to sound hard and glassy, and this is probably due to harmonic distortion generated by running out of linear excursion. This is more evident with demanding music, such as the BIS CD of Vivaldi's *Four Seasons*, with

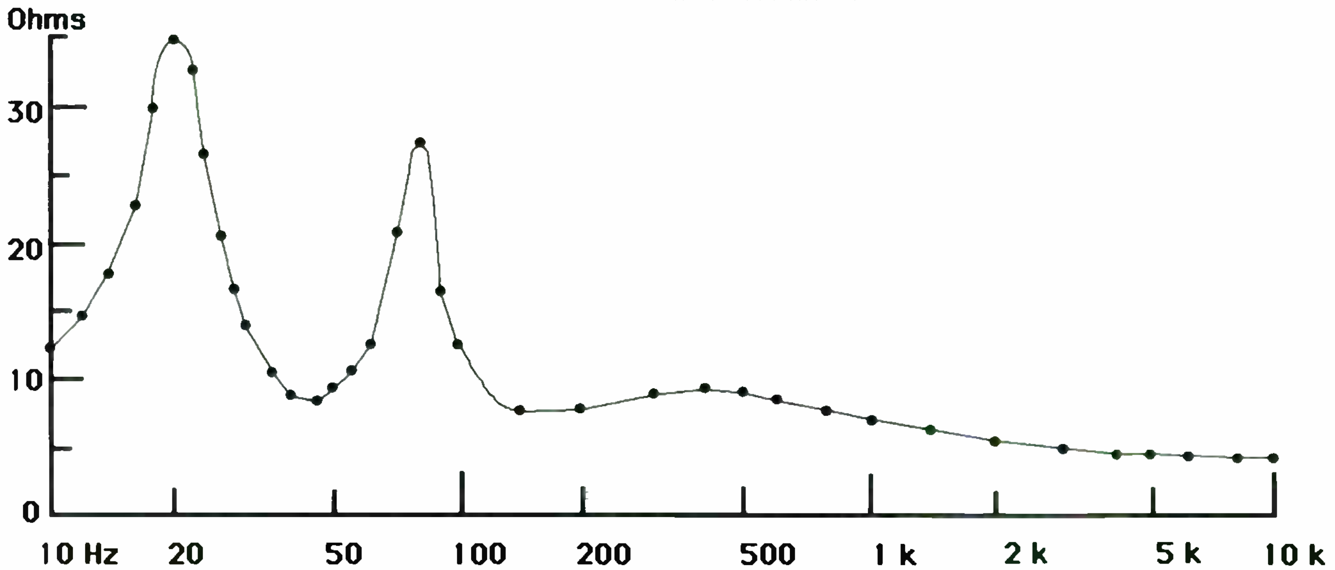


FIGURE 5: Modulus of system impedance vs. frequency.

its bowed double bass and original instrument-type violins. With less demanding music, such as the LP of The Doors' *L.A. Woman*, there is little problem. On most material, the speakers will play far louder than is pleasant to hear for any length of time.

WARNING! Before I summarize construction techniques, I must spell out legal limitations on the use of this design. I consider myself its owner, and I will permit no commercial use of it. This is because the University of California provides state funds for educational computer accounts, and these cannot be used by students or others to earn money. Designers are free to use my topologies as starting places for their own optimizations, but if anyone manufactures my design for profit, their profits may become the property of the University of California. So, amateurs, enjoy this project.

CONSTRUCTION. Two-views of the loudspeaker enclosure are shown in *Figures 6a* and *6b*. The details of rabbeted joinery are included for those constructors who have access to a router. If you use any power equipment, be sure to read the user's manual and heed all the safety precautions. It is also a good idea to wear hearing protectors when working around power tools, in addition to eye protection, so that you will still be able to enjoy your speakers when they are finished.

My pair of speakers have removable front panels, which make it easier to make changes during development. If you wish, you can make the back panel removable and glue the front solidly in place. This should result in fewer panel resonances.

BOX CONSTRUCTION. The dimensions are for panels composed of ¾-inch thick, "floor underlayment" grade particleboard. This grade, while cheap and dirty, has the best internal damping for loudspeaker use. The only difficulty is making strong joints and living with its crumbly nature.

For strong joints, I recommend the fully rabbeted construction. Straight joints are also practical as long as you use ¾-inch square stock to reinforce all corner joints. With rabbeted construction, reinforce the joints with triangular stock ¾-inch on a side, cut into 1½"-long glue blocks, added after assembly. Make 14 of these per cabinet.

The removable panel, front or back, is supported on ¾-inch square stock glued inside the entire box opening to form a ledge. This must be added carefully to provide a flat mounting surface, and I always add a gasket of ¾" vinyl electrical tape to the mating surface of the removable panel, cut out to clear the screw holes.

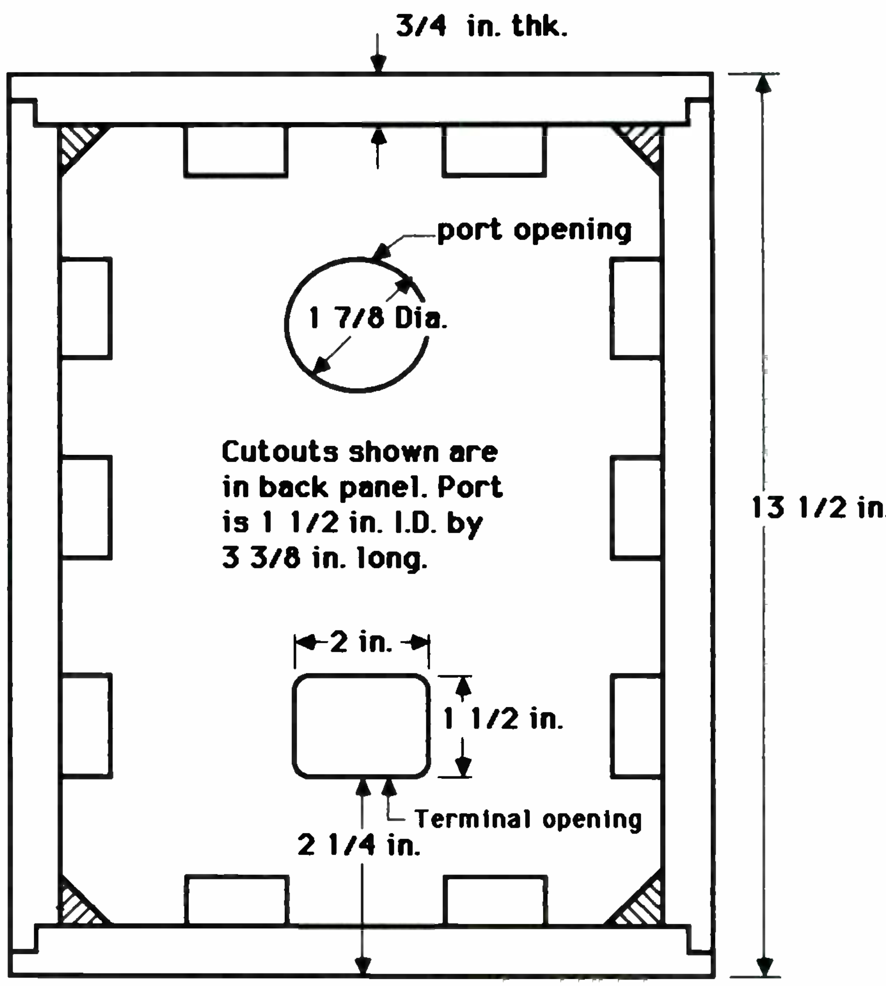


FIGURE 6a: Front sectional view.

Water-based carpenter's glue is ideal for joining the particleboard seams, as long as you observe precautions. Since the boards are porous, pre-glue all joints, letting the liquid soak in and dry while the panels are still apart. Sand mating surfaces smooth, and then glue anew and assemble. The first coat seals the surface, allowing the second to bond the joint firmly.

Add shaped stock to the cabinet's front edges for two reasons: rounded or beveled corners aid diffraction control and the crumbly particleboard edges will be sealed to avoid chipping.

If you intend to veneer the cabinets with wood or a composite, the beveled corners are easiest to work with. If you paint the cabinets or cover them with a plastic wrap, then the quarter-round molding is convenient. Also, mounting the port in the rear panel helps to keep the front panel free of sound-scattering edges and results in cleaner treble.

CUTOUTS AND MOUNTING. The driver mounting layout is shown in *Fig. 6c*. Be sure to make a left and a right front panel for mirror-imaged speakers. If you center the drivers to simplify the work, the low-diffraction performance of these speakers will be impaired and neither the acoustic measurements nor the optimized dividing network will be valid.

Changing the box's outer dimensions will similarly vitiate the optimal quality of this design. Only the depth can be changed without causing problems, and you can increase the front and rear panel thickness by maintaining a constant internal depth.

Holes must be drilled and countersunk for the #8 flat head wood screws that secure the removable panel. Position the panel in the enclosure and drill pilot holes through the panel and into the mounting ledge, to make sure all the holes match. Next, enlarge the panel's holes with a clearance drill. [Stanley's "Screwmate" does this job in one step.—Ed.]

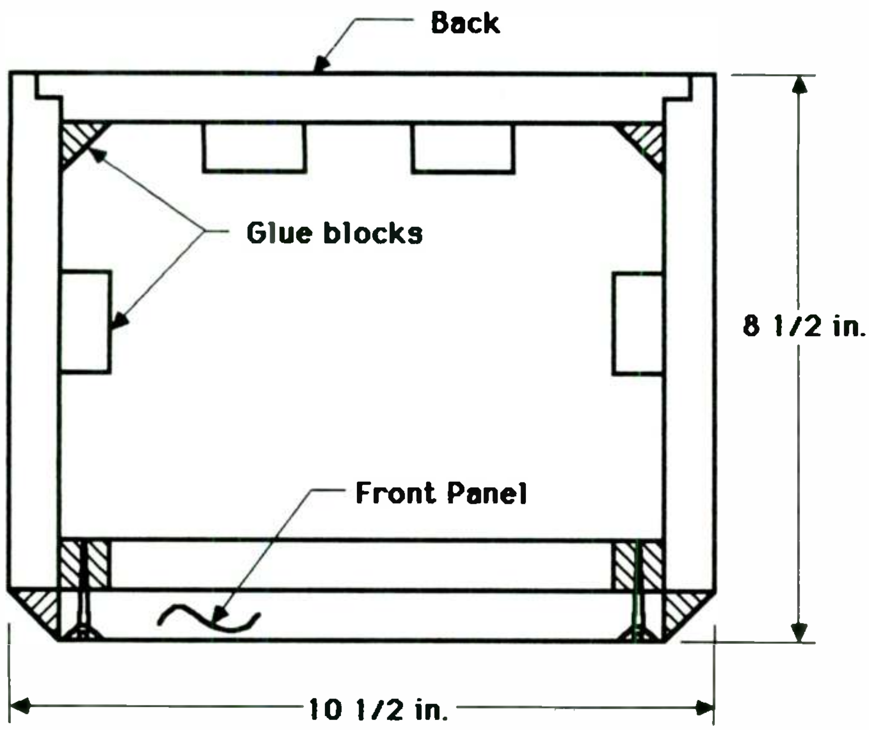


FIGURE 6b: Top sectional view for removable front.

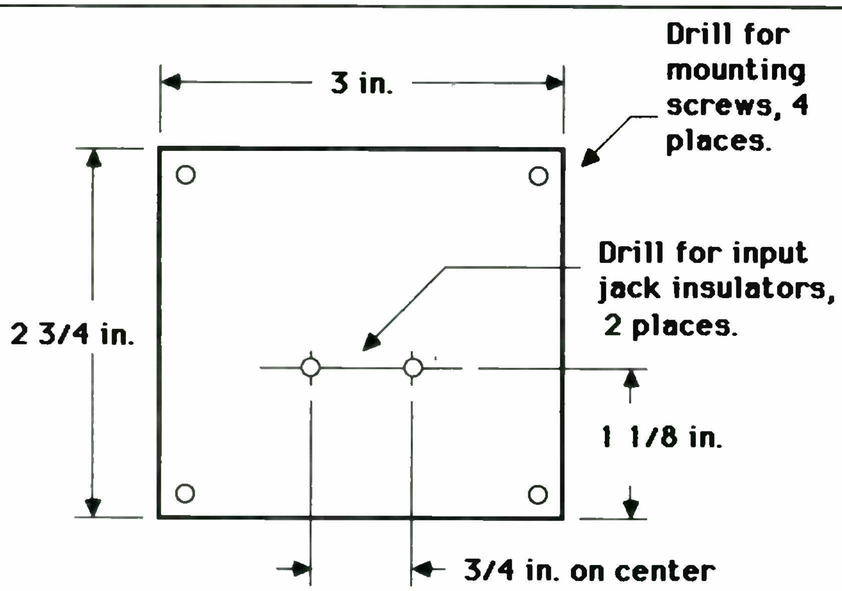


FIGURE 6c: Input terminal panel layout.

The rear access panel cutout is low in the box so that an inadvertent tug on the cables will not bring the speaker tumbling down quite so easily. The port is mounted in the rear panel about one-quarter of the way down. My terminal panel is made of ⅛″ surplus aluminum screwed in from the inside, with added RTV sealer. The metal looks classy, but ¼″ plywood or Masonite will work. The rear cutout prevents the five-way binding posts from protruding and protects them from damage.

Neither of the prototype pairs of my speakers is bi-wired, so each has one pair of input connectors. Two pairs for bi-wiring are possible with a larger cutout, but would make the cabinets more vibration susceptible. In such a case, it may be preferable to mount the dividing network in an external box and use connectors which extend through the full wood thickness.

LINING THE BOX. Line the interior of the box with 1″ polyurethane foam plastic, except for the rear of the front panel. It turns out that this fills nearly one half of the box internal volume, and accounts for much of the reflex port lossiness. Perhaps ½″ foam would do just as well, but to me the midrange sounds better with the full amount.

When purchasing the foam lining, try blowing through several samples to see how much the air resistance varies. Avoid those samples with little air resistance and those that you can't blow through. This test requires one to press his lips against the foam.

Install the foam lining with care. Some factories stuff their boxes loosely with polyester or glass wool, which I consider to be a construction error for a ported enclosure. The air's velocity will drag the resistive material along with it, and low frequency energy will be lost to flexing the fibers. In addition, the material may buzz at some frequencies.

My plan is to lay a square grid pattern of white glue on the inside surface to be covered with foam plastic, with a grid spacing equal to the foam thickness. Don't smear it out, because the glue bead must penetrate some distance into the foam to take hold. White (not carpenter's) glue is one of the very few adhesives I have found to work with urethane foam.

Hold each foam panel firmly on its mounting surface while drying by weighting it with a book or two. Cut the next foam panel with a razor blade or a very sharp knife while the previous one is drying. This is more efficient than it sounds. Use a metal straightedge and a scrap wood cutting surface.

DIVIDING NETWORK HINTS. The dividing network is quite a lot of circuitry to stuff into a small enclosure, but it can be done. A parts layout is shown in *Fig. 8*. The mounting board just barely fits within the foam lining, and some of the circuit elements are mounted in two layers. Terminal strips having tall straight lugs, rather than eyes, make this possible.

Inductor L1 for the woofer is a Madisound Sidewinder unit with a Ferrite core. All the others are Sidewinder air core units. L3 for the woofer can be anything, as it is a small value in series with a resistor. Some of the tweeter parts are wired piggyback on the tweeter network's inductor. A budget of 0.5Ω per inductor was allowed for in the design, but you needn't stick to this slavishly.

Note that some coils are mounted flat and some are on edge. This achieves compactness and minimizes the mutual inductance between coils. Avoid laying out coils either all lying down, or all on edge. This maximizes mutual inductance.

When building the crossover, pencil in the layout on the boards and drill for the terminal strip mounting screws first, before gluing the coils. Mount terminal strips at this time. Mix and match terminal strip sizes to get the right number of lugs. Mount the inductors with RTV silicone after snipping bobbin flanges to make straight edges for mounting ease. Use a generous, very strong bead on the edge-mounted units. The largest inductor, woofer L1, is mounted flat for support.

Put the boards aside overnight to allow the silicone to set completely. The next day you can mount the components and wire the connections. It is easiest to strip and connect the coil leads before finishing the rest of the wiring. The flying lead pairs going into the board as well as those out to the woofer and tweeter should be arranged as individual twisted pairs, which keeps them from radiating and receiving each other's magnetic fields. Untwisted wires have a surprisingly large effect on the sound.

ASSEMBLY AND TESTING. The crossover boards are glued to wood or fiber standoffs about 1″ long by ¾″ diameter. Empty thread spools will work, dowel stock or whatever you can find. This whole assembly can be glued or screwed onto the rear surface of the front panel, directly behind the tweeter. The standoffs prevent the coils from being unduly influenced by the soft iron in the tweeter's magnet, so do not make them too short.

A truly fastidious builder would measure the inductance of all of the coils while they are in position behind the tweeter, and adjust their values downward if there is interaction with it. The author was not this fastidious.

Teflon-insulated 18-gauge wire is ideal for the flying leads to the woofer and the tweeter. I crimped and soldered clip terminals onto these leads to avoid soldering to the woofer and tweeter tabs. The tweeter tabs must be bent back to allow the wiring to enter from the rear and clear the tweeter mounting hole with its wiring reliefs.

Adjust the wires so that the flying leads do not rub against anything that could cause them to buzz, such as the back of the woofer magnet. Make sure the crossover parts are either glued down or are supported away from the other parts by their leads. Tap the crossover boards to test for buzzing.

A checkout is wise before installing the crossover networks. We know that a "node" is a terminal where two or more components connect. It is a single node even if it consists of two terminals connected by a wire link, because all parts

Continued on page 56

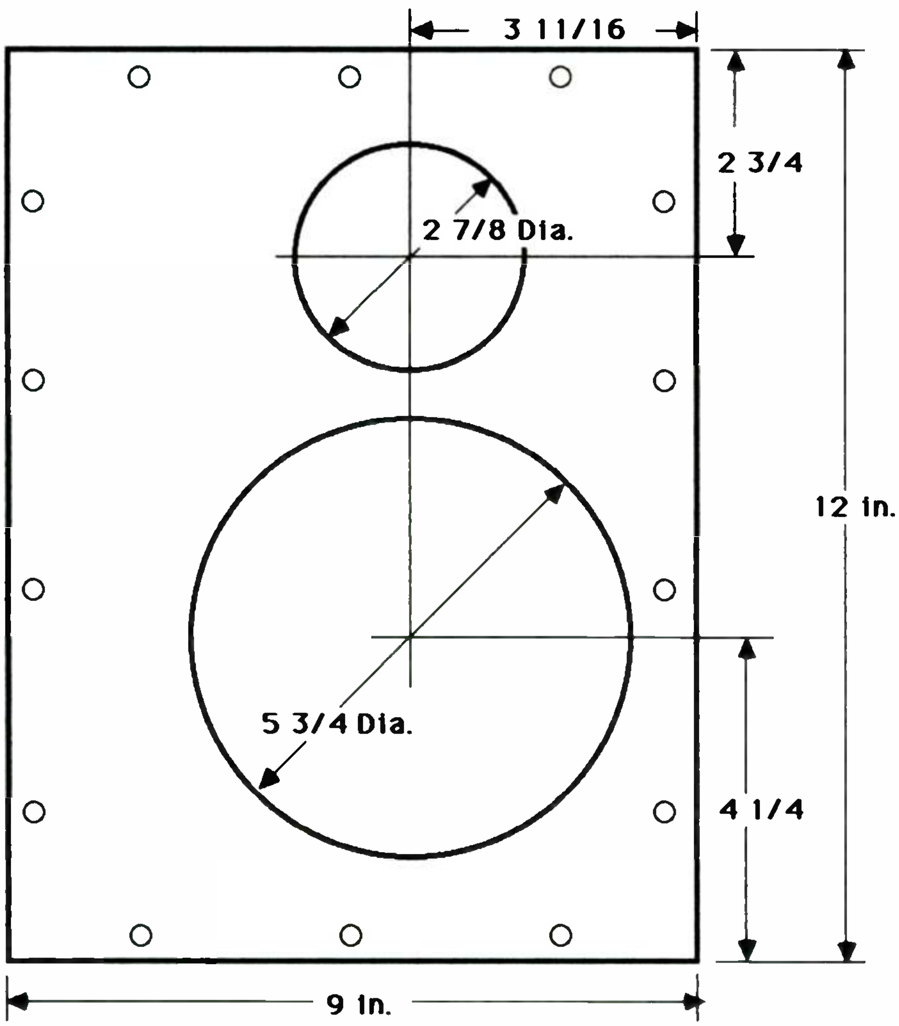


FIGURE 7: Left-hand front panel layout.

RESOURCES

Madisound Speaker Components
8608 University Green
PO Box 44283
Madison, WI 53744-4283
(608) 831-3433
Peerless woofers, Son Audax tweeters, crossover parts, custom wound inductors.

A & S Speakers
3170 23rd Street
San Francisco, CA 94110
(415) 641-4573
Peerless woofers, Son Audax tweeters, ask about crossover parts.

Meniscus
2442 28th Street S.W.
Wyoming, MI 49509
(616) 534-9121
Peerless drivers, crossover parts.

Note: This list is not exhaustive.

REFERENCES

1. Zaustinsky, Eugene, "Measuring and Equalizing Dynamic Driver Complex Impedance," AES Preprint 2299, Oct. 1985.
2. Elliott, Brian J., "Accurate Methods for Determining the Low-Frequency Parameters of Electro-Mechanical-Acoustic Transducers with BLI Excitation," AES Preprint 1432, Nov. 1978.
3. Vanderkooy, J., "A Model of Loudspeaker Impedance Incorporating Eddy Currents in the Pole Structure," AES Preprint 2619, March 1988.
4. Wright, J. R., "An Empirical Model for Loudspeaker Motor Impedance," *JAES* Vol. 38, No. 10, Oct. 1990.
5. Ureda, Mark, "Determination of Thiele/Small Parameters of a Loudspeaker Using Nonlinear Goal Programming," AES Preprint 1953, Oct. 1982.
6. Zaustinsky, Eugene, "Is Sophisticated Loudspeaker Crossover Design Possible Without Sophisticated Measurements?", AES Preprint 2032, Oct. 1983.
7. Olson, Harry F., "Direct Radiator Loudspeaker Enclosures," *JAES* Vol. 17, No. 1, Jan. 1969.
8. Gander, Mark R., "Ground Plane Acoustic Measurement of Loudspeaker Systems," AES Preprint 1648, May 1980.
9. Audio Engineering Society, Loudspeakers, an anthology of articles on loudspeakers from the pages of the *Journal* of the Audio Engineering Society, Vol. 1-Vol. 25 (1953–1977).

SpeakerBuilder

BACK ISSUES

1981 A Testing Unit for Speaker Parameters • Variable-Volume Enclosure • Thiele/Small Theory (in 3 parts) • Easy to Make Enclosure Using Concrete Blocks • The Tractrix Horn: Good Dispersion Bass From an Old Design • Diffuser Port for Small Boxes • Mini-Speaker Made From PVC Tubes • Closed vs. Vented Box Efficiency •Interview with P.G.A.H. Voigt • Dual 8" Symmetrical Air Friction Enclosure • Thiele/Small Calculator Computations • Thiele/Small Parameters for Passive Radiators •

1982 Transmission Line Theory • Thiele/Small Sixth-Order Alignments • The Quad 63 • Table Saw Basics • AR-1 Modifications • Active Crossovers and Phase • Three Transmission Line Speakers • A Beginner's First Speaker • How Passive Networks Interact with Drivers • Horn Loaded Heil • Phase Correcting Active Crossover • Wind Your Own Inductors • Series and Parallel Networks • High Performance Corner Speaker • Using Zobels to Compensate for Driver Characteristics •

1983 Building the Two-Way Dynaudio • A Crossover That Offsets Speaker Impedance • Using a Calculator for Box Design • Choosing a Calculator • A Simple Peak Power Indicator • A Small Horn Speaker • Audio Pulse Generator • How to Use Speaker Pads and Level Controls • An Easy-to-Build Voltmeter for Speaker Measuring • Nomograms for Easy Design Calculations • Interview with KEF's Raymond Cooke • Build a Simple Wattmeter • A New Type of Speaker Driver •

1984 Build an Aligned Satellite/Woofer System • BOXRESPONSE: A Program to Calculate Thiele/Small Parameters • Casting with Resins • A Phase Meter • An Interview with Ted Jordan • Building the Jordan-5 System • Self-Powered Peak Power Indicator • Closed Box Design Trade-offs • How to Build Ribbon Tweeters • Build a Dual Measurement Impedance Meter • A High-Power Satellite Speaker System • Build and Use a White/Pink Generator • Sound Pressure Level Nomographs •

1985 Compact Transmission Line Subwoofer • Bullock Crossovers, Passive and Active, three parts • Drive Attenuator, Computer-Designed • Curved Vertical Array • Sontek Powered Subwoofer • An Isobarik System •Modifying Strathearn's Ribbon • Ambience Systems • Experimental Transmission Line • Small Double-Chamber Reflex • Loose Walled Speaker • Modifying the Daline •

1986 The Edgar Midrange Horn • Sand-Filled Stands • Crossover Networks: Passive and Active • 5-Sided Boxes • A 2×4 Transmission Line • The Free-Volume Subwoofer • Notch Filters • By-Wiring the LS3/5A • A Push-Pull Constant Pressure System • Current and Power in Crossover Components • The Unbox (Egg) • Upgrade Speakerlab's S-6 Crossover • Measure Speakers with Step Response • A Gold Ribbon System • A Visit with Ken Kantor • A Tractrix Horn Design Program • Reviews: Audio Concepts " G"; Seven T.L. Midranges; Focal's Model 280; the Audio Source RTA-ONE.•

1987 A Compact TL Woofer • Frequency Response and Loudspeaker Modeling (three parts) • A Manual Coil Winder • The Model-One Speaker • Designing a Listening Room • A Six-Order Vented Woofer • Tapered Pipe Experiments • Visiting Boston Acoustics • A Vented Compound System • The Octaline • Spreadsheets for Speaker Design • In Memoriam: Richard Heyser (two parts) • Using Non-Optimum Vented Boxes • Building Speaker Stands • Evaluating Driver Impedance Compensation • Tuning Bass-Reflex • Six Woofers Compared • Bullock on Passive Crossovers: Alternate Bandpass Types • Fast, Easy Filter Calculations • A Mobile Speaker • Modifying the Polk 10 •

1988 Electronic Turns Counter • Two-Way Design • Minimus-7 Mod • Dome/Midrange/Tweeter Array• Plotting Complex Impedances • A Driver Design Primer • A Cabinet Primer • Tuning Up Old Systems • Low-Cost AR-3 Upgrade • Electronic Time Delay • Enclosure Shapes and Volumes • Minimum-Phase Crossovers • Spot Sound Absorbers • How to Add a Subwoofer • The Swan IV System • Sub-Bass Power Boosting • The Unline: A Short TL • Active Filter Computer Design Program • Low-Cost Two-Way Ribbon • Amp-Speaker Interface Tester and Construction Plans • The QB_3 Vented Box is Best • A Pentagonal Box System • Keith Johnson Profile • Sheathed Conductor ESL • A Symmetrically Loaded System: Part I • Ceramic Enclosure • Inductance Measuring Technique • Polk 10 Mods •

1989 A Middle/Side Speaker • A Satellite/Woofer System • A T/S Driver Program • All About Cordless Drills • Symmetrically Loaded System, Part 2 • Adjusting Auditorium Acoustics • The Audio Laboratory Speaker System • A Passively Assisted Woofer • Digital Filter Tutorial • The Listening Arc Alignment • Small IC Power Amp & Crossover • Easy Surround Sound • Building Speaker Spikes • An Isobarik in a Thunderbird • Sheetrock Cabinetry • A Picket Speaker • Servo-Controlling AR-1 • Silent, Safe Muting System • Equalizing the Klipsch Cornwall • A Test Switcher • Visiting the Klipsch Kingdom • Rehab for Kitchen Music • Spreadsheet Design • A Subwoofer/Satellite System • Impedance Measurement as a Tool • Practical Passive Radiators • A Symmetrical Dual Transmission Line Parts 1 & 2 • The Microline • A Voice Coil Wheatstone Bridge • Tweeter Q Problems • A Dipole Subwoofer for the Quad • Adjusting Woofer Properties • Modifying Paradigm's 7se •

1990 Acceleration Feedback System • Cylindrical Symmetric Guitar TLs • Compact Integrated Electrostatic TL (Parts 1-3) • Minimus-7 Super Mod • The Show (Bass Horn) • A Small Two-Way System • Helmholtz Spreadsheet • Heresy Upon a (Klipsch) Heresy • Beer Budget Window Rattler • Contact Basics • MDT Mini-Monitor Speaker System • Titanium + TPX + Polypropylene = Fidelity • Tom Holman, Skywalker, and THX (Parts 1, 2) • Bud Box Enclosure • Klipschorn Throat Riddle • Modular Three-Way Active Speaker • CD Speaker System • SPEAKER DESIGNER Software • Symmetrical Isobarik • Novice Crossvers • Triamplified Modular System • Magnetic Crosstalk in Passive Crossvers • Mitey Mike Loudspeaker Tester • Symmetrical Loading for Auto Subwoofers • Improved Vented Box with Low Q_{TS} Drivers • BOXMODEL Woofer System Design Software • Four Eight By Twos • Dynaco A-25 Mod • Klipschorn Throat Revisited •

1991 Student Systems • Developing Servo Subwoofers • Apartment TLs • L-R Crossover for the Swan IV • More Power for Less • Guidelines for Vented Box Construction •The Pipes • MacIntosh Programs for Speaker Design • Octaline Meets D'Appolito • Small Loudspeakers for a Small Room • Creating Grilles • Measuring Drivers with Radar • Deep Bass for GMC • PSpice LF Calculations • Pipe & Ribbon Odyssey • Delac S10 • Infrared Remote Volume Control • Back-Loaded Wall Horn • Integrite • Upgrading a Bronco's Sound System • Mod for Minimus 7s • Fibrous Tangle Effects on Acoustical TLs • DOALS Party Speaker • Loudspeaker Cable • Speaker-to-Ear Interface • Quasi Electronic Crossovers • Vented Box Sensitivity •TL Evaluation • Subwoofer/satellite system • Cable & Sound Delivery • Reviews: Little V • Precise Monitor 10 •

Please send me the following back issues of *Speaker Builder*:

☐ **1991 $25**	☐ **1987 $18**	☐ **1983 $15**
☐ **1990 $25**	☐ **1986 $18**	☐ **1982 $15**
☐ **1989 $23**	☐ **1985 $18**	☐ **1981 $15**
☐ **1988 $23**	☐ **1984 $15**	

SHIPPING: BACK ISSUES
UPS: Domestic ground service by value of order:
Less than $60.00—$4.00 $131.00-$220.00—$ 8.00
$61.00-$130.00 —$6.00 $221.00 + —$10.00
Canada: Add $6 per year.
Foreign Air:
Europe: 40% of total order.
Other destinations: 50% of total order.
Foreign Surface: 20% of total order.
Rates subject to change without notice.
All remittances must be in US $ only ☐ MC ☐ VISA ☐ CK/MO

Total Magazines ________
Postage ________
Total enclosed $ ________

NAME ________ MAGIC NO ________
STREET & NO. ________
CITY ________ STATE ________ ZIP ________
MC/VISA ________ EXP ________

Speaker Builder Magazine
PO Box 494, Dept. B92,
Peterborough, NH 03458-0494 (603) 924-9464 FAX: (603) 924-9467

Answering machine for credit card orders only–before 9:00 a.m., after 4:00 p.m. and weekends.
Please have all information plus MC/VISA available.

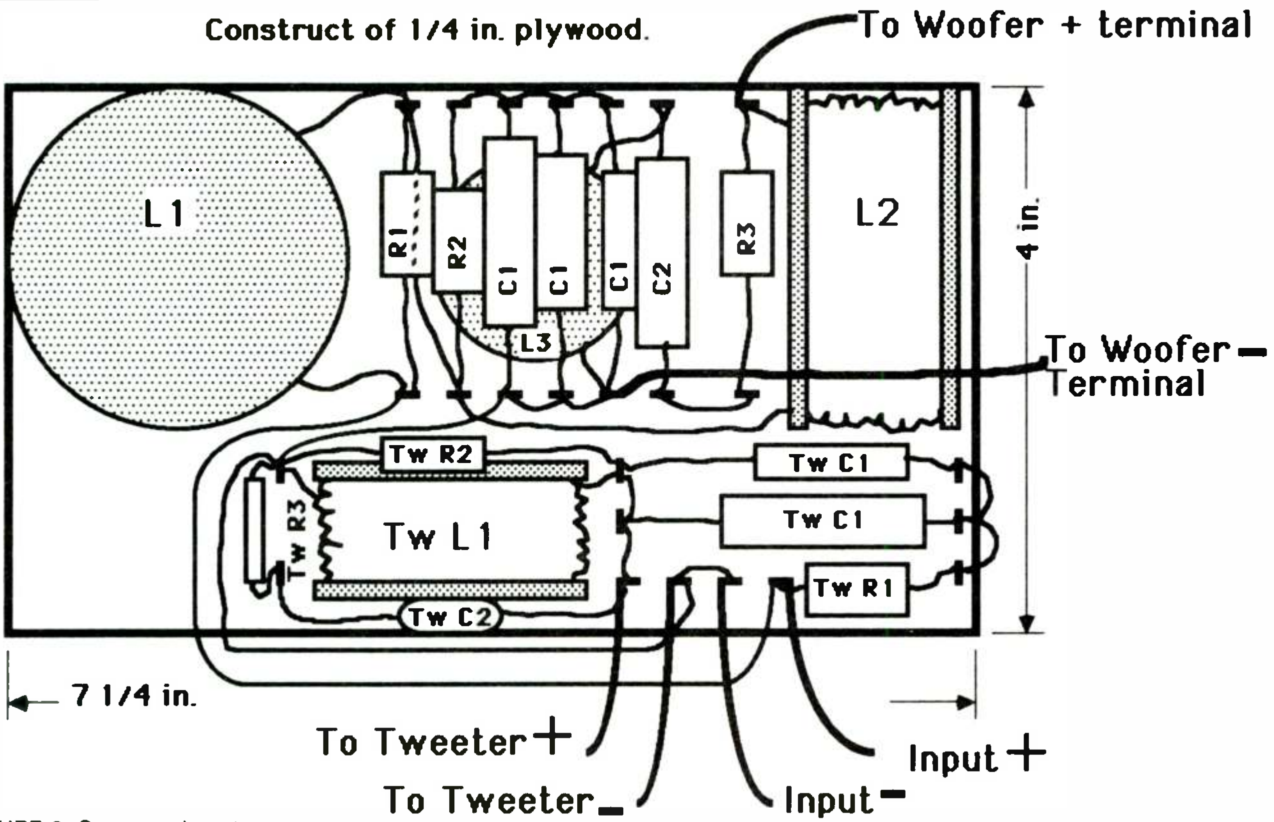


FIGURE 8: Crossover layout.

Continued from page 54

of that node are at the same potential. Number each node on the wiring diagram, and make certain that each physical node on the crossover board has all the correct parts connected to it. Also, check through the component list making sure each one is connected between the correct pair of node numbers.

If your boards pass this test, your chance of success is very high. Now mount the boards and connect them to the drivers. As a conservative startup procedure temporarily connect the boards with drivers to an amplifier and listen carefully. Make sure the woofers are receiving only lows and the tweeters are receiving only highs. After this, assemble everything and make a listening test.

ACKNOWLEDGMENTS

My gratitude to Paul R. McManus of McManus Enterprises for providing a back yard and much instrumentation with great good humor, which turned out to be a fine thing under the circumstances. I am further grateful for Paul's ear in hearing out the author's stories of backtracking and success along the way. Another person to thank is Jim Kemmish for providing the indoor measurement environment and instrumentation.

Roger Sutor is thanked for reading the very first version of this article and making critical comments about it. Finally, I must thank those of my friends who wanted to build a pair of these speakers for being patient while I finished my thesis and then wrote this article.

When mounting the woofer, a good technique is to apply 1'4" × ⅛" foam sealing tape to the rear of its mounting flange, and to insert small rubber grommets into the frame mounting holes. The driver can be mounted with small roundhead wood screws with smooth shanks that just fit into the grommet holes, whose heads are slightly larger than the flange hole. This floats the mounted woofer on rubber, and it needn't be tightened down very hard to work. This minimizes vibrations conducted into the cabinet.

The tweeter mounting is less critical as long as it seals, so I used foam tape behind each and put one half a rubber grommet under each screw head.

Theories abound concerning how to support speakers on their stands. I just used low rubber stick-on chassis feet on the bottom of his enclosures, positioned near the corners to avoid the possibly flexing bottom panel. This location minimizes transmission of vibration from the enclosure to its mounting surface. Individual builders will doubtless try their own experiments with various supports.

These speakers have been played sitting on chairs and on tripod stands. The chairs introduce audible coloration even if they are of open-back design. Stand-mounting the speakers will produce impressively neutral sound, although they aren't that bad on chairs, either. Furthermore, the adage that one will hear all amplifier improvements through good little speakers seems to be true in this case, as well. You will be rewarded if your amplifier is in good shape.

FINAL WORDS. I am unable to provide copies of his optimization software to other users, so you needn't waste ink asking for one. More detailed questions about the optimization procedure will be answered in these pages, but be ready to get escorted past minima in higher dimensional spaces. Not all of the tricks for arriving at this speaker's design will be divulged, but anyone may build a copy—for no profit. I wish prospective builders and listeners as much pleasure from this design as he has derived for himself.

SpeakerBuilder
THE LOUDSPEAKER JOURNAL

Subscriber Services

WOULD YOU LIKE TO . . .

Complete the applicable section(s) below and mail to:
Speaker Builder
PO Box 494
Peterborough, NH
03458-0494

❑ SUBSCRIBE?

YES! Just check the appropriate boxes in the "renew" section and fill in name and address.

❑ RENEW?

Fill in the appropriate sections below. For faster service use your mailing label and magic number.

❑ 1 year at $25
❑ 2 years at $45

❑ Check/MO enclosed
❑ Charge MC/VISA

Canada add $6 per year postage.
Remit in US $ only
Overseas rates on request.

☐☐☐☐☐☐
MAGIC NUMBER

NAME ____________________

STREET & NO ____________________

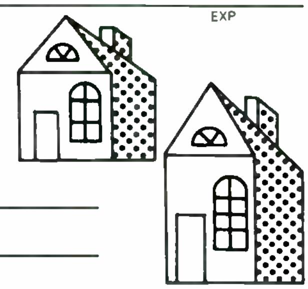
CITY ____________ ST ____ ZIP ____

MC/VISA ____________________ EXP ____

❑ CHANGE YOUR ADDRESS?

Fill in your new address to the right and old address and magic number above.

MY NEW ADDRESS:



NAME ____________________

STREET & NO ____________________

CITY ____________ ST ____ ZIP ____

❑ GIVE A GIFT?

What a thoughtful idea! Fill out your name and address above, the recipient's name and address to the right. We'll even send a gift card announcing your gift.

NAME ____________________

STREET & NO ____________________

CITY ____________ ST ____ ZIP ____

❑ TELL YOUR SPEAKER BUILDING FRIENDS ABOUT US?

Print their names and addresses below.
We'll take care of the rest.

NAME ____________________

STREET & NO ____________________

CITY ____________ ST ____ ZIP ____

NAME ____________________

STREET & NO ____________________

CITY ____________ ST ____ ZIP ____

YOU CAN CALL US
(603) 924-9464
Subscription Department
9-4 Business Days
Answering Machine
All Other Hours
FAX: (603) 924-9467
24 Hours

❑ COMPLAIN ABOUT YOUR SUBSCRIPTION?

Yes, it sometimes happens. We try to avoid it at all costs, but if you have a problem, we want to know. Call or write with all pertinent information.

Software Report

Quick Box

Reviewed by G. R. Koonce

Quick Box by **Sitting Duck Software,** available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467. $34.95 plus $2 S/H.

Quick Box is a software program for the IBM PC and compatibles. It allows the user to design speaker enclosures for non-equalized closed-box, vented-box, and one form of bandpass. Quick Box includes a library of speaker driver data that is expandable to 200 entries and utilities. One utility supports the Hercules graphics software.

This program is the work of Bill Fitzpatrick, the programmer at Sitting Duck Software.

Requirements

Quick Box is designed for any IBM-compatible computer running DOS 2.11 or later, 384K RAM, and Hercules, CGA, EGA or VGA graphics card, and supports the H-P LaserJet as well as dot-matrix printers. A math coprocessor is not required to run the program.

All information in this review is based on running Quick Box on an AT-compatible with an EGA color graphics card. The graphs were generated on a printer emulating an IBM Proprinter.

Manual

The program manual is concise. It should be noted that it does not cover the topic of speaker enclosure design, since it assumes the user understands speaker terminology, such as Q_{TC}, V_{AS}, and other like terms.

The manual includes references to informational sources on enclosure design. Another feature is plenty of context-sensitive help screens, reducing the need to reference the manual. With a little practice, program operation becomes quick and intuitive.

Program Setup

No special installation is required; Quick Box can be run from a back-up working copy, made from the supplied program diskette. Hard disk installation is also available and will allow the program to run faster.

Three utilities are supplied that should be considered before starting:

QBHERC: If you have a Hercules graphics card, you must run this utility first. It is located in the root directory.

TURBOKEY: Allows control of the keyboard response time on AT class computers. See TURBOKEY.DOC file (in the UTILITY subdirectory) for instructions.

QUICKEYS: Allows speeding up the keyboard on XT class computers. See QUICKEYS.DOC (located in the UTILITY subdirectory) file for instructions.

If you are running the program from a floppy diskette, do not write-protect the diskette. Start Quick Box by typing: **QBOX <Enter>**.

I recommend you immediately configure the program to your printer by selecting the **Print** option from the Main Menu. This will write the file SD-QBOX.INI to the disk, so when you start Quick Box the printer will be automatically configured.

Menu Selection

Once the program is started and the printer configured, you must specify a speaker driver before beginning enclosure design. From the Main Menu select either **Driver** (discussed later), or **LibMgr**. The library manager provides speaker driver data for use in enclosure design. Note, this is catalog data and may not accurately represent the characteristics of the driver you purchase.

To load a specific driver's data into the program, highlight your choice and press the **<Enter>** key. This will select your driver and return you to the main menu. Once you have made a selection, the **<F2>** key will **Preview** the data (*Fig. 1*).

The Main Menu screen provides useful information about your present status. It displays the name of any loaded driver, current vented-box (VB) and closed-box (CB) status, number of drivers in the library, and the printer setup status. Pressing **<Esc>** returns you to the Main Menu from other screens. *Table I* shows the assignments of the function keys when doing enclosure design.

Dimension Ratios

With the driver data loaded, select either design type: closed-box (CB) or vented-box (VB). Selecting **Closed** will result in a screen similar to *Fig. 2*. The initial design is for a CB with a Q_{TC} of about 0.7. You can change this design using selections found in box volume.

The CB screen provides system and driver information and identifies the driver. The values for the impedance-corrector network are shown, to cancel the single impedance peak of a CB system.

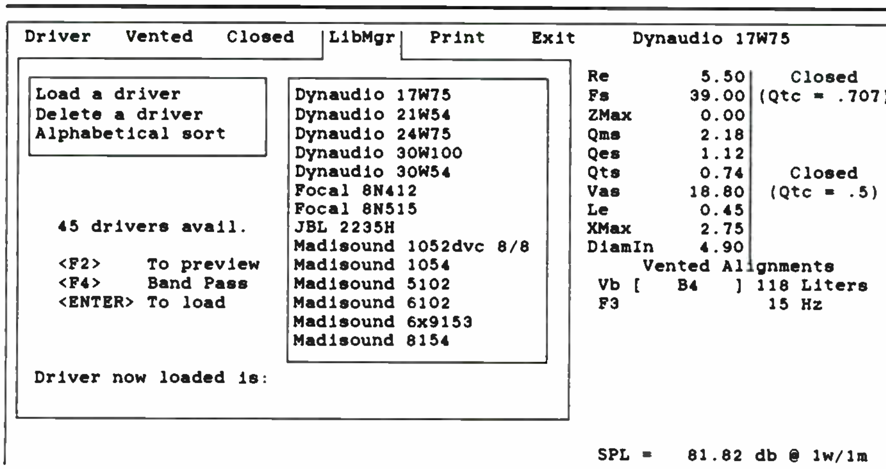


FIGURE 1: Library Manager Preview of Dynaudio 17W75 driver.

This will compute and display the internal dimensions of enclosures with various amounts of over volume. The estimated driver volume has been included in these dimensions and the dimension ratios have been selected to spread out the standing-wave frequencies within the enclosure.

```
Driver   Vented   |Closed|  LibMgr   Print   Exit                   <F1> Help
                   ________________________________________
                  |         Closed Box Design              |
                  |                                        |
                  |             Focal 8N515                |
                  |                                        |
                  |   Vb     <F10>            16.0 liter   |  Alpha      6.88
                  |                            0.6 cu.ft   |  Fs        27.3   Hz
                  |   Rg     <F6>              0.2 ohms    |  Fc        76.6   Hz
                  |                                        |  F3        75.4   Hz
                  |                                        |  Qts        0.26
                  |                                        |  Qtc        0.72
                  |                                        |
                  |  <F2> Do Plot        <F6> Change Rg    |  ηo         0.80 %
                  |  <F4> Plot   ON                        |  SPL       91.03 db
                  |  <F10> Change Vb     <ESC> Main menu   |             @ 1w/1m
                  |________________________________________|

 ____[ Resonance Impedance EQ ]_____   _________________<F9> Inside Dimensions__
|      Ceq   =   391.98 mfd         | |   0% over volume      8.3   10.4   13.1  |
|      Leq   =     1.50 mh          | |   5% over volume      8.4   10.5   13.3  |
|      Req   =     9.36 ohms        | |  10% over volume      8.5   10.7   13.5  |
|___________________________________| |_________________________________________|
```

FIGURE 2: Typical closed-box design screen display.

```
Driver  |Vented|  Closed   LibMgr   Print   Exit                 <F1> Help
         _____________________________________    _<F9> Inside Dimensions__
        |        Vented Box Design            |  | 0%ov     9.8   12.2   15.5 |
        |     Standard B4 Alignment           |  | 5%ov     9.9   12.4   15.7 |
        |          Focal 8N515                |  |10%ov    10.1   12.6   16.0 |
        |                                     |  |_____________________________|
        | Vb     <F10>              28 liter -|----- Box volume   0.99 cu ft
        | Fb     <F8>               42 Hz    -|----- Box frequency
        | Rg     <F6>              0.2 ohms  -|----- Added resistance
        |                                     |   _____________________________
        | Vent:   Diam  2.00     Length  3.32 |  |    Alignment options        |
        |         Vol   10 cu.in.             |  |    <1>   Standard  B4       |
        |                                     |  |    <2>   SBB4   or  BB4     |
        | <F2> = Do Plot      <F8>  = Change Fb |  |    <3>   QB3    or  SQB3    |
        | <F4> = Plot  ON     <F10> = Change Vb |  |    <4>   SC4    or  C4      |
        | <F6> = Change Rg    <ESC> = Main Menu |  |  Selection resets Vb & Fb  |
        |                                     |  |                             |
        | Use  + or -  to change vent diameter. |  |      Select a number        |
        |_____________________________________|  |_____________________________|
                                                   ηo      0.80 %
                                                   SPL    91.03 db @ 1w/1m

              All calculations based on losses (QL=7).
```

FIGURE 3: Typical vented-box design screen display.

Vent Volume

You can return to the Main Menu, to get to **Vented**, or simply jump directly to VB design by pressing the <V> key. The screen will look somewhat like *Fig. 3*. Again, function key assignments are displayed in *Table I*.

Note that if you had entered an R_G value in the CB screen it is not carried over to the VB screen; you must re-enter R_G if desired. With VB design, you can choose to select one of the various alignments offered (the number offered varies with the driver) or create a modified design using entry of V_B or F_B. If you do enter a V_B or F_B value, the VB screen is marked as modified so that you know it is not a standard alignment; a good idea.

All of the VB designs are for a box Q of 7. On the VB screen, the driver is identified, system information is provided, and inside dimensions for various enclosure over volumes are provided. While not included in the indicated V_B, both estimated driver volume and vent volume have been added to the inside dimension data. I looked at the allowance for the driver on a few examples and it ranged from 2%-9%. Vent duct design information is provided along with the ability to change the vent duct diameter.

Note that the indicated vent volume is for the total vent and the portion of the vent duct in the front panel has not been subtracted. This is generally of little importance unless the vent volume is rather large and the front panel portion is a major part of that volume.

A Handy Feature

While you are in either CB or VB design screens, if plot is **ON**, toggling key **<F4>** then **<F2>** will produce a plot of the small-signal frequency response of the system. This plotted graph will contain either a VB design, a CB design, or both, depending on the plot toggle status that is set on each design screen. In *Figs. 2* and *3*, plot is **ON**, so pressing **<F2>** from either screen will produce a display similar to *Fig. 4* showing both systems.

The display was somewhat different on my screen in that only the +3dB, 0dB, and −3dB horizontal lines were shown and no vertical frequency lines were visible. When Sitting Duck told me this should not occur, I was able to get the missing lines to appear by adjusting my monitor. The lines are rather dim, so be on guard for a similar problem.

The internal print routine will print an output similar to *Fig. 4*, except the options along the bottom are omitted. The driver is identified, the date is given, and the required system data is shown, so that the plot represents all you need to print a complete design record.

I do thank Sitting Duck Software for including the ability to send a form feed to the printer from the graph. This may seem like a trivial issue but I have a buffer between the computer and the printer which can hold several pages—making it mandatory that I be able to inject form feeds into the data stream and not be required to manually advance the paper.

TABLE 1

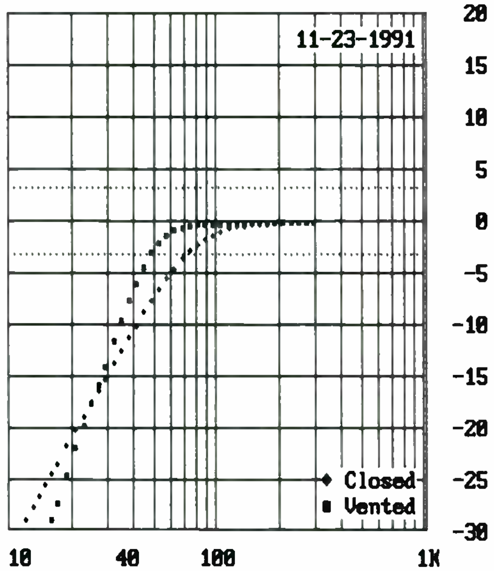
CB AND VB DESIGN SCREEN FUNCTION KEY COMMANDS

Key	VB Screen	CB Screen
<F1>	Help	Help
<F2>	Does Plot	Does Plot
<F4>	Toggles VB plot ON/OFF	Toggles CB plot ON/OFF
<F6>	Enter/Change R_G	Enter/Change R_G
<F8>	Change F_B	
<F9>	Toggle inside dimensions between 0, 5, & 10% and 15, 20, & 25% over volume	
<F10>	Change V_B	Change V_B

Diagonal Driver

Fourth-order bandpass (BP) enclosure design is handled a little differently. Note that these enclosures are of the type covered by Jean Margerand in "The Third Dimension: Symmetrically Loaded," Parts I and II (*SB* 6/88, p. 29 and 1/89, p. 27), where the rear volume is sealed and the front volume vented.

To design a BP enclosure, you must enter the driver into the library. Select **LibMgr** from the main menu and then choose your driver. Pressing **<F4>** will provide a chart of design information, for




```
            Focal 8N515
      —— Vented Box Design ——
       Standard B4 Alignment
Vent diameter        2.0  inches
Vent length          3.3  inches
Vent volume         10.0  cu.in.
Vb                  28.1  Liters
                     1.0  Cu.Ft.
                  1714.9  Cu.In.
Fb                  42.2  Hz
Rg                   0.2  ohms
      —— Closed Box Design ——
Alpha               6.88
Fs                  27.3 Hz
Fc                  76.6 Hz
F3                  75.4 Hz
Qts/Qtc             0.26/ 0.72
Rg                   0.2 ohms
Vb                  16.0 Liters
                     0.6 Cu.Ft.
                   976.4 Cu.In.
<F3> Print          <F> FormFeed          <ESC> Main Menu
```

FIGURE 4: Quick Box plot screen—menu options not shown on screen, see text.

a BP design with 0dB ripple, and a menu of options allowing various changes in the design—such as using two or four drivers in various combinations.

Follow the excellent Help screen instructions, available with the BP screen, to aid selecting the options and then display the information you need. No provisions are available for plotting the frequency response of a BP system. There is, however, the ability to design a cubic enclosure with a diagonal divider to form the front and rear volumes.

Displays

To enter a new driver for design, select **Driver** from the Main Menu. You may also modify the data for a library driver, after it is loaded, using the driver screen. Remember, the disk from which you are running must not be write-protected if you plan to save driver data to the library.

The driver entry screen requires the following data entries:

- Driver name—to identify the driver in the library and on the output screens.
- R_E—driver voice coil resistance measured in ohms.
- F_S—driver resonant frequency in Hz.
- Q_{MS}—driver mechanical Q at F_S.
- Q_{ES}—driver electrical Q at F_S.
- V_{AS}—driver equivalent compliance entered in either liters or cubic feet.
- L_E—driver voice coil inductance in mH. You can bypass this entry field with the arrow keys, but then you won't be able to save the data file to the library.
- Diam—driver piston diameter measured in inches.

You may enter two optional pieces of information:

- Z_{MAX}—the maximum resistance displayed by the driver at F_S in ohms.
- X_{MAX}—the maximum one-way linear voice coil excursion in millimeters.

Once you enter this data the driver screen will display the following computed information:

- Q_{TS}—the driver total Q at F_S.
- SPL—the driver efficiency in dB at 1W input @ 1M.
- Zobel—the series resistance and capacitance to put across the driver in order to *flatten* its impedance at higher frequencies.
- EB Product—the efficiency bandwidth product (EBP). This helps to indicate the driver suitability to VB and CB

TABLE 2

COMPARISON OF QUICK BOX WITH RESULTS BY CONVENTIONAL TECHNIQUES

Focal 8N515 driver with added 0.2Ω R_G—driver Q_{TS} = 0.256

Closed-Box Design: Quick Box V_B	Q_{TC}	F_3	Conventional Q_{TC}	F_3	Vented-Box Design: Align	Quick Box V_B	F_B	F_3	Conventional V_B	F_B	F_3
0.581	0.71	75.4	0.71	75.5	QB_3	0.848	42.7	52	0.938	41.1	51.3
0.408	0.83	77.3	0.83	77.4	SBB_4	1.050	27.3	62	1.14	27.3	62.5
0.277	1.00	83.8	1.00	84.0	SC_4	—	—	—	1.06	28.8	60.8
					B_4	0.992	42.2	49	1.65	43.2	43.2

Audio Concepts AC7-8 driver with added 0.2Ω R_G—driver Q_{TS} = 0.433

Closed-Box Design: Quick Box V_B	Q_{TC}	F_3	Conventional Q_{TC}	F_3	Vented-Box Design: Align	Quick Box V_B	F_B	F_3	Conventional V_B	F_B	F_3
0.976	0.71	65.0	0.71	65.7	SQBC	2.03	38.0	35	2.28	37.2	34.5
0.617	0.83	66.7	0.83	67.0	BB_4	1.38	40.0	41	1.46	40.0	40.6
0.381	1.00	72.4	1.00	72.7	C_4	2.00	37.7	35	2.20	36.8	34.6
					B_4	1.93	38.0	35	1.78	37.4	37.4

Dynaudio 17W75 driver with added 0.2Ω R_G—driver Q_{TS} = 0.757

Closed-Box Design: Quick Box V_B	Q_{TC}	F_3	Conventional Q_{TC}	F_3	Vented-Box Design: Align	Quick Box V_B	F_B	F_3	Conventional V_B	F_B	F_3
0.896	1.01	40.2	1.0	40.5	C_4	—	—	—	4.06	22.7	19.6
					B_4	3.95	22.0	19	—	—	—

design, along with recommendations based on the EBP value.

- S_D—the effective cone area in square meters.
- V_D—if X_{MAX} was entered then volume displacement in cubic meters is also displayed.

Driver Modifcation

Any driver data, new or modified, can be entered into the driver library. Pressing <F2> will add your newly defined driver or pressing <F4> will overwrite the driver's library file. A good practice is to use unique driver names for the library file, this will prevent confusion with duplicate file names. I put two drivers in the library with the same file name and successfully deleted one without problem, but avoid this since you could delete the wrong driver. Once deleted, the file cannot be recovered.

You can exit the driver screen after entering data for a new driver, when you first start the program, but you will not be able to design enclosures. Upon initial entry of a new driver, you must add it to the library before CB or VB design is permitted. If you wish to omit it later you can.

While the use of multiple drivers was supported with the BP design, it is not offered for CB and VB design. This can be accomplished by defining or modifying the driver data to represent multiple drivers on the driver screen.

For assistance in handling this modification, refer to Contributing Editor Robert Bullock's reply to David Del Zotto (*SB* 2/87, p. 60) or my response to Herbert Meyers' query (*SB* 3/90, p. 3).

With practice, using the various menus and screens, it becomes easy to accomplish the design task you have in mind. On my computer all functions were handled quickly; I had only to wait for plotting or printing a frequency response.

Nitpicks

To have a software program do exactly what you wish, you'll have to write it yourself. What follows are things I would have done differently. That does not mean that Quick Box should be changed. I list these and their consequences for your consideration and indicate how I worked around them, when necessary.

1. Driver data entry—sometimes you are trying to evaluate a driver shown in a catalog with a minimum of data given. With the required data entries on the driver screen this causes a problem:

A. R_E—If you do not know R_E, then make one up. You should not attempt to add R_G to VB or CB designs or use the Zobel or Resonance Impedance EQ values shown if you made up R_E.

B. Q_{ES} and Q_{MS}—If you are given only Q_{TS} for the driver, then enter 1.1 × Q_{TS} for Q_{ES} and 11 × Q_{TS} for Q_{MS}. The computed Q_{TS} should agree with your starting value. Again, do not expect accurate results with added R_G if you did not enter the actual Q_{ES}. Also, efficiency, SPL, EBP, and Resonance Impedance EQ values will not be correct.

C. L_E—You may not know the driver voice coil inductance so you'll have to make up a value; something like 0.5-1MHz is probably workable. Do not use the computed Zobel values if you estimate L_E.

D. Diam—If the true piston diameter is not known, put in a value about 80% of the nominal driver diameter. Note that the computed S_D and V_D values will not be accurate.

E. V_{AS}—While V_{AS} may be entered as cubic feet or as liters, the number of data places is limited. When you must affix F to the end of cubic feet entries (and for small drivers) you may want to externally convert to liters and enter in liters. Conversion formulas are: liters times 0.035285 equals cubic feet, cubic feet times 28.34 equals liters.

F. Clearing driver data—If a driver has been loaded into the driver screen there is no way to clear the screen

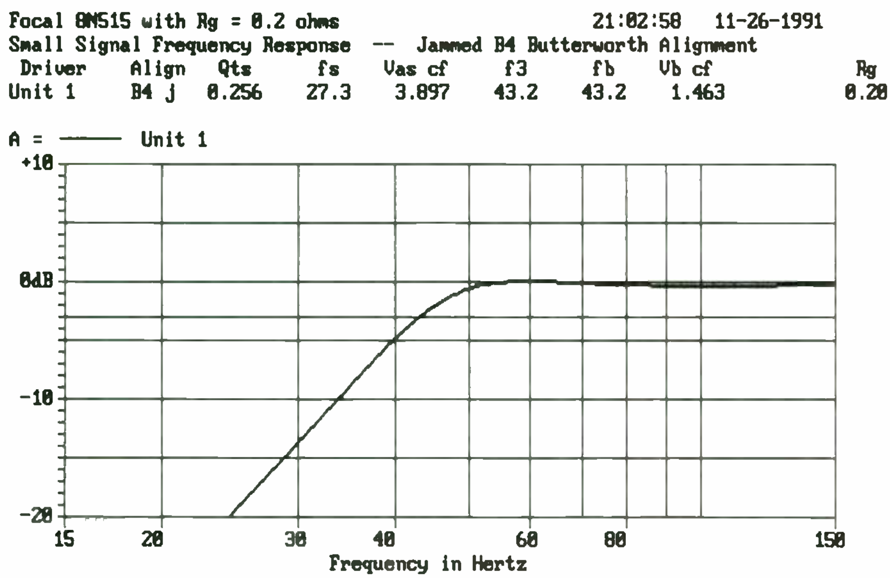


FIGURE 5: Focal 8N515 driver B_4 response by alignment jamming—not produced by Quick Box.

for new data. You simply overwrite the existing data being careful to cover all the items.

G. Driver name—Be careful when entering data and modifying the driver's file name. You cannot tell for sure if the change was made, because the color of the data field is the same before and after entry.

H. Need to add to library—When you first start Quick Box and supply data for a driver, you cannot begin designing with the driver until it has been added to the library. I found this inconvenient.

2. Duct diameter changes—For VB designs the vent duct diameter is made larger by the + key, which is a shifted key [not the gray "+" key—Ed.]. This may not cause you a problem, but I somehow fail to adjust quickly to this.

3. Volume changes—While you can enter V_{AS} as liters or cubic feet, on the VB or CB screens, you must change or enter V_B in liters. I prefer cubic feet to liters. The programs I write allow data entry in either liters or cubic feet, but thereafter insist on cubic feet. It is impractical to always allow entry using both dimensions, but you will have to think in liters. Also, as shown in *Fig. 2*, the V_B, displayed in cubic feet, provides one decimal place for small enclosures. You should work from the liter number, or go to the plot function <F4> and use the cubic inch number, to get greater accuracy for small enclosures.

4. CB design—The CB screen will not permit designs using an entered Q_{TC}, which is how I like to work. You must change V_B, in a trial-and-error fashion, to achieve your desired Q_{TC}.

5. Printing CV and VB design screens—The layout of the CB and VB design screens are excellent for screen viewing. If you want to print them (using **Shift PrtScn**), you will get menu lists and boxes that do not add information.

6. Printing driver data—To output the data for a single driver you must print the driver screen or, with the driver in the library, the preview screen. Again these are excellent screens for display viewing but they use printer ribbon unnecessarily. I recommend printing the entire driver library using the option provided with the Print screen.

Performance Items

Quick Box is a well-designed and fast program. When comparing its speaker designing features with other software packages, it is a strong, capable performer.

I picked three drivers from the library and compared the performance of Quick Box with my present design approach. Results are shown in *Table II*. The expected slight differences in results are caused by rounding errors and reading data from curves; the only major difference is the B_4 alignment I discuss below.

Some items are presented for consideration, on which I have a slightly differing personal opinion. Performance and capability of the BP design was not reviewed because I do not have experience in this area. The following are comments about the quality of information generated by Quick Box:

1. Zobels—The presented Zobel components should be taken as a guide and not used without testing. My views on

why this is so are documented in my article, "Crossovers for the Novice" (*SB* 5/90, p. 26).

2. The recommended EBP limits for driver use in VB or CB designs seem to be: CB only for EBP < 50; VB only for EBP > 100; and both VB and CB suggested for EBP 50 to 100. Not everyone will read EBP in quite the same way; for my opinion on this topic read "Trade-Offs in Closed-Box Alignment" (*SB* 2/84, p. 26) and see Reference 1 for the original work by Small.

3. CB Resonance Impedance corrector—The CB design screen presents values, for the series R-L-C conjugate network, to be placed across the CB system and result in a constant input resistance in the frequency range of the system resonance. I tried a couple of library drivers, with code I had written some time ago. The resistor and capacitor values agreed, but the inductor value did not. I re-derived the impedance conjugate network equations and solved one case by hand. The result agreed with my code. This, coupled with the fact that L_{EQ} and C_{EQ}, developed by Quick Box, do not produce the required resonance of F_C, indicates there is some problem in the L_{EQ} computation. In addition, I found that if you modify the driver data affecting these components (R_E or Q'_S), then exit and go to CB design, only the R_{EQ} value has changed. C_{EQ} and L_{EQ} should change, but do not. This is true even if you update the driver data file in the library before doing the design. I believe the Resonance Impedance EQ function of Quick Box is not presently working correctly and should be revised.

4. B_4 Alignment—Each VB design screen starts with an alignment designated as a Standard B_4. I am not quite sure what this means. Normally the B_4 alignment for a box Q of 7 only occurs if the driver Q_{TS} is about 0.4. A characteristic of the natural B_4 alignment is that $F_3 = F_S = F_B$. It is possible to force drivers with a lower Q_{TS} into a B_4 alignment using Complex Alignment Jamming. Such jammed B_4 alignments show a slight amount of passband ripple, but do meet the requirement of $F_3 = F_B$. *Figure 4* shows the Standard B_4 alignment curve developed by Quick Box for the Focal 8N515 driver, used for the tabulated results in *Table II*. Note that F_3 does not equal F_B. *Figure 5* shows a jammed B_4 alignment curve for this same driver and F_3 does equal F_B with a passband ripple of less than 1dB. The B_4 jamming equations produce strange curves for driver Q_{TS} above 0.4 and are not useful. I question what the Standard B_4 alignment, produced by Quick Box, really accomplishes. Perhaps Sitting Duck Software will supply this information.

Recommendation

Quick Box is a fast, capable design program for CB, VB, and one class of BP enclosure designs where electronic equalization is not employed. The program allows for user driver data entry, along with a driver data library which the user can expand to 200 entries. Provision is included to graph VB and CB small signal frequency responses and to print these graphs. User driver data entry requires information that the user may not have and *creating* some data values may be required. A variety of useful information is provided with each design screen.

I cannot explain what the Standard B_4 Alignment, provided for VB designs, really represents. Plus, the code generating the Resonance Impedance EQ values, developed for CB designs, needs to be revised. Overall, Quick Box consists of a single disk and a manual that provides a lot of capability for the price. This

Continued on page 86

REFERENCE

1. Small, R.H., "Suitability of Low-Frequency Drivers for Horn-Loaded Loudspeaker Systems," Audio Engineering Society Preprint No. 1251 (K-8).

Tools, Tips & Techniques

SLEUTHING "Q"

Most speaker-design books tell you that $Q_{TS} = (Q_{ES} \times Q_{MS})/(Q_{ES} + Q_{MS})$, but none that I have seen tell you how to find Q_{MS}, or Q_{ES} if you already know Q_{TS} and one of the other Q parameters. For any of you who know two of the three Q values for your driver, here are the formulas to find the third Q value.

To find Q_{TS}:
$Q_{TS} = (Q_{ES} \times Q_{MS}) / (Q_{ES} + Q_{MS})$

To find Q_{ES}:
$Q_{ES} = (Q_{MS} \times Q_{TS}) / (Q_{MS} - Q_{TS})$

To find Q_{MS}:
$Q_{MS} = (Q_{ES} \times Q_{TS}) / (Q_{ES} - Q_{TS})$

Gregg S. Irwin
Whittier, CA 90601

SUCCESSFUL CIRCUIT

In Bruce Edgar's interview of Ken Kantor (*SB* 5/91, p. 22), he asked: "What is a precision current source?"

A broad reply to this question could interest *SB* readers. *Figure 1* shows a circuit I've been using with great success when measuring loudspeaker parameters.

Any voltage signal (DC or AC) fed to the input jack will be translated into an output current of the same waveform according to the simple relation i = e/10 where 10 is the value of the precision output resistor (which you may change, if necessary) and e is the instantaneous voltage the 50k trimpot delivers to pin 5.

Applying a sinusoidal voltage source to the input and adjusting e to 0.5V RMS, you will get the 50mA AC constant current. I use this to drive my speakers under test, wiring them to the "hot" output lead and ground.

Check the circuit with an RMS meter or read the voltage drop across the output resistor. In both cases, readings should keep constant even with output loads ranging from 0–250Ω.

Use only metal film resistors (⅛W) within ±1% or better, except for 4k7 where ±10% suffices.

Homero Sette Silva
Boa Vista—RR, Brazil 69300

QUASI-12 CROSSOVER NETWORK

In "Tools, Tips & Techniques" (*SB* 3/91, p. 68), Jerry Stump writes about the Ashley and Kaminsky AES article on crossovers.[1] Like Mr. Stump, this article caught my eye so I tried some of these techniques.

To understand this crossover, we must recognize that these are "Quasi-12dB/octave" designs. They differ from "true" 12dB/octave designs in significant ways. This difference is crucial to anyone attempting to use this design. Without taking these factors into account, you may or may *not* get an electrical 12dB slope.

The way these crossovers create a 12dB slope (as best I can tell) is by precisely "peaking" the Q of the LC network with a slight peak created just before the roll-off. Without it, the crossover is a classic 6dB/octave network. Thus the "Quasi-12" designation.

At first this relationship was not apparent to me. I had tried these crossovers in several speakers with varying degrees of success. They simply didn't "sound the same," speaker-to-speaker, or design-to-design. They are immediately attractive, since the series networks all absolutely guarantee "correct" placement of the high- and low-frequency slopes, regardless of the values you choose. So, it's virtually impossible to leave a sonic "hole" between two drivers, or to have them overlap (as with parallel networks). My problem was that even though I had followed design rules in the article, the sonic results did not seem consistent.

To make a long story short, I looked at the frequency response of the crossovers by measuring the crossover *and* the drivers together. I used an oscilloscope and sweep generator in the same network, replaced one or both of the drivers with noninductive resistors, and compared the results. It turned out the crossover behaved differently dependng on the load presented. It was a true crossover chameleon.

To make it work properly, I found it

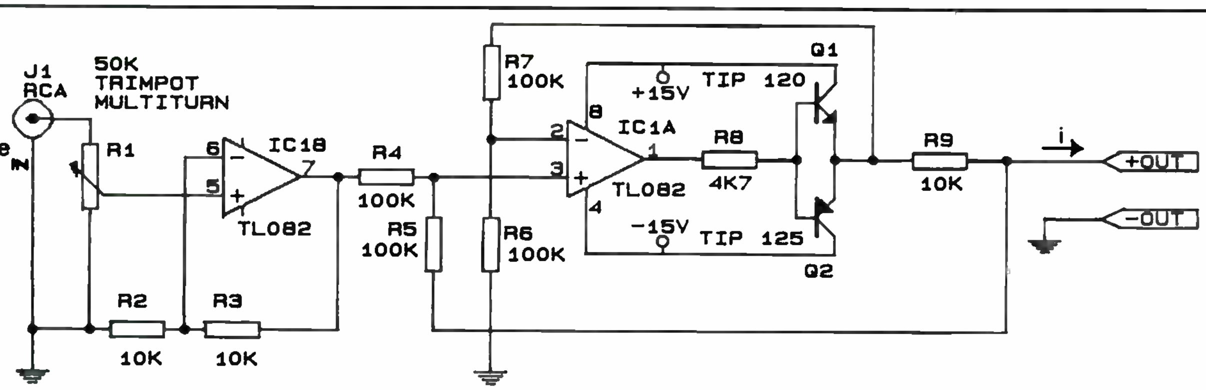


FIGURE 1: Silva's current source device.

necessary to run the sweep with the drivers in circuit (and sometimes in the cabinet), and then beginning with a slightly (50%) smaller cap than the chart (or formula) called for, I added caps until the response "bump" was exhibited on the 'scope. Typically, the bump for each slope was about 0.2dB. When it appeared, the slope of the rolloff changed dramatically. Without the frequency response "bump," the crossover is a 6dB/octave type. With the bump, it's 12dB/octave.

To use these crossovers effectively, it was essential to look at each *individually*, using a sweep generator and a 'scope. Without doing this test, it seemed impossible to control the true response of the network, or to know whether it was 6 or 12dB/octave. Also, the bump would be difficult to detect without the use of both instruments.

Because each driver has slightly different parameters, you cannot build the crossover with Quasi-12dB networks by counting on any driver-to-driver, model-to-model, or brand-to-brand consistency. Each network must be built and tweaked for individual sets of drivers.

Given your chosen value for L, your final crossover frequency will end up somewhere near the design value (usually within the octave), but not necessarily where you intended, or where the chart plots it. Surprisingly slight variations due to the actual impedance of the driver will cause the frequency to change, as will the trimmed value of C required to create the Quasi-12dB bump. The impedance of the drivers should be identical. If it is important to have an "exact" crossover center frequency, then *always* use a larger L than you calculated, and be prepared to spend lots of time alternating between trimming wire from the inductor and adding those caps to find the bump, while watching the 'scope.

These crossovers do not sum "flat." Also, as with any 12dB/octave slope, a 180° phase shift occurs between the high- and low-pass. Even so, they might be closer to being flat, and therefore a better choice than guesswork or an approximated parallel network. This may not be a significant factor, but should be noted.

With this crossover, you can't use a simple series resistor, "pot," or rheostat to set driver "level." The driver impedance *must* be preserved or the crossover frequency will radically change. Always use a voltage divider made up of two resistors and the driver. The total resistance, including the driver, should be as close as possible to the driver alone. This allows you freedom to do a few tricks, such as using two drivers with different impedances, like a 4Ω tweeter with an 8Ω woofer. (Tweeters almost always require a reduction in level to match the woofer.) You'll need a few resistors in parallel to get correct levels and preserve the impedance.

This crossover format is among the most sensitive to individual drivers I have used. While it does offer simplicity and low parts count, it can be tricky to align properly. Much of the above process (and more) is required for any type of crossover network to yield accurate results. Since I first experimented with the Quasi-12 design, my preference has shifted to higher (24dB) or lower (6dB) order filters, and usually parallel networks.

Randall Bradley
Hannacroix, NY 12087

REFERENCE

1. Ashley, J. Robert and Allan L. Kaminsky, "Active and Passive Filters as Loudspeaker Crossover Networks," 39th Convention of the Audio Engineering Society, New York, October 12, 1970.

CONVERSION FORMULA

Frequently when I read specs from speaker catalogs, I need to convert SPL (dB) to percentual efficiency n_O, and vice-versa. Because using a hand calculator is boring, I developed a graphic method to accomplish this task.

Using the scales depicted in *Fig. 1* is easy: find the range of interest, enter the known variable in the proper scale, and read the corresponding value of the unknown quantity.

For example, suppose you want the SPL level corresponding to a 2% efficiency. By hand, you can use equation (1) to get: 112 + 10log0.02 = 112 − 17 = 95dB.

$$SPL_{(dB)} = 112 + 10\log[n_{(\%)}/100] \quad (1)$$

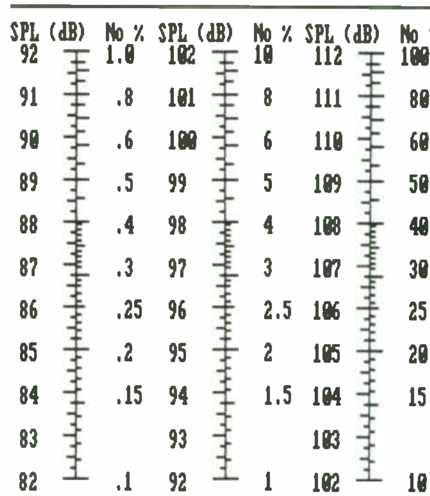


FIGURE 1: Graphic conversion between SPL (dB) and reference efficiency (%).

Using *Fig. 1*, all you do is read the corresponding value to 2% (central area) and promptly get the answer: 95dB.

Homero Sette Silva
Boa Vista—RR, Brazil 69300

TV CART SUBS

I use a method for building subwoofers I think you will find cheap and easy, but yields wonderful results.

Start with a TV cart: the sort you can find at discount stores like Costco, Target, or KMart. They are made of particleboard and fake wood veneer, and are usually just one item in a line of similar products. Brand names vary, and they usually come in different sizes. I paid only $25 for one at Costco, years ago. Anyway they are usually cheap.

They come with screws and casters, and are easy to assemble, but we can ignore the directions, since we want to add a front and back panel of thick plywood to make a complete box. Measure the openings carefully, being sure to account for the overhang of the trim pieces along the top. Your plywood panels should be recessed so that the outside top edge of the panel can be glued to the side of the trim pieces that face inward. Make two identical panels, making sure they are perfectly square. Next make holes for your woofer and tuned port in one of them. Remove the veneer where the new panels will be glued to the cabinet. Wood glue will work fine, but Liquid Nails is better. It is very strong, sets fast, and fills gaps for an airtight joint. Construct the box from the bottom up, keeping all the panels square in every direction. Use the screws that come with the kit to hold everything together. Put the top on to help align the panels, but don't glue it until you have caulked all the internal joints, installed some braces, and installed the binding posts. After the top is on you will have to reach inside and caulk the last joints. I put a sandwich bag "glove" on my hand, squirt caulk onto my fingers, and spread it by touch.

I wanted a mono subwoofer, but I did not want to buy an electronic crossover or amplifier. I looked in the Madisound catalog for a dual voice coil woofer, and read their tables to find an alignment that fit the size of my box. In case you haven't seen a Madisound catalog, get one. For each of their house brand woofers, they provide a table of box sizes, port dimensions, f_3 points, and so on, that make it easy to pick the right woofer for your cart. My box's net volume was $4.25ft^3$ and the panel was big enough for a 12" woofer, so I looked at the tables until I found one that could use all that volume. I bought the woofer, made a tuned port of the correct diameter and length, and installed it.

The crossover is second-order, "by the book," and glued to the underside of the cabinet. No one can see it there, yet it is easy to access. I also put the binding posts there, so that speaker cables could be run underneath the carpet and then brought through a cut in the carpet up underneath the sub cabinet. That way the cables are invisible, as well. If you build a mono sub with dual voice coils, you can use flat, four-conductor cable, available at car stereo outlets. For grilles, front and back, use square molding.

Size it carefully so that after grille cloth is wrapped around the edges, it will fit snugly into the recessed area between the trim pieces and sides of the cabinet. This eliminates the need for attachment devices and makes it easy to remove.

Your result is a decent piece of furniture and deep bass, and it only takes a few evenings to complete. My sub is flat to 30Hz and very efficient. To push this method farther, you could use other pieces of furniture for a similar result. For example, you could buy a matching bookcase and convert the lower shelves to a subwoofer box or put tweeter and midrange units in the top half and use it for surround sound, or just to save space.

David West
Kent, WA 98031

Technology Watch

THE PARAMETRIC LOUDSPEAKER

By Peter Muxlow

It is not every day a new type of loudspeaker is developed, particularly not one based upon a totally different principle of sound radiation. Such is the case with the parametric loudspeaker,[1] which uses the nonlinearity of air compression/expansion to create a sound source with a highly directional sound beam. Never before has an acoustic spotlight existed in the audible sound region.

Because of its superior directivity pattern, the parametric loudspeaker is well-suited for use in public address systems. With this setup, it is possible for one group of people to hear the sound it emits without disturbing a neighboring group, thereby eliminating the need for expensive sound barriers between them.

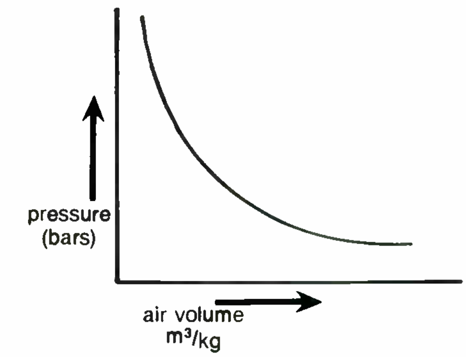


FIGURE 1: The effect on air volume as pressure increases.

Pain of Performance

A sound wave is a small pressure disturbance travelling through the air. For small changes in pressure (at normal loudspeaker levels), the change in volume for positive and negative values are identical. At high sound levels (120dB and higher), this relationship becomes nonlinear.* If large and equal, positive and negative changes in pressure occur in a mass of air, the volume change for pressure increase will always be less than the volume change for an equal decrease. If we take a high-amplitude ultrasonic wave of 40kHz, amplitude modulate it at 1kHz, then at 40kHz + 1kHz and 40kHz-1kHz, the carrier frequency 40kHz radiates in the air.

Because nonlinearity at high sound levels (*Fig. 1*) makes the air act as a "rectifier," demodulation takes place resulting in the original modulating frequency of 1kHz. The reproduced audio signal is highly directive because the "radiating diaphragm" size equals the ultrasonic beam wavelength.

The parametric loudspeaker does, however, come with disadvantages. You must stop dangerous high-level ultrasonic frequencies from reaching listeners to prevent injury. Screening the loudspeaker radiator with an acoustic filter will protect listeners. Soft 120mm-thick polyurethane foam acts as a low-pass filter for ultrasonic frequencies and allows only the audio to pass through. The patent details a number of acoustic filters and their placement, and provides methods for focusing the ultrasonic beam with moveable reflective plates and parabolic reflectors.

Another disadvantage of the parametric loudspeaker is its inefficiency. To obtain a audio level of 90dB, you must radiate 140dB of ultrasonic frequencies. The patent describes 120 ultrasonic transducers, 9.7mm in diameter set in an array 130mm by 100mm.

Despite these disadvantages, the main virtue—superdirectivity—ensures a practical future in the audio spotlight. If you are interested in experimenting with parametric loudspeakers, you will find an excellent article in *Measurement Science Technology*[2] to help you design and construct your own ultrasonic transducers.

REFERENCES

1. American Patent No. 4823908.

2. M. Rafiq and C. Wykes, "The Performance of Capacitive Ultrasonic Transducers Using V-Grooved Backplates," *Measurement Science Technology* 2, 1991, pp. 169-174, Printed in the UK.

FOOTNOTES

* Large horns which have narrow throats between the pressure chamber and the horn have this problem. This nonlinearity was first documented by A.L. Thurus, R.T. Jenkins and H.T. O'Neil, 1935, *Journal Acoustical Soc.* January Vol. V1 pp. 173-180.

Book Report

The Audio Dictionary

Reviewed by Gary A. Galo
Contributing Editor

The Audio Dictionary, 2nd Edition, by **Glenn D. White**. University of Washington Press, Seattle and London, 1991. 413 pages, paperback. Available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458. $19.95 plus $2 S/H.

In 1989 I offered an enthusiastic review of the first edition of Glenn White's *The Audio Dictionary* (*TAA* 2/89 p. 40 and *SB* 3/89 p. 49). This revised and expanded edition is a worthy successor.

White has added over 500 new entries, expanding the book by over 100 pages. Many entries from the first edition have been revised. White has apparently taken previous criticisms to heart.

In the first edition, under *microgroove*, he stated that Edison's LP record played for 40 minutes per side. I pointed out that this was incorrect; the actual playing time was 20 minutes per side. White has corrected this in the new edition.

I also criticized his definition of *bias*, as related to magnetic tape recording. Although White stated that the high frequency bias signal reduced distortion, he didn't explain why. This has also been corrected in the second edition. His definition is expanded, and easily addresses any reservations I had originally.

Improvements

As in the first edition, White's revision contains a useful type of cross-referencing. In each definition, he capitalizes any term which has its own entry. This makes it possible for you to acquire a substantial amount of knowledge on a subject by looking up all of the related terms.

Two appendices have been added, and both are well done. The first covers musical scales and tuning. White offers an explanation of the harmonic series and explains the concept of equal temperament. The author raises an interesting question about how modern instrumentalists play when they aren't accompanied by a piano (the piano is always tuned in equal temperament).

Some musicians still believe that an unaccompanied violinist reverts to something closer to just intonation. Audibly, if a string player performs in just intonation, the enharmonic tones don't produce exactly the same pitch. In other words, C-sharp is not the same pitch as D-flat. In any given key, the flats will sound "flatter" and the sharps will sound "sharper" when compared to equal temperament.

Suggestions

I find twentieth century string playing has reverted to equal temperament. It has become so familiar to the ears of performers and listeners, that most string musicians perform this way whether or not they're accompanied by a piano. Modern string players generally don't apply one tuning concept when they're performing Baroque string orchestra music, another for a Mahler symphony, and yet another for a Beethoven piano concerto. Some string groups may argue this point, but I believe them to be exceptions rather than the rule.

White's other new appendix is titled "Some Notes on the History of High Fidelity." He covers some of the important advances in high fidelity and stereo recording, going back to the experiments of Bell Labs in the early 1930s. There's one very minor error. White says Edison introduced his long playing record in 1929, but the year was actually 1926.[1,2] Otherwise, it's a brief, worthwhile survey.

Conclusions

The Audio Dictionary is quite different in scope and purpose from J. Gordon Holt's *The Audio Glossary* (*SB* 2/92, p. 32). First, *The Audio Dictionary* doesn't cover subjective terminology. Second, *The Audio Glossary* gives the reader brief, concise definitions without going into lengthy technical detail. It succeeds in its purpose.

The Audio Dictionary, on the other hand, offers a more in-depth technical explanation, yet it is still written so the non-engineer can understand the concepts. These two books complement each other nicely.

White's second edition is an excellent update to an already indispensable reference. You shouldn't choose between the two, both titles belong in your library.

REFERENCES

1. Gelatt, Roland, *The Fabulous Phonograph*, 2nd Ed. Collier Books, New York, NY, 1977.
2. Read, Oliver and Walter L. Welch, *From Tin Foil to Stereo*, 2nd Ed. Howard W. Sams & Co., Indianapolis, IN, 1976.

Craftsman's Corner

He Did It!

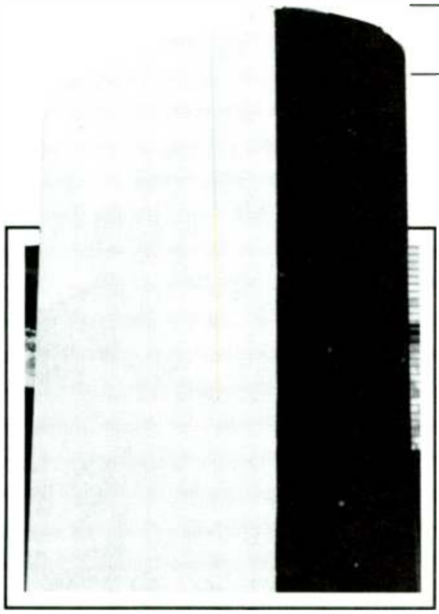


PHOTO 1: Though slender, this subwoofer's voice belies substantial weight.

Recently, I bought a pair of Magnaplaner SMGa speakers which I enjoy immensely. They are, however, a little weak on bass. After finding only high-priced subwoofers (of moderate quality) in retail stores, I bought four years of *Speaker Builder* back issues and decided to design and build my own subwoofer. The results cost less than $150 and the sound is magnificent.

My design requires 10" Philips woofers (at $10 each) with 10 oz. magnets, paper cones, and foam surrounds. I modified them with epoxy and BBs per Brian Smith's article, "Adjusting Woofers for High Performance" (*SB* 6/89, p. 22), then added a twist of my own—a compound driver sub-enclosure which tightens transients and lowers distortion. Instead of adding magnets *a la Smith*, I placed the driver magnets back-to-back. (It also helps to alleviate the speaker frames' tendency to vibrate at resonance.)

The enclosure is a simple, tapered transmission line stuffed with Dacron. The top, sides, and back of the enclosure are waferboard sandwiches filled with sand. These enclosures are so heavy, they are sonically "dead." I included internal wiring and terminal blocks on the rear (in a panel which unscrews from the front) to make the midrange- and tweeter-speaker installation easier.

The crossover is an active fourth-order kit from Marchand Electronics. It is inexpensive (less than $150 for two channels, the power supply, and a box) and simple to build.

The idea of building something this good for so little money really is exciting! I was listening to a Russ Freeman CD of The Rippingtons, and just couldn't believe what I was hearing. The bass was so loud and powerful, yet free of "boom" or "boxiness." On one cut, I put the volume up louder than normal and hit play. A tidal wave of clean sound surprised and delighted me.

I am not an engineer. I do not have a great deal of test equipment at my disposal, and this is only the second speaker I have ever built. I am an inexperienced builder who took low-cost materials and applied a simple design. After nearly 20 years of auditioning and buying audio, I have never heard anything quite like the sound my own speakers emit (at any price).

Edward Allen Thomas
Shelbyville, Kentucky

PHOTO 2: Apenthouse view of the Thomas tower.

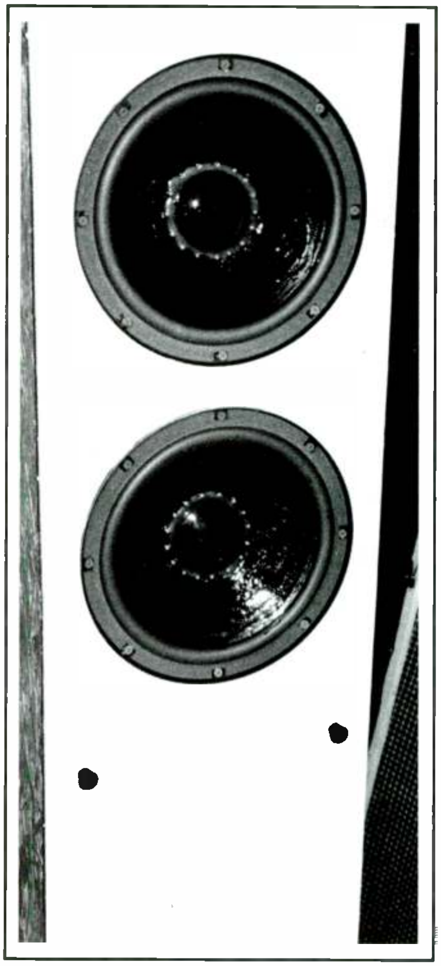


PHOTO 3: The horse's mouth: 10" Philips woofers.

SB Mailbox

CROSSOVER ERROR

Concerning the *Fig. 1* from Mike Somers' "A Loudspeaker with Active Crossovers and Delay" (*SB* 1/92, p. 37), the IC1 a-d has reversed inputs on all four segments. The corrected schematic appears in *Fig. 1*.

MORE INFO PLEASE

Chris Clarke's letter ("*SB* Mailbox," 1/92, p. 72) from Wynberg, South Africa, was most helpful as I have been seriously considering building a low-power amp with the TDA1520 chip. It looks like a natural for tweeter use in a multi-amp system. I am certainly interested in circuit boards if he has them available.

His reference to the TDA1514 chip was even more intriguing, however, and any more information he might have would be greatly appreciated. For example, where is the link-omission error? What size heatsink was found to be necessary? Getting information from Signetics/Philips (in the US) is proving to be not so easy.

Ron Neilson
Howell, MI 48843

CROSSOVER HELP

I've been trying hard the past four months to build a crossover for my newly finished speakers. After much work, I came up with the design in *Fig. 2*, but to my surprise my speakers just don't sound as good as the temporary Radio Shack crossover I was using before. The bass response is better—a little boomy and not

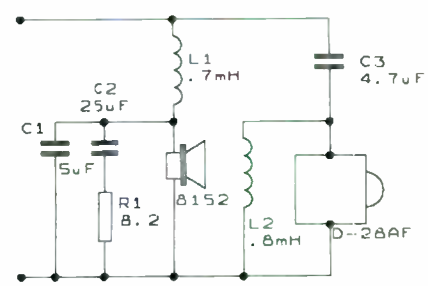


FIGURE 2: Diagram showing a Linkwitz-Riley crossover and 12dB/Zobel network with crossover at 2.5kHz. Equipment is a Madisound 8" polywoofer Model #8152 and Dynaudio D-28AF tweeter.

real tight—but the tweeter response is muddy and quieter than before. The overall sound isn't as dynamic as before.

I don't know where to go from here. Is my design at fault? Is there anything I can

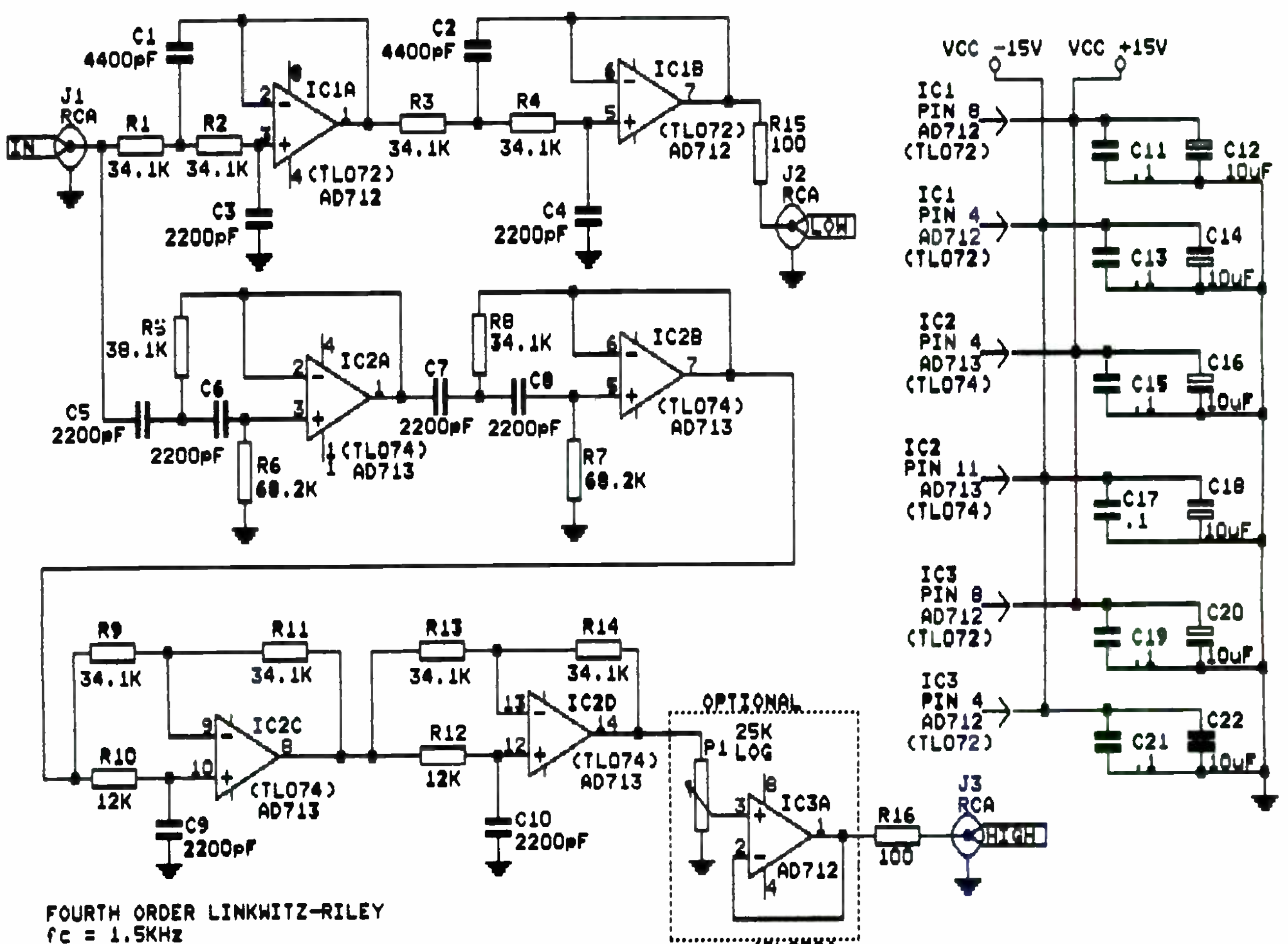


FIGURE 1: The corrected Linkwitz-Riley fourth-order crossover and the delay for one channel.

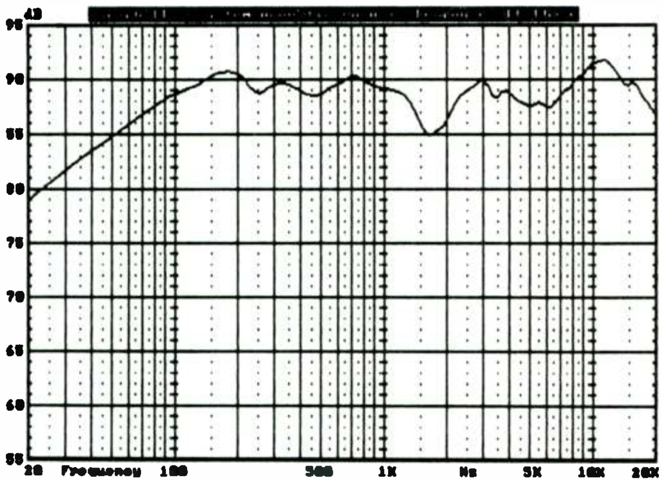


FIGURE 3: dB with driver offset; tweeter out of phase.

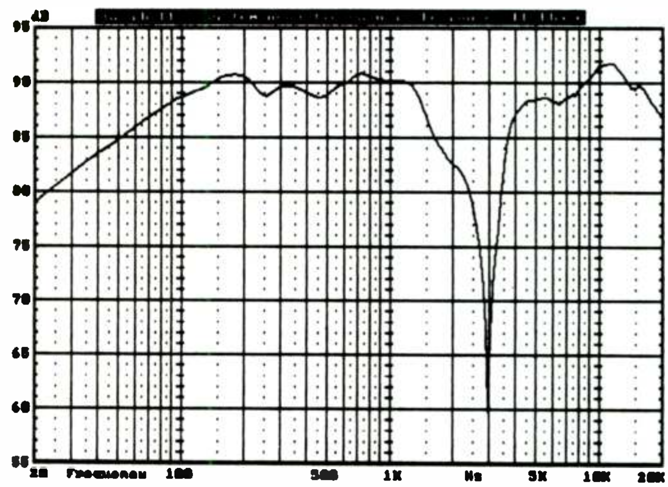


FIGURE 4: dB with driver offset; tweeter in phase.

change or do differently? Or should I try a new design? Can anyone help?

Matthew J. Masters
Columbus, IN 47201

Terry Cejka replies for Madisound:

We modeled the drivers and your crossover design. As you can see from the graphs (*Figs. 3* and *4*) with the driver offset taken into account, if the drivers are wired in-phase, there will be an audible suck-out around the crossover frequency. Therefore, you might try reversing the phase on the tweeter. This should flatten out the frequency response.

WORTH THE WAIT

I write in regard to Rick Bolen's request for crossover help on his E-V drivers (*SB* 1/92, p. 75).

I built a cabinet according to E-V's last plan, updated in 1982, for the Patrician system a few years ago using plans I bought from Dennis W. Ehricke (309 W. Harvard, Visalia, CA 93227). He is a retired E-V executive who has a lot of other material on E-V products. I do not know the dates of the printed schematics. I would bet that Dennis could provide much of the information Rick seeks.

I had the same experience that Rick had with the E-V customer assistance, but I received very valuable help from Joe Catowitz, an engineer at E-V. My E-V driver supplier, Donald J. Berg (2921 Meridian East, Puyallup, WA 98371, (206) 845-9764), was also quite helpful. He is an engineer, and I believe he has collaborated with E-V on some equipment.

I had wanted a Patrician system since the '60s. (No mere copies of the Klipsch speakers of the era. Differences in sound were obvious to listeners then and were quite a topic of discussion amongst audiophiles who knew a lot more than me.) However, I had no room or stable environment for mega-cabinets until I retired. By then E-V had stopped making them. At the suggestion of David Helder of San Francisco Soundworks (269 Bartlett St., San Francisco, CA 94110), I used a Marchand XM 1 electronic crossover which he custom built to accommodate some of my weird requirements (*Fig. 5*).

I use Parasound HCA-800II amplifiers to power the Patrician and the other channel (a Thiele criteria vented box built from E-V plans with E-V drivers new and old inspired by the original companion to the Patrician); couldn't be happier with the beauty and realism of the result. Worth a 30+ year wait.

D. S. Slack
Tracyton, WA 98393

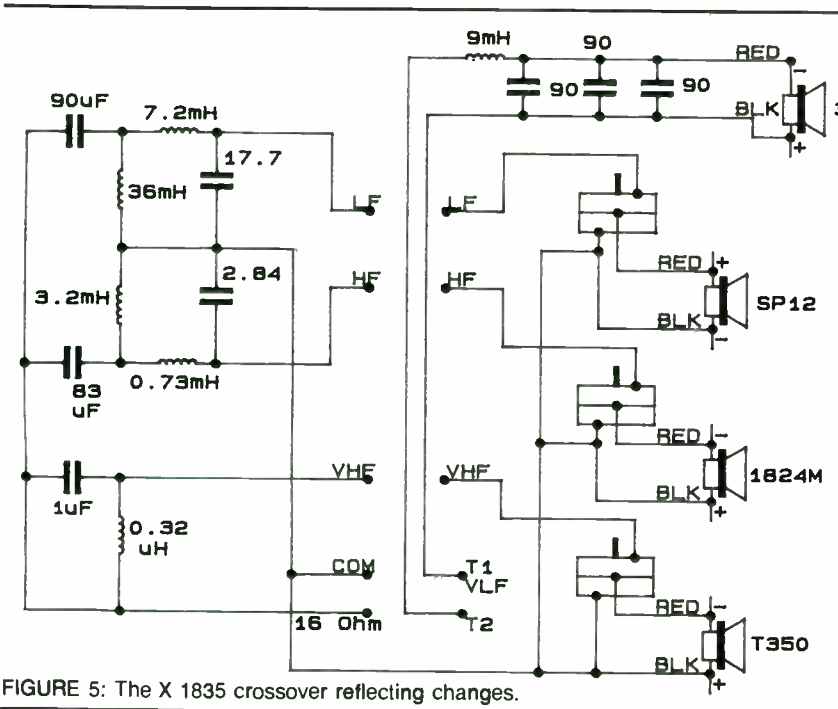


FIGURE 5: The X 1835 crossover reflecting changes.

ALTERING Q_{TS}

In G. R. Koonce's article, "Improved Vented Box with Low Q_{TS} Drivers" (*SB* 6/90, p. 23), the author examined empirically enlarging boxes and later tuning to extract lower bass. I have questions about altering the Q_{TS} of the driver itself.

In a QB_3 enclosure, the f_3 of a low Q_{TS} woofer is too high to be attractive and the V_B is so small you can't fit the driver *and* a tweeter on the baffle. A three-way design is out of the question. (For example, the Focal 7N515 has a Q_{TS} of 0.24, the f_3 is 60Hz, and V_B is 11.5 liters.) I propose using these woofers by raising Q_{TS} to the point where a low f_3 and a large enough box size result. I would merely put four or five paralleled power resistors in series with the woofer and its impedance compensation, recalculate Q_{ES} and Q_{TS}, and proceed. (For example, 2.5Ω of resistance—crossover inductors plus pure resistance—in series with the above woofer raises Q_{TS} to 0.33; then f_3 is 40Hz and V_B is 27 liters.

1. What are the unintended effects of adding this resistance, say up to 2 or 3Ω?

2. I realize that I will raise the impedance of the loudspeaker somewhat and reduce efficiency, but if I do this to 4Ω (or two paralleled 8Ω) drivers, do I care?

3. I am naive about amplifiers—does adding this much pure resistance (say, up to 3Ω) harm or change the speaker/amplifier relationship?

4. Are there guidelines for how much Q_{ES} and Q_{TS} can be raised without bad effects? Is there a desirable relationship between Q_{ES} and Q_{TS} that should be maintained?

5. I hope that paralleling several resistors will increase power handling, reduce heating, and minimize the inductance effects of the resistors? Yes/No?

6. What are the designers' and manufacturers' intended uses for low Q_{TS} woofers?

In the fourth edition of the *LDC*, Vance Dickason discusses a subject of great interest to me, the use of commercially available vibration damping materials to tame enclosure/panel resonance. There he references the relevant articles in *Voice Coil*. It seems to me these articles belong in *SB* as well, don't you think? I'm not anxious to subscribe to two magazines from the same publisher to find out what I need to know.

Thanks very much for any and all help. Keep up the great work—*Speaker Builder* is improving every year.

Ray Montoro
Plano, TX 75075

Contributing Editor G. R. Koonce replies:

Mr. Montoro has run into the problem that bothers all those trying to design vented box (VB) systems with low Q_{TS} drivers; even with a fairly low driver f_S, the system's f_3 is rather high. His basic question is: Will adding resistance in series with the driver to raise Q_{ES} and thus Q_{TS} produce a good VB system?

I will address this question in three parts: first, my actual experience that relates to the problem, secondly with direct answers to Mr. Montoro's specific questions, and finally what I have discovered you can do with low Q_{TS} drivers to produce acceptable VB designs.

The math is very simple. If you add a resistance R_G in series with a driver of voice-coil resistance R_E you change the Q_S as follows:

$$Q_{ES}' = \text{new } Q_{ES} = Q_{ES} \times (R_G + R_E) / R_E$$

$$Q_{TS}' = \text{new } Q_{TS} = Q_{ES}' \times Q_{MS} / (Q_{ES}' + Q_{MS})$$

I will relate three actual experiences that bear on this topic.

1. Mismatched drivers in a pair of Closed-Box (CB) systems:

I once built with two 8Ω drivers supplied by their owner which were badly mismatched. I felt the owner would want both boxes the same size so I decided to "fix" the lower Q_{TS} driver by adding resistance, about 1.5Ω being needed. The boxes were built to match the higher Q_{TS} drivers and I tried the 1.5Ω with the lower Q_{TS} driver. The sound was bad! Experimenting showed that up to about 0.5Ω was OK, but over that the sound quality decreased rapidly. With the 0.5Ω resistor the individual box sounded OK, but the pair did not. For those running into a similar problem, the cure was to introduce the loss in the box rather than electrically by adding resistance to the driver. I decided to try small, lossy ports in the box and started to drill ⅜" holes in the back, the size picked so it could easily be plugged with dowels! To my surprise each hole produced a Q_{TC} rise of about 0.1 and the system still behaved as a CB. With the right number of holes the two systems matched and sounded fine. I don't know the mathematics involved, but I have found this to work.

2. Series-Connected Woofers:

I build a lot of small systems with dual woofers. Many times I have tried hooking the two matched woofers in series rather than parallel to maintain a higher system impedance. Every time I have tried this, the bass was much better with the woofers in parallel, even though the math says it should not make any difference. I believe the sound is improved when the drivers are connected directly across the amplifier output.

3. Added Resistance with VB Systems:

I once tried adding resistance to an 8Ω 10" VB system and retuning it to see the effect. This work was aimed at establishing the maximum crossover choke resistance I could use and not really correcting for driver Q_{TS}. I found that if I got the resistance over about 0.5Ω the bass lost punch; became less solid. I have used this limit in my choke selection for years.

How can it be that adding resistance between the amplifier and the driver could cause problems when the math does not indicate any problem? Some tube amplifiers have rather high output impedance and seem to work just fine. I believe there are three possible effects involved here.

The first is that dynamic drivers are always specified and tested as terminal-voltage-in vs. sound-level-out. Even though we know it is the current that causes the cone motion, the manufacturer has gone to a lot of trouble to make the acoustic output flat based on input voltage. The T/S design alignments make this same assumption. When you put resistance between the amplifier and the driver, then you form a frequency-sensitive voltage divider that will increase the driver output at the two impedance peaks displayed by the VB system and these are not normally good places to exaggerate. If the resistor is small the effects are not great, but when you talk about adding a resistor that is a large portion of the driver resistance, I believe you get frequency sensitive variations along with any loss of damping control.

Next, it is common for power amplifiers to have feedback taken right at their output. This feedback sees the voltage generated by the driver and helps the amplifier control the driver motion. With added resistance you have again added a voltage divider, this time in the driver voltage fed back to the amplifier. I suspect this may be part of the problem.

Finally, for me, I have always preferred the sound of VB systems built with low Q_{TS} drivers and do not want to raise Q_{TS} any more than I can.

I have no mathematics to support these concerns, but have built enough systems to consider them "rules." I never use woofers in series, VB or CB, and will not allow the choke resistance to get to 0.5Ω for 8Ω drivers. I would love to hear from other speaker builders on this topic.

Now to Mr. Montoro's specific questions:

1. "What are the unintended effects of adding this resistance, say up to 2 or 3Ω?" I believe I have already answered this to the best of my ability.

2. "I realize that I will raise the impedance of the amplifier somewhat and reduce efficiency, but if I do this to 4Ω (or two paralleled 8Ω) drivers, do I care?" I believe that you will. I believe you can only put about half the resistance in series with a 4Ω driver before it would start to hurt the sound relative to what you could put in series with an 8Ω driver. Also, I believe that driving paralleled woofers through any major resistance, is a bad choice. Usually low Q_{TS} drivers have fairly high sensitivities so I don't think woofer sensitivity is the problem.

3. "I am naive about amplifiers—does adding this much resistance (say up to 3Ω) harm or change the speakers/amplifier relationship?" Yes, it does harm this relationship in both directions and is to be avoided. In the case of tube amplifiers, which are sometimes bothered by driving a load impedance higher than what they are rated for, raising the load impedance can be dangerous.

4. "Are there guidelines for how much Q_{ES} and Q_{TS} can be raised without bad effects? Is there a desirable relationship between Q_{ES} and Q_{TS} that should be maintained?" The first part of this question has been answered from my experience as about 0.5Ω per 8Ω driver or R_G about 6% of nominal driver impedance. When we talk about the relationship of Q_{ES} to Q_{TS} we really mean the ratio of Q_{ES} to Q_{MS}. After I had built about a dozen VB systems, I looked at this and thought I had a relationship for the systems that gave the best results. After building many more systems that ratio was not sustained. At the present time I know of no ratio of Q_{ES} to Q_{MS} that produces better VB systems. I still tend to like the damping to be in the electrical circuit and tend to go for low Q_{ES} with a high Q_{MS}, but have heard fine systems built with relatively low Q_{MS} drivers.

5. "I hope that paralleling several resistors will increase power handling, reduce heating, and minimize the inductance effects of the resistors." Using parallel resistors will reduce the total circuit inductance and increase total power handling. The heat from each resistor will be reduced, but the total heat dissipated by the group will remain the same as for a single resistor of the same resistance as the group.

6. "What are the designers' and manufacturers' intended uses for low Q_{TS} woofers?" There is no way I could know the intentions of designers and manufacturers. I do know that professional sound-reinforcement people like to build with low Q_{TS} drivers; partly because low Q_{TS} generally means a big magnet and thus high efficiency. This indicates two items you must look for when buying a low Q_{TS} driver. First, you want a very low f_S to result in a system with acceptable f_3. Second, you must be sure the sensitivity is reasonable. If you build using woofers with a sensitivity of 98dB at

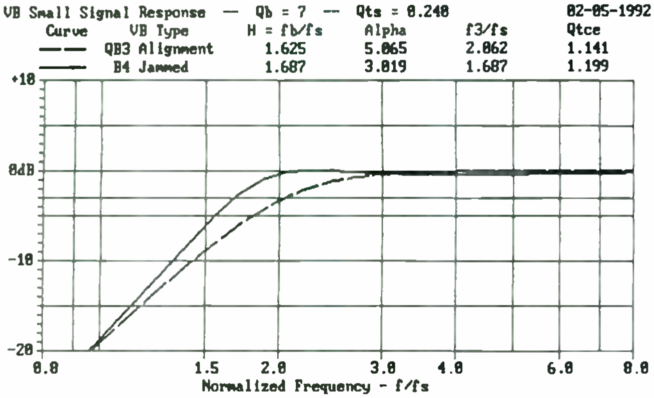


FIGURE 6. Comparison of QB_3 and B_4 alignments for a driver with Q_{TS} = 0.24.

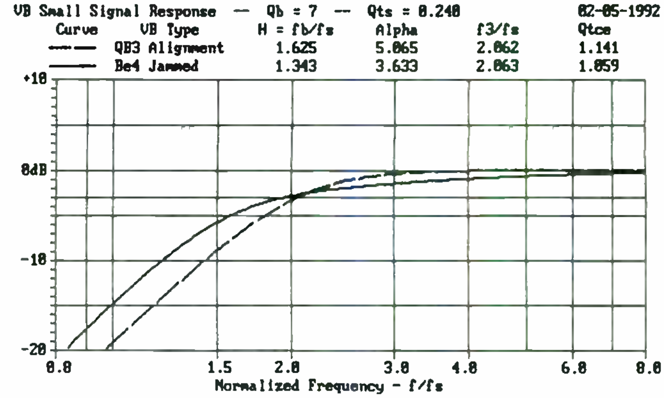


FIGURE 7. Comparison of QB_3 and Be_4 alignments for a driver with Q_{TS} = 0.24.

1M for 2.83V RMS input you will not find midrange drivers and tweeters to run with them and be forced to use horn upper-end drivers or biamp the system. This high sensitivity prevents me from using the drivers designed for professional application.

Let me now address what the options are for the use of low Q_{TS} drivers other than adding series resistance. Part of the problem is that most people tend to use these drivers in a QB_3 alignment. In my article which Mr. Montoro referenced, I tried to show how I was getting around this problem with a trial-and-error alignment approach. Since that time, I have been contacted by Mr. R. O. Wright, Jr., who showed me there is a direct mathematical way to produce VB designs with improved performance over the QB_3 alignment. The technique is called Complex Alignment Jamming and Mr. Wright and I are finishing a manuscript for submission to *Speaker Builder* on this topic. *Figure 6* shows how the Focal 7N515 driver, or any driver with a Q_{TS} = 0.24, can be used in a "jammed" Butterworth B_4 alignment and how that alignment compares with the more standard QB_3. A considerable extension in f_3 is demonstrated by the B_4 alignment. A second alignment choice is demonstrated in *Fig. 7*, where the QB_3 alignment is compared with a "jammed" Bessel Be_4 alignment. The Be_4 has nearly the same f_3 as the QB_3, but provides a very slow roll-off rate. At −6dB and −10dB the Be_4 shows considerable advantage. The Be_4 alignment is very popular with professional sound-reinforcement people and is believed to offer lower distortion than the same driver in other alignments. I have never built using the Be_4 alignment, but I plan to give it a try.

I thank Mr. Montoro for this opportunity to "push" the use of low Q_{TS} drivers as I believe they produce the best sound. I also want to encourage builders to break away from the conventional QB_3, BB_4, and C_4 alignment curves as they cover only a very small portion of the "alignment space" available for VB design! What experiences are readers having in this area?

CALSOD HELPS

This letter is a follow-up of my letter in response to a letter of Herbert Meyers (*SB* 3/90, p. 3) in which I pointed out that the phase-correcting portion of the Dynaudio

Myrage crossover was in fact a passive time-delay network and urged further information on calculation of the properties of these potentially useful networks. This request has become moot by my recent acquisition of the CALSOD software (which OCSL distributes). I am now in a position to offer some specific comment to Mr. Meyers and to sing the praises of this amazing software (if one is considering any design software, CALSOD should be first on the list; it does everything).

While I was not (and indeed am still not) able to provide Mr. Meyers with formulas for computation of time delay, I can provide some general comment and, with the help of CALSOD, some specific and perhaps helpful data:

By inspection these networks are passive, all-pass filters derived by summing first-order and low-pass sections. Therefore, rule 1 is that there must be a (center) frequency (f_O at which $IMP_{inductor}$ = $IMP_{capacitor}$ = $R_{load\ (driver)}$. If such a filter is terminated properly, i.e., constant resistance, it will have a flat amplitude response and introduce a time delay determined by the properties of the low pass section. The lower the frequency (f_O) at which $IMP_{inductor}$ = $R_{load\ (driver)}$, the greater the delay. The delay is relatively constant from DC to somewhat below f_O, then begins to diminish—reaching roughly 25% of its initial value about three octaves above f_O. While I haven't found formulas or tables that associate the delay with f and f_O, simulation with CALSOD (which affords simulation and comparison of both phase and amplitude of the driver and driver/filter combination) suggests that the network in the Myrage, terminated with 4Ω (and therefore having f = 796Hz) produces a delay of 0.37mSec at low f. This diminishes to 0.33mSec at 600 and 0.94mSec at 5kHz. A similar network found on the D-28AF tweeter in some Dynaudio designs employs 6.8μF caps and 0.39mH inductors for an f_O of 2930Hz (with a nominal termination of 8Ω) and a delay of about 0.075mSec at 3kHz. Interpolation or extrapolation from these values should produce reasonable delay approximations for other f_O. It must be emphasized that a constant, resistive termination of these networks is vital if the all-pass, flat-amplitude response is to be preserved (note the Zobel on the D-76s). Otherwise amplitude response will sag at frequencies where impedance is below nominal and be boosted at those where impedance is above nominal. Simulation suggests that the uncompensated D-28 network is offensive in this regard.

"Reverse engineering" of complex crossover circuits is merely one of the areas where CALSOD excels. Along with its capacity to model and optimize crossover circuits which incorporate driver phase and amplitude behavior into realistic acoustic models, it has adequate, simple routines for modeling low frequency behavior (closed box, vented box, PR, etc., with and without equalization), and the ability to incorporate user-defined transfer functions expands these considerably (one needn't understand the TFs, you merely calculate coefficients and copy them). For instance, one could incorporate the sound pressure TF for a symmetrically loaded alignment (given by Mr. Margerand, *SB* 6/88, p. 29) and the impedance TF for same (alas, not given) with a series cap and inductor and simulate a sixth-order bandpass alignment; moreover, one could call upon CALSOD to optimize the values of passive components to achieve a target alignment.

While this software permits anyone with a little algebra to enter the arena of design on equal terms with the pros, there is a good deal of art in achieving a good CALSOD model of driver behavior (particularly near the HF cutoff) and judgement is required in choosing functions to simulate diffraction loss (the 6dB step), rising high end, etc. I would like to suggest a CALSOD user group or some other mechanism (including articles in *SB*) to circulate good driver models and useful TFs.

Robert Hesse
Cambridge, MA 02442

BANDPASS QUERIES

I have been reading *SB* for a year now and I haven't seen anything on off-shoots of the vented/sealed designs. Are there any formulas available to the hobbyist for designs of sealed-vented bandpass (sixth-order), double-vented bandpass, or series-vented bandpass? If so, can you provide me with any of them?

I also have a question to either Bob White or Bob Bullock regarding the computer program "Passive Crossover." I don't have copies of *SB* 1-3/85 but I assume that Vance Dickason's *Loudspeaker Design Cookbook*'s crossover drawings are similar. What is the definition of R_A and the difference between T-bandpass and C-bandpass? Could you please clear these up for me? Thanks.

David Samek
Blue Earth, MN 56013

Contributing Editor Bob Bullock replies:

There are no simple formulas for bandpass systems. However, Old Colony sells a program of mine called BANDPASS BOXMODEL that allows you to model a wide variety of bandpass systems with passive and/or active equalization. It is easy to use this software to come up with a design for a given driver.

Your question on passive crossovers is answered in my three-part series on "Passive Crossover Networks" (*SB* 1-3/85).

Briefly, C-bandpass and T-bandpass circuits are just two different topologies for realizing the same transfer function. The main difference is that the C-bandpass circuit generally provides more midband gain. Thus, it can sometimes be used more effectively to match sensitivities in three-way systems.

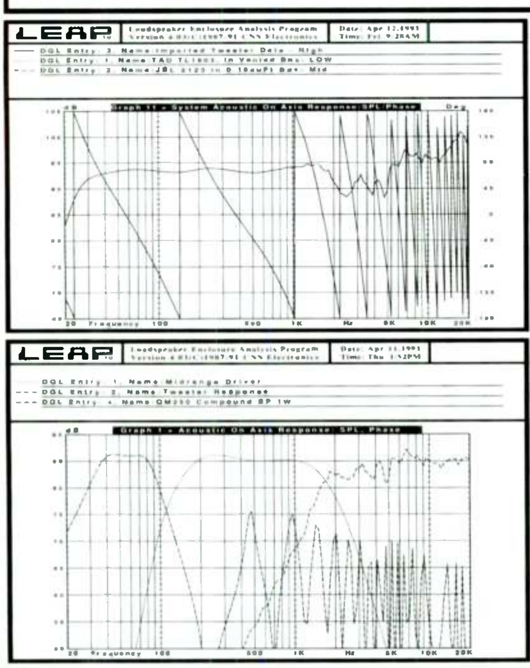
The role of R_A is to attenuate the midband gain so the bandpass and the low- and high-pass sections combine correctly to provide an overall all-pass response.

The "gain cut" value is the amount R_A attenuates the bandpass output. If your midrange driver is this much less sensitive than your woofer and tweeter then you do not need R_A. In a few cases, the band-pass section does not supply enough gain to match the low- and high-pass sections. In this case a "gain error" is displayed. This means that you must supply this much additional gain in the bandpass section in order for the three sections to match correctly.

BANDPASS BOXMODEL with Graphics (#SOF-BPB1B5G) is available from Old Colony (see ads in this issue for ordering information) for $50 plus $2 S/H. A demo version (#SOF-BPB1B5GD, $5, plus $2 S/H) is usable as credit toward later purchase of the full package.

ACOUSTICAL IMPLICATIONS

May I take this opportunity to say how I have enjoyed reading *SB* over the past few years and I am very encouraged by the strong commitment shown by your readers and both the amateur and professional authors. I found the article by Dick Campbell on MLSSA (*SB* 1/92, p. 26) to be particularly interesting and Dick has



made a genuine contribution to the art of testing. Likewise Dave Moran (*SB* 1/92, p. 82) has a refreshingly objective attitude to the performance of commercial loudspeakers. Frequently they are as poor as Dave suggests.

He is right to focus on frequency response as a primary parameter, if only because so many examples are so dismal in this area. Assuming reasonable design in terms of directivity and driver integration, dangerous assumptions I know, then an inability to throw a decently flat frequency response in the forward direction must be regarded as a major weakness. If serious variations are present, the speaker in question is acting as an unwanted and rather colored graphic equalizer.

If the response varies a lot with axis, then nowhere is flat. Such a sonic disaster will be different to every listener, at every angle, in every environment. Moreover, the acoustical implication is that this type of speaker has a discontinuous phase response imparting rapid shifts between

CAVEAT CORRESPONDENTS

Things that go bump in our round file:

1. "I'm thinking of building a 16-in, 8-out console in my basement. What tape recorder should I buy?"

2. "Is my Fisher Z-705 receiver worth updating? Where should I begin?"

3. "Although I forgot to enclose a stamped, self-addressed envelope, please answer the following nine questions based on my experiences building your inverted RIAA kit."

4. "Please forward this (unstamped) letter to Ralph J. whose letter appeared in one of the 1970 issues—don't remember which."

5. "I have a Milhous 10W integrated stereo amplifier and a Gesundheit turntable. Which of the following six cartridges would you recommend?"

6. Queries with no stamped, self-addressed envelope or postal coupons enclosed.

7. Letters without return addresses on them whose envelopes have strayed away somewhere.

8. Illegible hand-written letters scrawled on odd scraps of paper. If you have no access to a typewriter, please try to be sure our typesetter doesn't lose his eyesight and his mind in deciphering your writing. (This is especially important if you want us to publish your classified ad.)

the two channels. The consequence is poor, unstable stereo imaging.

For normal-sized, $2ft^3$ systems it is routinely possible, with careful design, to generate a well-integrated forward output. If this is measured in ⅓-octave weighting, it might well deliver +2dB, 60Hz–18kHz for a 15° included vertical window and a 40° horizontal window. If basically clean-sounding, such a design will sound neutral and well-balanced in a range of rooms. Even when measured under non-anechoic conditions a good proportion of that intrinsic flatness and frequency control will be clearly evident in the results.

Large panel systems pose special difficulties due to the proximity effects which make assessment of test data difficult. Conversely this very fact also confirms David's findings that some large panels are often less accurate (by whatever definition you care to apply) than conventional boxes. A good system design of essentially flat frequency response and good driver integration will remain accurate under most conditions.

Keep up the good work.

Martin Colloms
London NW6 1BA, England

FOREIGN READERS

Due to the limitations of our Fast Reply program, we are unable to process requests for information from our foreign subscribers.
HOWEVER, our advertisers would certainly like to hear from you directly via mail, phone, or FAX. Please don't hesitate to contact them for the products / information you are interested in.

EFFICIENCY ALIGNMENTS

I must first commend you on a truly fascinating publication. My colleagues and I, in the automotive audio industry, turn to the pages of *SB* for its technical insight into the formidable challenge of accurately reproducing what has been captured live onto recordings; a challenge further complicated by the grossly unpredictable acoustics of the automotove environment. In my endeavor to achieve realistic bass response, "Symmetrical Loading for Auto Subwoofers" (*SB* 6/90, p. 20) by Matthew Honnert, caught my eye.

Further exploring the subject, I referenced Jean Margerand's article, "The Third Dimension: Symmetrically Loaded," Parts I and II (*SB* 6/88, p. 29 and 1/89, p. 27). His articles are extraordinarily informative; however, there are a few questions I would very much like to have answered.

Is there an upper limit to the amount that the reference efficiency of a driver may be increased? Or is it determined by the qualities of the driver, i.e., surround and cone materials, X_{MAX}, and so on?

If not, what are the logarithms used to obtain the values on the tables? I would like to increase the reference efficiency of very low F_S/Q_{TS} drivers +10–12dB by extending the data range on the tables—if that much increase is possible. Attempting these very high efficiency alignments for several low F_S/Q_{TS} drivers has forced me to resort to a lower "S" table, while I could possibly have achieved a higher efficiency and a flatter response by using the higher "S" value table. Since most of my applications will be used strictly as subwoofers, the narrower bandwidth would be tolerable, even welcomed.

Is there a formula for determining a predicted, normalized frequency response graph for an alignment? If not, are there any suggestions as to how I may depict a response graph on a computer screen, transposed onto a known typical response analogous to each "S" parameter, and then simply apply the frequencies to the bottom and the relative sound pressure levels on the side of the chart corresponding to the data calculated for a custom alignment?

Is there a general consensus as to how low FL may be "set" for a given driver, dependent upon either the alignment's "S" parameter or the F_S of the driver itself? Or is it assumed that the driver will, in fact, reproduce frequencies in the lower extremeties (20–25Hz) even without an F_S that low, and will merely excurse more since the drivers must excurse much less to reproduce any given frequency while symmetrically loaded?

When designing multi-driver woofer systems, should I determine F_S, Q_{TS} and V_{AS} parameters for the alignment by simply adding the individual F_S values to one another, then dividing by the number of drivers—repeating this process for Q_{TS}, then V_{AS}? Or use a different approach?

Most of my automotive applications for subwoofers are driven by mono-bridged

Why Your Magazine Doesn't Arrive

If you move and do not send us your new address in advance, your current issue is sent to your old address and may arrive after the Postal Service's six-month time limit on forwarding instructions has expired. If this happens, the Postal Service sends us a notice saying, in effect, they have forgotten where you now live, along with a bill for 30¢ to pay for this information. Meanwhile, they have trashed your magazine.

Since we have already paid for your destroyed copy, paid postage to attempt to deliver it to you and put a holding code in your computer record saying that we have no idea where you are, and further since we routinely act on the assumption that Murphy is overactive within most offices of the Postal Service, we send a first-class letter to your old address to determine whether you have, in fact, moved. We do this because from time to time some clerk mistakenly supposes a subscriber has moved and he or she has not. Then we write a nasty letter to the Postmaster suggesting the clerk is a vandal of sorts and how about paying for the magazine? So far, not one has sent any postage or even a repentant letter. Replies usually range from form to frowning to foaming.

So please inform us early when you move. Peel off your label, put it on a postcard, write in your new address and send it to us. We cannot afford to send you a duplicate, unfortunately. If your copy becomes a Postal Service casualty, the replacement copy will cost $5.

amplifiers—usually high current models—and I oftentimes use more than four drivers. With these high current amplifiers, all drivers are wired in parallel. Is there a reference in *SB*, or in another publication, that explains the calculations of series/parallel acoustic and voltage gains using these multi-driver systems?

I realize that the box volumes, tuning, and enclosure construction are critical to achieving these precise alignments. Of particular concern, I understand, is the volume front (VF). I am sure that I should add an approximate amount to the net volume for the drivers when mounted in a push/pull or Isobarik configuration, as determined by the diameter; then add an equivalent vent volume. What values should I use for approximated equivalent driver volumes for drivers of 6", 8", 10", 12", and 15" diameters?

Afterwards, shall I add the "mystical" 10–15% minimum overvolume, as do so many avid speaker builders? Or shall I assume, after having accounted for all port, driver, and bracing volumes in my gross volume, that my figures are quite accurate enough?

Concerning the tuning of the system; is tuning achieved by the same means as a bass reflex enclosure—by tuning to the low impedance trough between the twin impedance peaks?

William D. Sanders
FPO, AA 34080-2730

STUFFING 8′ TLs

Just when you thought it was safe, here comes another question concerning the Dynaudio 30W54.

I am a beginner speaker builder trying to gather my courage to start construction on my first speaker project, a transmission line (TL) subwoofer using the 30W54. After doing my research and calculations, I am still unsure about the optimum length of my line. From past issues of *SB*, I have read many articles describing the relationships of different stuffing materials and densities on optimum TL length. However, most authors advise "fine tuning" the line by adding or subtracting stuffing material to the completed line. This seems to suggest building the TL backwards. That is, figure my stuffing densities first, then build the line.

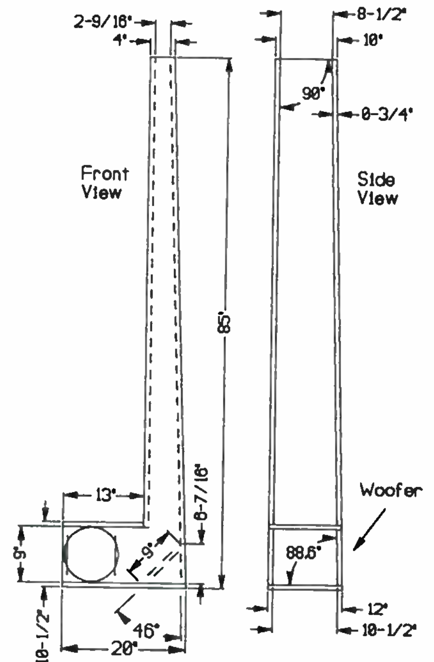
Due to my limited resources, I am unable to build a changeable prototype and test various length and density combinations. When I was about to scrap the whole project, I read Roger Sanders' reply to Gary Beckstrom ("Mailbox," *SB* 3/91, p. 89) where he showed a subwoofer design using 30W54 in a TL of 8′. Unfortunately, Mr. Sanders did not mention his stuffing strategies.

In addition to this, Bob Bullock's response to Robert Hesse ("Mailbox," *SB*

4/91, p. 80) showed a simulation of the behavior of a 30W54—also in an 8′ TL.

Could it be that 8′ is generally agreed upon as being the optimum length for a TL enclosure using the 30W54? If so, how can such a line be stuffed?

Thomas Gillin
Palm Coast, FL 32137



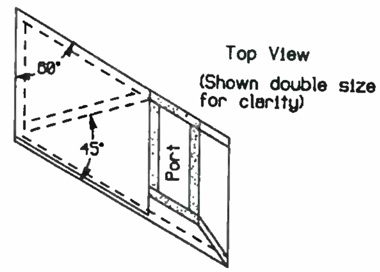


FIGURE 8: A prototype transmission line.

Roger R. Sanders replies:

Your concerns about line length and stuffing are understandable and fortunately, are easily resolved.

Usually scientifically developed devices (like speaker enclosures) require precision in their design and execution. Interestingly, TLs are an exception. They are remarkably non-critical in the areas of cross-sectional area, line length, and stuffing density.

Very briefly, a TL's purpose is twofold. First, it slows and delays the rear wave from the driver so at some low frequency it will have shifted the phase

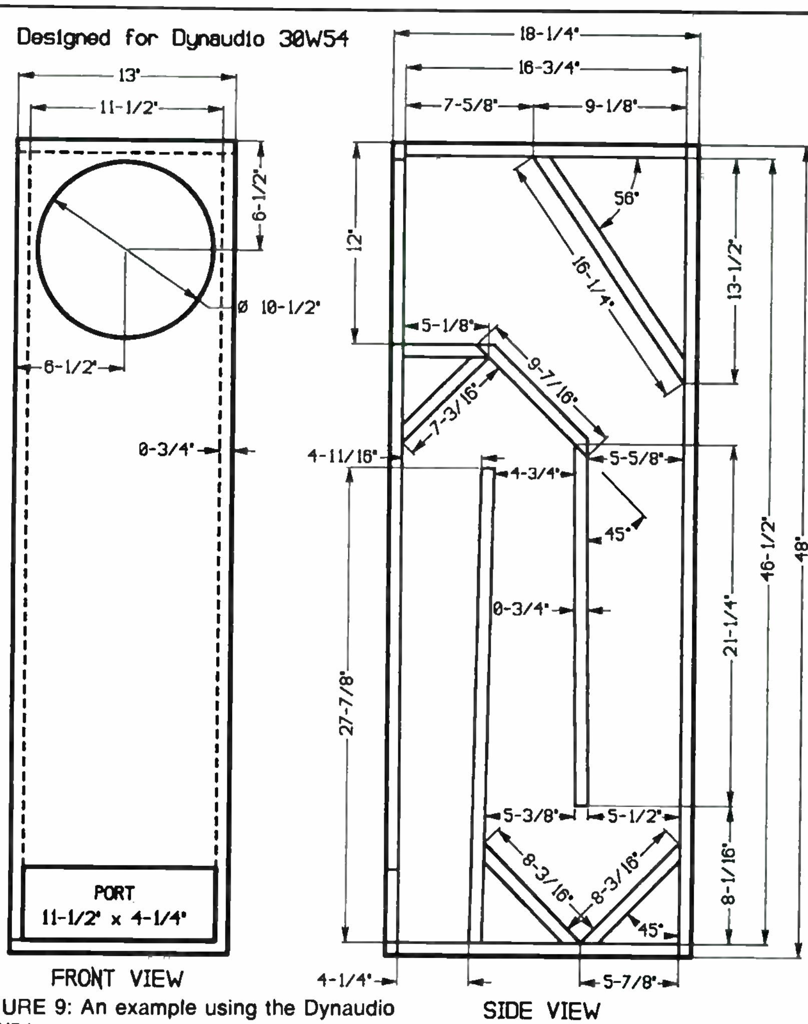


FIGURE 9: An example using the Dynaudio 30W54.

180°. At this frequency it exits the line in-phase with the driver's front radiation. This augments and supports the deep bass (which will be falling due to lack of radiation resistance). Shifting the phase is the TL's way of extending the deep bass.

Second, a TL minimizes resonances. A properly designed TL is tapered and stuffed with damping material. A typical box enclosure has two or three large resonances. The taper in a TL causes the line to have an infinite number of very tiny resonances. The damping material attenuates these so a well-designed TL is nearly resonant-free (*Fig. 1*).

The general rule for line length expounded by Bailey, one of the early pioneers in TL development (*Fig. 2*), is that the length should be one-quarter of the wavelength of the lowest frequency you want linearly reproduced. For example, a 32Hz sound wave has a wavelength of about 32′ (divide 1024 by the frequency to find the wavelength). One quarter of that is 8′, so 8′ is a commonly used line length.

But line length is not critical. After all, the stuffing in a TL changes the effective length of the line (because it slows the propagation of sound), so the above formula is only an approximation in any case.

Let's look at the extremes of line length. Long lines are large, so some designers have tried to shorten the line as much as possible to make TLs more compact. They run into problems when the line approaches about 6′ because it decouples from the woofer at low frequencies.

A properly operating TL has very tight control of the driver's excursion, which it damps heavily. Woofer excursion in such a TL is very small. But when the line is too short, it loses control and the woofer can have extremely large excursions. At some low frequency the excursion can be so severe that it can damage the woofer. This is not likely to be a problem with music, but is often troublesome when you rapidly run through the FM band on an analog tuner, drop the tonearm on an LP, have excessive turntable rumble, or fast que tape.

At the other extreme, a TL that is 12′ long becomes so large as to be impractical. Also, the 180° phase shift point in a 12′ line is around 20Hz, and few drivers perform well enough at such low frequencies to effectively use such a long line.

You can see that the range of line length lies between 6′ and 12′. I tend to lean toward shorter lines for smaller drivers, because these usually have higher fundamental resonances. Also, the only reason to use small drivers is for tight space considerations anyway. I use longer lines for larger woofers with low fundamental resonances. The Dynaudio 30W54 is such a woofer, and it performs splendidly in transmission lines in the 8-10′ range.

Although some designers try, I am not aware of a *precise* way of determining stuffing density in a TL. Any method of calculating density is only useful for a specific type of material, because different materials (wool, polyester, foam, etc.) have different damping properties. A meaningful formula must somehow account for all these, and I have never seen it done.

Even if you could devise such a formula, how would you accurately measure your work? For example, a common damping material is natural long-fiber wool. In lines of large cross section, a value of one-half pound/ft^3 is often stated in the literature. But just how do you determine what is 1ft^3 in the crazy shapes we make into tapered lines?

To further complicate things, we usually arrange the damping material to produce a constant impedance. This means we stuff a little tighter near the driver than at the port. How much denser does it have to be to maintain constant impedance? Also, as you reduce the cross section of a line, you must reduce the damping density. How can you account for all these variables?

Fortunately, it isn't at all difficult to do subjectively. If the line is understuffed, it will sound "boomy," "hollow," "muddy," or any other subjective term to describe a speaker with poorly controlled midbass. If the line is overstuffed, it is clean, but lacks deep bass.

I haven't found that the *type* of damping material matters. There are those who believe that wool is best, and I agree that it is very effective, however, it requires moth-proofing, is expensive, and must be supported with netting or dowels every 10″ to prevent settling. As you will see shortly, the need for support makes it difficult to adjust.

Polyester (Dacron®) fluff is commonly available from fabric stores (it's used to make pillows), and is cheap. Moths despise it and it doesn't settle. It is not as effective a damping material as wool, but this is not a problem—just use a little more to get the necessary effect.

Other synthetics on the market work well, but are very expensive and you can get the same result with other materials. I used to use wool, but now I use polyester fluff.

OK, you say, all this theory is fine, but I still don't know how to stuff a TL—tell me what to do! First, pick the length of line that best meets your needs. Eight to ten feet is a good compromise. Make your choice *without* regard to the type of damping material you plan to use.

Build the enclosure, but before attaching the last side, stuff with damping material. To get the density right, *don't* think in terms of stuffing the enclosure. Instead, think in terms of just lightly and gently *putting* the material in the line. Most builders use too much!

I find the problem is getting the line completely filled with fluff while keeping the density low. I have to tease it out to get complete coverage without over-stuffing the line.

You want the material a *little* denser behind the woofer than at the port. The way I do it is simple: I omit damping material in the last 2′ of the line (near the port), *very* lightly put material in the middle, and lightly stuff near the driver.

Put on the last side of the enclosure. Go ahead and make it permanent—if necessary, you can adjust the damping without taking it apart later. Complete the woofer system as though you were confident that the stuffing is perfect.

Hook it up and listen. I'll bet it will sound clean with excellent bass extension just as you've built it. You probably won't have to adjust the density of the damping material.

Occasionally things aren't quite right. If there is no deep bass, you need to remove some stuffing. If it sounds boomy, you need to add some. Adjustments are easy.

The critical damping material is placed directly behind the driver. By removing the driver, you can put your arm into the enclosure about 3′ through the driver opening and easily adjust the density. You can now see why wool is hard to adjust—the support netting it requires every 10″ prevents you from reaching deep into the enclosure.

Replace the driver and re-test. Surely you will have it right. None of my readers/builders have ever reported having to make more than one adjustment—and most say their TL worked perfectly from the start.

I hope this information has been helpful. If you have questions or problems, feel free to write to me through *SB*. It has always been my policy to support my readers.

ON SPL FORMULAS

I have a question about the formula for converting eta nought to an SPL in dB. I used the formula SPL = 112 + 10log(eta). This doesn't seem to work correctly, even if I calculate eta as a function of the other parameters.

For example, the Ramsdell Audio 27″ driver has a published eta of 4%. This yields a sensitivity of 112 + 10log(0.04) = 112 + 10(−1.4) = 98dB, which is 5dB lower than the published SPL of 103dB.

Another example is the JBL E130. Eta is listed as 8.6%. 112 = 10log(0.086) = 112 + 10(−1.06) = 101dB, while the figure given in "Integrite: Rational Speaker Design" by Marc Bacon (*SB* 4/91, p. 25) is 105dB.

I posted this question on Madisound's Audio Projects BBS, but had no replies. Any help would be greatly appreciated.

Dave Degelau
Madison, WI 53703

Contributing Editor Bob Bullock replies:

The formula SPL = 112 + 10log(eta) cannot be expected to be accurate for the purpose of converting efficiency to sensitivity because it is derived from a formula for finding the steady state SPL in a room provided by a source of efficiency eta.

In particular, the value 112 results from making certain assumptions about room absorption, room size, and source directivity. These values will not necessarily reflect the conditions under which the sensitivity measurement was made. For example, in the appropriate room the 112 might become 117, in which case the conversion for the Ramsdell driver would be exact.

If you are interested in details, consult Chapter 10, Part XXIV of Leo Beranek's *Acoustics*. (#BKAC5 costs $27.95 plus $2 S/H from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467.)

Quality Cabinet

continued from page 23

she is in my livingroom when my eyes are closed. The amount of information in the sound, such as the room acoustics where the recording took place, is evident on good program material. I have enjoyed going through my CD and LP collection, playing favorite selections and hearing subtle details that were never apparent before.

HAPPY BUILDING. The quality or lack thereof is immediately apparent in the recording. In other words, poor recordings sound even worse and excellent recordings are awe inspiring. Some of my CDs sound thin and have no depth at all. I finally understand all those terms like "soundstage," "air," "front-to-back placement," and so forth. Obviously, I am very pleased with the way they turned out.

The pedal coupler and second amplifier were finished during the following winter and the bass is truly incredible. On the Kit and Coco CD *In Time*, several selections have really low bass that I never realized was there.

I hope you will be able to use some of the ideas presented here on your next project. I was amazed how helpful and time-saving the use of jigs can be, especially with a router. While not everyone may have room for or need a table saw, a router and router table are relatively inexpensive and allow you to perform many operations quickly that would take a long time by hand. Happy building!

Quick Box

continued from page 63

is a program I will use on a regular basis; definitely recommended.

Bill Fitzpatrick of Sitting Duck Software replies:

You can design either a closed or vented box after entering data for the driver. The driver need not be part of the library. The problem Mr. Koonce experienced was probably due to a 0 value for Z_{MAX}. All information for the driver had to be present in order to design an enclosure. The program had been modified to automatically substitute 999 for Z_{MAX} if 0 has been entered; this removes any potential problem.

Mr. Koonce is correct in his observation that the Resonance Impedance Correction formulas were incorrect. This has been corrected and any Quick Box users with a QBOX.EXE file dated earlier than December 1991 will receive a replacement disk if they send the original to me (Sitting Duck Software, PO Box 130, Veneta, OR 97487). I will send a corrected QBOX.EXE file to those who send in their registration cards.

The Standard B_4 alignment will be correct if the Q_T of the driver is near 0.4 but as the manual states, is to be considered a misalignment with other Q_T values.

Classified Advertising

TRADE

CUSTOM MADE CROSSOVERS, passive, active, audiophile tweeter protector, no active components in signal path, available for 4 or 8Ω or for odd impedances. For more info contact, ***CUSTOM SOUND***, Box 3853, Antioch, CA 94531-3853, (510) 706-1752. T4/92.

MYLAR FILM. ½ mil, genuine DuPont, 48" wide, $1 per running foot; also ESL transformers. ***ROGER SANDERS***, R1, Box 125, Halfway, OR 97834, (503) 742-5023. T4/92

MACH 1 ACOUSTICS introduces the TL100 and the TL2000 transmission line speakers. No risk in home auditions. TL100 MTM 2 6½" Bextrene midbass + 1 polymide dome, $1,500/pr; TL2000 MTM 2 6½" polyglass + 1 Accuton, $3,000/pr; SUBASS 500 Isobarik 10" subwoofers, $2,500/pr. Designed and tested with MLSSA, LEAP, and XOPT. ***MARK 1 ACOUSTICS***, RR2 Box 334A, Wilton, NH 03086, (603) 654-9826. T1/93.

OMNIDIRECTIONAL ADD-ON SUPERTWEETER 7 to beyond 20k, extremely smooth. ***G. MUELLER***, RR 3 Box 541, Glen Gardner, NJ 08826, (908) 537-6245. T3/93.

$$ I WANT IT ALL AND I WANT IT NOW —$$ — at a price I can afford! So Audio Research and Krell are pretty good, right? But also pretty expensive. Try the best of both worlds in a price-busting way! Convert your Hafler, B&K, Adcom, or other amp to TUBE hybrid design. Design by Greg Miller. (818) 982-6477 ***LAZARUS TUBE-HYBRID CONVERSIONS***. $799 T1/93.

BANDOR ALLOY DRIVERS now available in the USA. Call, ***BFW A/V SYSTEMS***, (415) 961-0380. T4/92.

CABINETS, KEVLAR DRIVERS, CROSSOVERS, etc. ***SPEAKERWORLD TECHNOLOGY***, (510) 490-5842, FAX (510) 490-1961. T2/93.

ORIGINAL QUALITY refoaming/reconing, (412) 375-9203. T1/93.

SPEAKER RECONING. All makes, models, stereo and professional. Fast service. Kits available, low rates. Refoaming $18. ***ATLANTA AUDIO LABS***, (800) 568-6971. T5/92

PREMIUM GRADE PARTS for modifications and upgrades, MIT/Cardas chassis wires for wiring speakers/electronics. MIT MultiCap, Solen cap, RelCap, WonderCap/ wire/solder products. Holco, Caddock, Resista, Vishay resistors and trim pots. Edison Price, WBT, Vampire, Cardas, etc. connectors. RAM tubes, silver contact toggle and rotary switches. Wide selection of parts and accessories, prompt service, catalog available. ***MICHAEL PERCY***, Box 526, Inverness, CA 94937, (415) 669-7181, FAX (415) 669-7558. T6/92.

SPEAKER BUILDER CLASSIFIED ORDER FORM

PLEASE FILL OUT IN CLEAR BLOCK LETTERS OR TYPE. PLEASE SPELL OUT ENTIRE WORD.

1	2	3	4	5	6	7	8	9	10
11	12	13	14	15	16	17	18	19	20
21	22	23	24	25	26	27	28	29	30
31	32	33	34	35	36	37	38	39	40
41	42	43	44	45	46	47	48	49	50

☐ Private For Sale ☐ Private Wanted ☐ If Trade, please indicate # of insertions ________

Private For Sale ads are for *non-profit* use. Individuals selling items for retail/profit should be using **Trade** ads. **Private For Sale** and **Private Wanted** ads are on a one-time insertion basis. Ads **must** be resubmitted for renewal. **Ads will not be taken over the phone. ILLEGIBLE ADS WILL BE DISCARDED.**

Definition of a word: A series of letters with a space before and after. Count words, not letters. Any words over 50, 20¢ per word. Trade ads are 90¢ per word and indicate number of insertions. Deduct 10% for a 6× contract in *Speaker Builder*. **Payment MUST accompany ad. No billing.**

Please remember to include your name, address and telephone number. In order for Private For Sale and Want ads to be **free**, you must be a current subscriber.

Please charge to my MasterCard/Visa:
Ten dollar minimum for charge card orders.

CARD NUMBER ________ EXP. DATE ________

☐ *Check/Money Order Enclosed* ☐☐☐☐☐☐ Magic No.

NAME ________

PHONE ________

COMPANY ________

STREET & NO. ________

CITY ________ ST ____ ZIP ____

WOOFERS REFOAMED to original specifications. Fast and inexpensive, guaranteed, professional work. Not a kit. Send stamped envelope for full details. ***NEVISONICS***, 650-B Baxter Ave., Louisville, KY 40204. T3/92.

DESIGN & INSTALL
YOUR OWN CUSTOMM A/V OR MULTIROOM SYSTEM with our informative cataloge of installation supplies. Complete Systems, Parts, Text, Diagrams, & Full Technical Support—Cataloge $3.00. R. Gladwin, P.O. Box, 221666, Carmel CA, 92922 or Call: 510-655-6653.

LOUDSPEAKER COMPONENTS: Many of our extraordinary crossover components have never been offered to the home builder before, yet they are the foundation of some of the world's finest loudspeaker systems. We feature Ohmite's best precision wirewound silicone-ceramic power resistors, magnificent polypropylene capacitors from Sprague and ASC, custom inductors, and a variety of other high-end necessities. Each of our exceptional products is chosen strictly on the basis of sonic quality, and although cost is no object our prices tend to be as good or better than our competitors! A subscription to our quarterly catalog is $3. A complimentary issue may be be requested. ***NORTH CREEK MUSIC SYSTEMS***, Route 8, Speculator, NY 12164, (518) 548-3623. T4/92.

CUSTOM MADE POLYACOUSTIC FOAM speaker grilles—any size, thickness, design, color, or quantity. ***CUSTOM SOUND***, PO Box 463026, Mt. Clemens, MI 48046, (313) 463-5039 or Fax (313) 463-1946. T4/92.

FOR SALE

Phase Linear 200 amplifier, 105W/channel, 2000 preamp, 5000 tuner, $600; Thorens TD125 turntable with Shure V-15V, $150. Larry Evers, Airport Rd., RR 4, Lincoln, IL 62656, (217) 732-6486.

JBL midrange horns, 500Hz with Acoustic Lens with model 375 driver, magnet assembly weighs 30 pounds, these have upgraded Beryllium diaphragms to extend HF response, included are aluminum diaphragms which are flat to 10kHz, $300 each in pairs. Charles Goepel, 181 Berrywood Dr., Severna Park, MD 21146, (410) 647-9135.

JBL L100 Century three-way speakers, very good condition, collector item, $375/pair. Tony Cheung, 805 Camelot Dr., Salisbury, MD 21801, (410) 749-2771 after 5:30 p.m. or leave message.

CASP'er
Computer Aided Sheet Planner
Panel Cutting Software
Optimizes, Files, Prints Layouts, Lists, Labels and More! **$249 + S&H**
ROGER DRUMMOND
10 Pike Street • Herminie, PA 15637
Phone/Fax (412) 446-0159

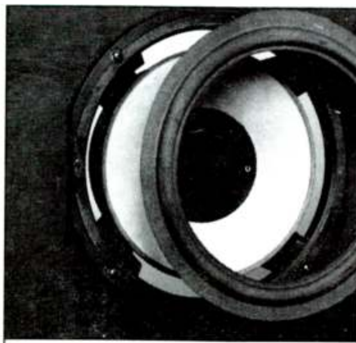
Pioneer A339 40W/channel amp, $185; PD7700 CD player, $250; JVC XL2-441 CD player, $155; Baby Advent IIs $145/pair; Shure ME97HE cartridge, unused in box, $45; (all less than 4 months old). Steve, (703) 463-5675 7:30-9:30 p.m. EST.

Kinergetics KBA-75 Class A power amp, 75W/channel, Gold Series, excellent condition, $700. Paul, (914) 528-1510.

Pair Morel MDT-33, $65; Mallory 50V DC 51,000µF, $6 each; Mepco 50V DC 12,000µF, $2 each; Sprague 50V DC 15,000µF, $2 each; Sangamo 50V DC 12,000µF, $2 each; Mallory 50V DC 12,000µF, $2 each; all used but working. Chris, (314) 993-1603 before 10 p.m. CST.

Ramsa power amp, balanced, 200W/channel, $550; Crown DC-300A-II, $425; two Altec 1270 stereo power amps, 200/400W/channel, 800W 8Ω mono each, $450 each; DB Systems rackmount preamp with tone controls, $400; Eumig preamp, $175; Tascam 122 cassette deck, $450; Teac 25000, $550. David, (914) 688-5024.

Panasonic Leaf tweeter, EAS-10TH400A, $60/pair; Stax Gamma Pro earphone with SRD-7 adapter, $350. Brian, (213) 732-4532.



Save hundreds of dollars on your speaker repairs.
A highly innovative speaker surround system designed for you is now available. All sizes. Easily installed & affordably priced at $27.95/pr. (incl. s/h). No more costly repairs/freight. You can replace ruined surrounds yourself with the SAT system. ORDER NOW. TOLL FREE 1-800-747-3692.
STEPP AUDIO TECHNOLOGIES
PO BOX 1088, Flat Rock, NC 28731
VISA/MC/Check/M.O. • 1-800-747-3692

NEC AV-20 surround, $100; vintage KEF speakers, $200; Pioneer CD-700 disc, $75; Infinity speakers, $100; AKG and Azden phones, $50 each; Pioneer V5X 5000 tuner amp, $100; Advent sound control, $150; three Marantz amp units, $100 each; everything in good shape. Ed, (305) 891-2267.

Technics SLPC-20, $175; Carver C-4000, $300; Phase Linear 4000SII, $175; Earthquake MD-5 cast mids, $140 retail, unused, $85/pair; ADC PSX-30 phono cartridge, $150 retail, new, $35; Audio Technica PC550ML cartridge, $300 retail, new, $75; Apature interconnects, $200 retail, 1+2 meter, $85; Apature interconnects, $90 retail. Andy, (414) 458-2057.

Topaz Ultra Isolator power line isolation transformer, 2.2KVA, 146dB common mode, 60dB normal mode rejection, less than 1% harmonic distortion, $350; Mogami Neglex 2534, $55/100'; back issues of *Audio Amateur* and *Speaker Builder*; people wanted to share bulk order of Jordan drivers. John, (703) 425-7482.

Pair Audio Control C-131 ⅓-octave equalizers, mint $750; Magnavox CDB-650 CD player, $125; older pair Accuton C2-11S, $150; newer pair Accuton C2-11S, purhcased 12/91, $250; four Focal 5K13-LS, $30 each; pair Swan Leda crossovers, $40. Leigh, 13306 SE 9th Pl., Gainesville, FL 32601, (904) 378-7485.

Two pair Dynaudio 30W-100s, 4Ω version, Fs-19Hz, Qts-0.41, 4" voice coil, 91dB @ 1W, excellent condition, $150 each; prices firm including shipping. Paul Champlin, 1016 Quail Gardens Ct., Encinitas, CA 92024, (619) 632-7463.

Fine silver 20-gauge multi-strand, Teflon insulated, new, outstanding, $5.65/ft; toroidal transformers, 2 × 20VAC/250VA, small, new, $20 each; nice 675VA toroidal, 2 × 40VAC, new, eight, $80 each; Hitachi MOSFETs 2SK134/J49, new, matched, $5 each; some 2SK176/J56, new, $12 each. Randy, PO Box 144, Hannacroix, NY 12087, (518) 756-9894 after 6 p.m.

Dynaudio D54, $40; Intel 8087 math coprocessor, $40; Hafler DH-110, $115; AudioQuest Indigo, 10 foot/pair, $45; Gauss Bullet tweeters, 111dB @ 1W/M, (loud), $75. Dan Patten, (801) 225-8577 home, (801) 224-8080 work.

Paradigm 7se MK II speakers, oak vinyl finish, Premier W-30 stands, bought four months ago, excellent condition, $490. Northwest Ohio, Ron, (216) 677-0913 leave message.

KEF B-139 SP1044, mint, $200/pair; nice walnut 2ft³ cabinets with black grille cloth, $100/pair; Realistic 2 × 6" horn Piezo tweeters, catalog #40-1379, $25/pair; Richard Allen HP12B 12" woofer, cast frame, $40; smaller Advent loudspeaker, $40. Jerry Feldstein, (212) 364-3485 evenings, NYC.

Carver CT-7 AM/FM sonic holography tuner/preamp, 9 memory presets, remote control, A.C.C.D., multipath noise reduction, video input tape dubbing, $350; SAE A202 dual high resolution power amp, 100W/channel 8Ω, 20Hz–20kHz to −0.5dB S/N ratio −110dB, $250 or $550 both. Mr. Willes, (800) 227-3390 weekdays.

Eton 1" metal dome tweeters 25DTF250, new $120/pair, sell $65/pair postpaid. Alan Nettleton, (302) 764-7479.

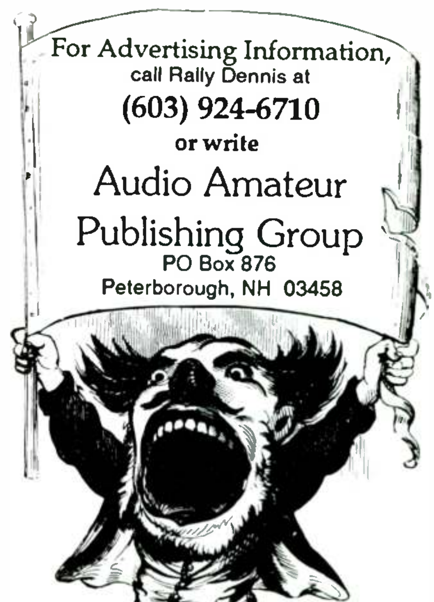
Spectral DMC-10 Alpha, $1,000; mint matched Focal 5" 5K013Ls and VIFA 1" D25AG-35s with grilles, $175 or best offer; JBL 2380As, $200/pair; *Stereophile* 1991 and more, $25; Lazarus H-1 amp, $700 or best offer. Steve, (203) 397-3888.

Test equipment, SWTPC multi-waveform audio generator, 5Hz-100kHz, works fine, $35; Heathkit IM2410 frequency counter, 10Hz–225MHz, excellent, $100; trades. Andy, (303) 781-5573.

Audio Authority 431-432 speaker switcher with level matching, up to 32 pair, $225; Shure SM-80 Omni condenser microphones, $375/pair; Focal 233 speakers, 8", two-way Kevlar with oak cabinets, $425/pair. Paul Hughes, (707) 542-4406.

Futterman tubes, twenty new 6LF6, seven 6LU8, $425; Dynaco wooden sleeve for preamp/tuner, factory original, new in box, $95; Harman-Kardon Citation IV preamp, excellent, $95; NYAL Minuet in a preamp, missing faceplate, lid works fine, $195; Sumiko Premier MMT tonearm, like new/box, $140; KEF B139 woofers, like new, $175/pair. Paul, 502-9857 Manchester Dr., Burnaby BC, V3N 4P5 Canada, (604) 421-4692.

Ampex 601-2 UHF portable stereo tube tape deck, carrying case, manual, mint, $650; Thorens TD150 MK II, armless, large solid Teak base, undrilled armboard, new $90, ³⁄₁₆" thick clear dust cover, fantastic bearing and motor, superb sound, barely used, pristine condition, $145; Dynaco PAS-2, excellent, $75. (703) 578-4929.



Curcio Daniel preamp in C-J PV1 chassis, will also include original PV-1 boards, needs retubing, $300; NAD 1300 preamp, one year old, *Stereophile* Class C, very quiet, good phono section, $275; Fisher tube, integrated amp built from kit, needs work, $50. Ted Kastelic, 5576 Glenoak Ct., San Jose, CA 95129, (408) 446-3417.

Luxman T-117 tuner, "one of the best sounding tuners ever, extremely sensitive and very low distortion," *Stereophile*, just checked and aligned, mint, if you listen to FM why not optimize the experience? $350 or best offer. Jim, (203) 739-2869 anytime.

Soundcraftsmen PM840 amplifier, 205W/channel, excellent condition, signal-to-noise 106, MOSFET output stage circuitry, no current limiting circuitry, bridgeable to mono for single subwoofer, original manual, $225. Bill Tidwell, (702) 385-7170.

Webster antique wire recorder with some spools, $50; Hafler DH-100 preamp, new, $125; Premier MMT tonearm, new, best offer; Pickering V-15 AT2, old but unused, $5; oak stereo cabinets, beautiful finish, priced very reasonably, local only. Walt Fleming, 182 Plain Hill Rd., Norwich, CT 06360, (203) 889-5018 5 p.m. best.

Yamaha T7 Analog FM stereo tuner, motor drive, 5 pre-selects, $150; Sanyo Plus 55 series stereo preamp, $100; all plus shipping. S.H. Grycz, 2935 Crehore St., Lorain, OH 44052, (216) 288-9480.

WANTED

Technics SB 7070 speaker systems. Cabinet must be in good condition. Margherita, (904) 392-0622.

JBL 075, 2405 or similar Piezo tweeters. Steve, (203) 397-3888.

Audio Control SA-3050A spectrum analyzer. Paul Hughes, 425 Lincoln St., Santa Rosa, CA 95401, (707) 542-4406.

Regency HF 50-A amp plus owners manual or schematic, two Dyna MK II amps; Output transformer for Fisher 500C receiver; Sherwood mode A3MX multiplex adapter. S.H. Grycz, 2935 Crehore St., Lorain, OH 44052, (216) 288-9480.

Two pair Jordan modules. Larry Evers, (217) 732-6486.

Parasound C/PT 1000R or Adcom 575 preamp tuner, pair 5K013Ls, four AC12 woofers, schematics with component values for Sub I crossover network, driver parameters for University C15W. J. Annal, (708) 425-6719.

Sony PCM-F1 or Nakamichi DMP-100 PCM processors. Gene Clough, PO Box 36, Laveen, AZ 85339.

Any electrostatic, magnetic planar or ribbon speakers, working or not; any Class A solid-state power amps, working or not. Paul, 502-9857 Manchester Dr., Burnaby BC, V3N 4P5 Canada, (604) 421-4692.

I am building a WAMM system and would be interested in hearing from anyone who has information on the individual drivers used or crossover design in Wilson's latest version. Paul Champlin, 1016 Quail Gardens Ct., Encinitas, CA 92024, (619) 632-7463 home, evenings.

Wharfedale Diamond II loudspeaker, must be in good condition, walnut finish, call or write with price, details. Walt Fleming, 182 Plain Hill Rd., Norwich, CT 06360, (203) 889-5018 5 p.m. best.

Sony TC-880-2, ¼" tape recorder, second choices are TC-854-2 or TC-756-2 or Telefunken M12. Pierre Sprey, (301) 627-0525 call collect.

Looking for Altec-Lansing model 411-8A 15" low-frequency speaker, got one of these you no longer need? Prefer single working unit, will consider pair or damaged unit. George (703) 434-5965, Ext. 274 days.

One A431 output transformer or single MK II, for home brew project. Don May, 129 Glenridge Rd., East Aurora, NY 14052, (716) 652-6107 EST.

CLUBS

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.

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.

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.

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! Toobers, Tranzzeestors, 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!

GREAT BASIN AUDIO SOCIETY. A club for those interested in better recording and reproduction of music, who live in Northern Nevada and surrounding parts of California, Idaho and Utah. Everyone is welcome to come to our monthly meetings, held in Reno on the last Sunday of each month. We also seek contributions to our quarterly newsletter, *The Singing Sagebrush*. For more info, contact the GBAS, c/o E.A. Barbour, 552 N. McCarran, #284, Sparks, NV 89431 or leave a message on (702) 358-2019.

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.

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

PIEDMONT AUDIO SOCIETY Audio club in the Raleigh-Durham-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.

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 Refrence* Newsletter. For information write or call, Frank Manique, President, 1219 Fulbright Ave., Redlands, CA 92373. (714) 793-9209.

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.

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.

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.

SPEAKER BUILDERS/AUDIOPHILES in the Milwaukee, WI area. I am a speaker builder with test gear looking to join or form an audio club. Anyone interested contact, Kirk Rontti, (414) 355-7509 leave message.

LOS ANGELES AREA ARIA AND SWAN Loudspeakers Users Group. If you have built or plan to build any of the Aria or Swan loudspeakers and would like to meet for listening sessions or have any questions or modifications that you would like to share, please call Geoffrey, (213) 965-9173 or Edward, (310) 395-5196.

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

THE ATLANTA AUDIO SOCIETY is dedicated to furnish 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 Denny Meeker, (404) 872-0428, or write: PO Box 361, Marietta, GA 30061.

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.

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.

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.

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.

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.

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.

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-B-ing equipment. New members welcome. Contact Bill Donnally, (201) 334-9412, RD2, Box 69D, Miller Dr., Boonton, NJ 07005; or contact Bob Young, (908) 381-6269, or Bob Clark, (908) 647-0194.

THE BOSTON AUDIO SOCIETY invites you to join and receive the bi-monthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses, and summaries of lectures by major engineers. Read about Apogee, Nytal, Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

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 bi-monthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212, or call Art Tedeschi, (303) 477-5223.

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.

THE OREGON TRIODE SOCIETY We are dedicated to the art and craft of music, audio DIY projects, and quality sound reproduction. Our 125 + members meet 8 times a year in the Portland area and our news magazine, *Positive Feedback*, has grown to 48 pages of challenging commentary, fun and information and is published six times annually. Ladies and gentlemen, you are cordially invited to join us. For information contact, David Robinson, 4106 N.E. Glisan, Portland, OR 97232, (503) 235-9068 or Ian Joel, (503) 233-1079.

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.

Advertising Index

Moran in the Market

DEEP-BASS TROLLING

By David R. Moran

Separately and together, the three or so octaves we call bass—from somewhere below 20Hz up to, say, 150Hz—are often misidentified or mischaracterized.

The upper part of this range, if boosted, can thicken male voice and all other sounds that are "thickenable." Rock bass (except for kick drum at approximately 45Hz) lies lower, in the 80–110Hz range, and can sound inexplicably deep, especially over a car-stereo system. Aggregate rock bass reaching its loudest levels below 80–90Hz is not common, however.

The octave below that point, down to 40Hz, is what should be identified simply as bass. Indeed, after he listened to the subwoofered Snells and then looked at a measurement of their response (see discussion and top graph of *Fig. 1, SB* 2/92, p. 79), showing a room-related valley between 30 and 125Hz which includes dips at 50 and 100Hz, speaker researcher Floyd Toole observed, "Hmm, I *thought* they lacked bass." (No, I am not going to drop his name every column, I promise.) When I countered with "But they have tons of it," he returned, "Low bass, yes, but I didn't hear much bass."

The soul-shaking crack of a bass drum, as in a large orchestral work or a holiday parade, which up close can sound like the end of time, has its fundamental somewhere around or just below 40Hz. (You'd swear it was DC, though, or at least 5-15Hz.) Note furthermore that the scary and thrilling part of this concussion—its crackly leading edge—lies well into the treble. Most of the time, then, you must have good tweeters to have superb bass sonics. Such a loudspeaker system, if it can also deliver 35–40Hz clean and loud, is going to sound as if it has a superlative—and very low—low end.

The majority of pipe organs (including monstrously sized ones) do not have truly low bass, i.e., below 35Hz, even when they sound thunderous and growly and have huge, towering pipes. Many produce little loud output below 50Hz. And in the rare instruments that do play the really shuddery stuff, it's sometimes produced by a woofer playing tones, not a pipe! Different centuries, different motors.

The octave below 35–40Hz is simply of another character. It sounds shuddery sometimes, more often just like thuds and thumps. Footfalls, heavy damped stage doors closing, steps and stomps, slowly beating helicopter blades: all such information lies in the realm of ±20Hz. If you want to hear it right now, stamp your foot on a wooden floor, or kick a couch, desk, table, or similar large heavy object with your shod heel. Such sounds, when continuous and loud enough and rolling through a cavernous cathedral, are indeed impressive: chest- and gut-shuddering.

Low Concoctions

From the very presence of bass overtones or harmonics, the human ear sometimes is able to make up the missing fundamental, so when listening to 4″ or 5″ "woofers" or 6″ modular "subwoofers" we can concoct lower bass. When actually there, on the other hand, bass levels have to be healthy, owing to the ear's progressive insensitivity. We are quite deaf to quiet low bass and especially to quiet very low bass; for instance, we don't hear weather fronts. But also note that once thresholds are reached, a little change, a very few dB, may go a long way as far as the ear is concerned, and may have surprising impact and consequences.

The cleaner and clearer the fundamentals—the lower the distortion, in other words—the more bass we sense and hear, as the true pitch comes through. So now let's turn to subwoofers, whose job this is. By the term subwoofer I refer to real ones, not the boxes that come with three-piece systems. I mean ones that, playing the right CDs and broadcasts, provide "startlement," which may take the form of head-turning (as you try to figure out if that unlocalizable thump was the dog turning on his side, your child falling out of bed, or something in the basement), or jaw-dropping, or head-shaking. These are subwoofer experiences that make a difference you subsequently don't want to go without.

Low Reproductions

In the latter 1970s I learned first-hand what proper equalization with infrasonic filtering (not, please, "subsonic") does for bass playback. The Allison Electronic Subwoofer (ESW), designed by then-freelance engineer Mark Davis to make various Allison loudspeakers flat to 20Hz, with steep skirts below that point and above 20kHz, worked superbly with any robust acoustic-suspension design. Still does, in fact. It was an absurdly precise, hard-to-demonstrate, and somewhat expensive piece of gear, but it showed what happened and could happen below 40–50Hz—given the rumbly sources of the time. (As regular readers might suppose, I believe any phase distortions due to infrasonic filtering are audibly unimportant to the ear.)

Ten years after the ESW, in the later 80s, I got to hear and review my first Velodyne subwoofer, a powered, distortion-canceling, equalized design (their 12-incher in this case). What a powerful treat that was. It performed just as claimed, and is an admirable achievement of clever engineering. When it played back some quiet 19Hz

TABLE 1

50Hz TONE

Four 10″ woofers with inductor:	
Loud level (0dB)	3% 2nd and 3rd HD
Louder level (+8dB)	9% 3rd HD, 2nd harmonic much lower
Loudest level (+9dB)	10% 3rd, 2nd harmonic much lower
Single DR SW10 subwoofer:	
Loud level (0dB)	2% 2nd and 3rd HD
Loudest level (+9dB)	6% 3rd, 2nd harmonic lower

TABLE 2

24Hz TONE

Four 10″ woofers with inductor:	
Loud level (−3dB)	5% 3rd HD, 2nd harmonic lower
Louder level (−2dB)	10% 3rd HD, 2nd harmonic much lower
Loudest level (0dB)	30% 3rd, 2nd harmonic much lower
Single DR SW10 subwoofer:	
Louder level (+2dB)	3% 2nd and 3rd HD
Loudest level (+3dB)	4% 3rd, 2nd harmonic lower
Max level (+4dB)	7% 2nd and 3rd HD

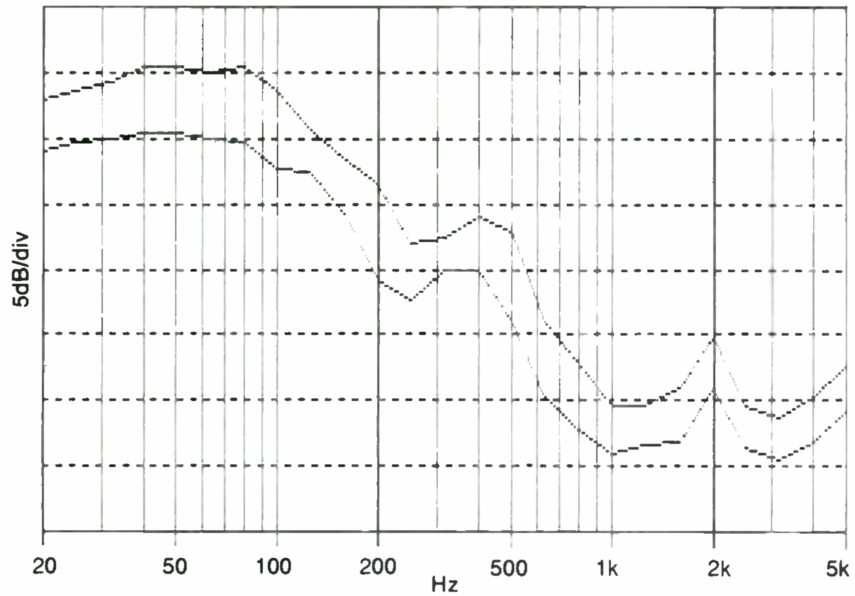


FIGURE 1: The ⅓-octave frequency response of a single Definitive Research SW10 subwoofer system measured outdoors, taken at two different angles. Response above 40Hz is affected by the 20kΩ input impedance of the amplifier used (DR recommends 50kΩ; see text). Plotted from 20Hz–5kHz at 5dB/division.

organ pedal, I remember that my dog, Woofer, was certain there was something hovering overhead outside the house.

Like many audiophiles for the last couple of years I've also been awaiting the advent of the Allison SW20, with twin 10″ woofers, inversely mounted to achieve the hard-to-surpass low-distortion low-frequency performance of the company's IC20 loudspeaker.

Although larger and more expensive subwoofer designs abound in the marketplace, I habitually focus my interest on such as these, which are both affordable and of a size which a listening room and a non-audiophile spouse can accommodate with the least upset.

Cylindrical Sounds

Into this educational journey through the realm of very low bass comes now the Definitive Research SW10 (20013 Rainbow Way, Cerritos, CA 90701, (800) 554-0150). How very nice that this new product, based entirely on simple, straightforward science, humbly conceived, designed, executed, and fairly priced, does what it claims: delivers clean low bass at high levels. The DR SW10 comprises two separate units in the form of rigid, vented cylinders, made of recycled-paper laminate with MDF end pieces, each cylinder containing one ultra-long-throw (± 10mm, or more than ¾″) 10″ woofer. Cabinet diameter is 14.5″ and height is 27″, with four thin 2.5-inch bolts to screw into the base and create each cylinder's stand. Plain-looking, the tube is covered in black knit cloth, and the standard cabinet top is matte black with walnut trim, although various dressier wood-veneer tops are available at extra cost.

The DR SW10 pair includes a required stereo (two-channel) passive speaker-to-line-level equalizer module, which you drive in parallel from your current speakers' amp and then feed the output to a separate 40–300W/channel, 50kΩ input-

Continued on page 94

DON'T COLOR YOUR SOUND

FIBERGLASS & WOVEN CARBON DRIVERS BY VERSA-TRONICS PRESENTS TRUE UNCOLORED REPRODUCTION

Only Versa-Tronics Presents:

- 4", 5-¼", 6-½", 8", 10" and 12" Woofers
- Die Cast Alloy Baskets
- Rubber Surrounds
- Aluminum & Kapton Voice Coil Formers
- Excellent Damping
- European Cone Materials
- Polymer Dust Caps
- Extremely light cone material

Available at the following distributors:

Just Speakers
3170 23rd St.
San Francisco, CA 94110
(415) 641-9228

Speakers, Etc.
1828 W. Peoria
Phoenix, AZ 85029
(602) 944-1878

Toutant Electronics
246 E. 131st Street
Cleveland, OH 44108
(216) 451-6325

Speaker City
10615 Vanowen
Burbank, CA 91505
(818) 508-1908

Speaker Works
2539 E. Indian School Rd.
Phoenix, AZ 85016
(602) 956-6344

ITC Electronics
2772 W. Olympic Blvd.
Los Angeles, CA 90006
(213) 388-0621

Speaker Center
10 Bellam
San Rafael, CA 94901
(415) 459-2213

Versa Tronics

VERSA-TRONICS INC.
Amherst, NH 03031

Continued from page 92

impedance, dedicated power amp (preferably with gain controls) for driving the SW10s. Obviously in this kind of design it is essential to observe ground hygiene. The EQ box has a benign input impedance of 400Ω. Note there is no highpassing crossover anywhere in the system to relieve bass strain on your main speakers; this is an augmentative subwoofer system only.

Price for the DR SW10 pair in standard finish is $480 (Eastern US) or $500 (Western). Quite the bargain, I would judge, for something that so entirely fulfills its promise. A bass-potent CD and a thorough instruction manual are included. The product is sold factory-direct, with a 30-day money-back guarantee and a five-year manufacturing-defect warranty. Other models from Definitive Research include the SW12, a slightly thinner, vented 7′ cylinder with a 12″ subwoofer, and the frightening-sounding InfraBass Reference.

The SW10 2π frequency response is said to be 20–40Hz ± 1dB when used with an amp of the suggested input impedance (see measurements). Other bandwidth, crossover, and EQ configurations are available, most at additional cost. Minimum loudspeaker system impedance is specified as 7Ω.

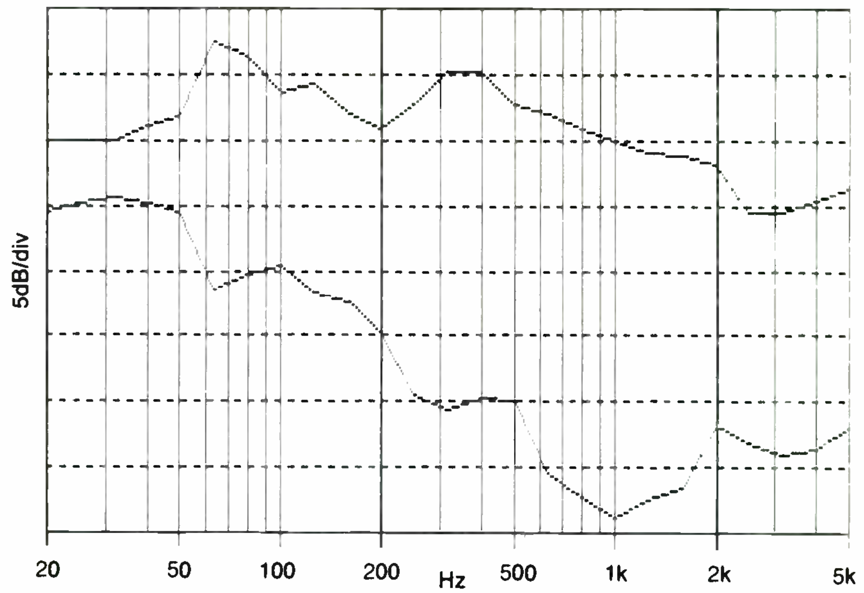


FIGURE 2: The top is the unequalized response of a single DR SW10, measured as in *Fig. 1*; below, is the derived response curve of the equalization module.

Listening Session

There is not much to report, really. The DR SW10s worked just splendidly, ably producing clean, loud rumbly tones when such was the input, without requiring a humongous amp. Good show, I wrote in my notes. Get out your "Rite of Spring." The subwoofers were able to take plenty of power, too. I sited them both in and out of corners, with the expected differences of room gain (loudness level). I was very pleased to learn they can also be used on their side—don't stick the woofer end right into a corner, naturally—which means greater versatility for less-obtrusive placement behind or beneath furnishings. After the audition I was eager to get to measurements. (Shouldn't I be sprinkling this paragraph with amusing sonic descriptions involving "transient speed," "neutrality," "fast response," "uncolored blending," and so on?)

Frequency Response

Figure 1 shows the gratifyingly flat ⅓-octave frequency response of a single SW10 system outdoors, taken at two different angles (vent side and nonvent side, there being no axis to speak of). It meets spec, to put it mildly, and the result speaks for itself: ± 1dB or so from 20–40Hz, and extending with the nonrecommended amp I used (see below) flat to 90Hz. The usual test protocols were observed: precision flat pink noise, continuous spatial averaging with the AKG microphone 7–9′ away, 32–42″ high (seated ear height), and ± 10–15° horizontally.

It is not easy to measure reliably outdoors under 80Hz even in a large quiet field: it's hard to hear what's going on and hard to get signal 10dB above the noise without overdriving. I managed, though. Note the graph in *Fig. 1* necessarily ends at 5kHz. Also note that the dbx RTA-1 realtime analyzer/PC does not measure much below 20Hz, so I cannot comment about the SW10's likely good performance for the last few infra-audio hertz, the range where the vent's own slug of air is what's hitting (vibrating) the air and producing the response. The manufacturer's specified −3dB point lies below 20Hz. At the other end, with the amps I was using, which have input impedances in the 20kΩ range, the response extends an octave or so higher than the specified 40Hz. This is how I would want to order the system anyway (why chop off so much of the woofer's output?). If you would be concerned about this extended out-of-spec behavior, and, for example, already have enough 60–100Hz from your main speakers, the input impedance of the amp for the DR SW10s is something to pay attention to.

Figure 2 (top) shows the raw or unequalized response of the SW10, and below it is the derived response curve of the little equalization box, which uses a simple RC network to knock down the natural 60–70Hz resonance. The lack of an inductor anywhere in the system must help contribute to the low distortion of the sound.

Harmonic Distortion

Harmonic distortion was measured playing tones in a listening room, with no EQ module in line. The power amp stayed well below clipping even at the highest input levels, but the room sure buzzed. Again the extraordinary results spoke for themselves, although I did feel making a comparison was important. So a single DR SW10 subwoofer in its cabinet was measured against a single loudspeaker system cabinet containing four beefy, entirely respectable 10″ Culver Tonegen woofers fed by a phased-array (non-paralleling) crossover with inductors.

For each system the front- and side-wall augmentation was identical. Distance to the floor was 4″ for the DR SW10 and approximately 10″ for the Culver Tonegen system, so boosting of the bass levels at these wavelengths by the floor was slightly different. Mike position was identical in each situation. All figures are rounded off. For this setup, the 0dB listed happened to equal 87dB SPL, but that's a meaningless figure, i.e., nothing can be extrapolated or inferred from it. (For these frequencies, a pair of speakers is going to give 4–6dB more output in most rooms with most stereo spacings, anyway.)

Even if one gives these data the most charitable interpretation, for example taking into account the conventional woofers' poor saturating inductors and the slight extra gain from the floor augmentation for the vented, cylindrically cabineted subwoofer, it seems clear that one DR SW10 subwoofer is *at least* twice as "good" as a conventional woofer.

Nice job, Definitive Research, nice subwoofer. When I have more means, I'm probably gonna get me some.

Definitive Research has announced a forthcoming company name change. In addition, the company will no longer offer the SW10 with the black top and walnut edges. Surfaces offered include walnut ($530), oak ($580), and rosewood ($630). Add $20 each for level control and an additional $30 each for 18dB/oct. Shipping charges are extra.—Ed.

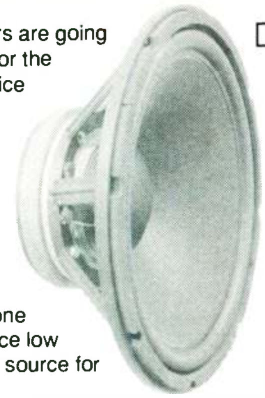
Parts Express International Inc.

INTRODUCES

BBC SPEAKERS

BBC Speakers, founded in 1945, is dedicated to the design and manufacture of superior loudspeakers and exponential horns. BBC bases its production on the work of a small number of collaborators who are specialized in the field of audio. The result is speakers of uncompromising excellence known throughout Europe. One look and you will see that BBC is dedicated to advancing technological standards in the manufacture of loudspeakers worldwide. For loudspeaker components that will be state of the art in the 21st century, check into BBC Speakers at Parts Express.

3C DUAL VOICE COIL SUBWOOFERS



BBC SPEAKERS

iese cast frame subwoofers are going set worldwide standards for the dustry. Large diameter voice ils and heavy magnet uctures combine to eate subwoofers that e capable of handling e high powered nplifiers that are pular today. Flexible iders and foam sur- unds provide maximum cone ovement to easily reproduce low ss. Parts Express is your source for 3C subwoofers.

3C COMPRESSION DRIVERS

BBC SPEAKERS

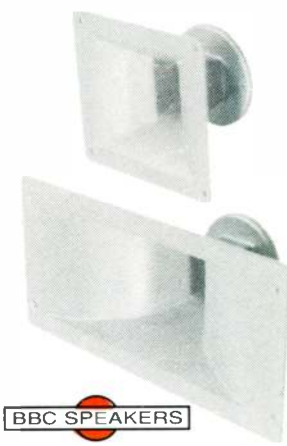
C compression horns e built to handle the ugh treatment and h power levels of blic address and ge monitor use, but eir outstanding tonal ality also makes them itable for use in studio onitors. These mpression drivers will on be in demand by all knowledgeable stereo enthusiasts. Parts Express s these rugged compression drivers now.

BBC TWEETERS

BBC SPEAKERS

BBC manufactures extremely efficient tweeters for use in studio monitors and home stereo speakers. Features such as mylar film diaphragms and cast aluminum frames set these Italian made tweeters apart from all others. The high sound pressure levels that these tweeters produce with minimum input is simply amazing. Shop Parts Express for BBC tweeters.

BBC CAST HORNS



BBC SPEAKERS

BBC high frequency horns are cast of aluminum for durability and good looks. Exponential design yields excellent horizontal and vertical off axis response. These superior quality horns are designed to complement BBC's bolt-on compression drivers. If you are looking for high frequency horns of uncompromising quality, look no further than Parts Express.

EMINENCE WOOFER

MINENCE

Eminence is known worldwide for their musical instrument speakers. Now Eminence applies their years of experience to manufacturing high quality replacement home stereo speakers. Ribbed paper cones, vented pole pieces and extended bump plates are just a few of the outstanding features incorporated into these exceptional speakers. Eminence speakers are available from Parts Express for as little as $23.50.

PYLE MIDRANGE HORN

PYLE

This well known midrange horn by the American manufacturer, Pyle, features a non-resonant, high density urethane horn body and a ferro fluid cooled voice coil. Built in phasing plug forms an acoustically accurate dispersion pattern. Famous Pyle quality midrange available from Parts Express for as low as $17.95.

POLYDAX MDA 100 MIDRANGE

polydax

Polydax combined all of their unique innovations into the design of the MDA 100 midrange, their first release in the new Prestige series. There has never been a more advanced midrange. TPX cone, Norsorex surround and phasing plug, pure titanium voice coil former, Supra magnet structure and more. A wide range of Polydax tweeters, midranges and woofers are available from Parts Express at very competitive prices.

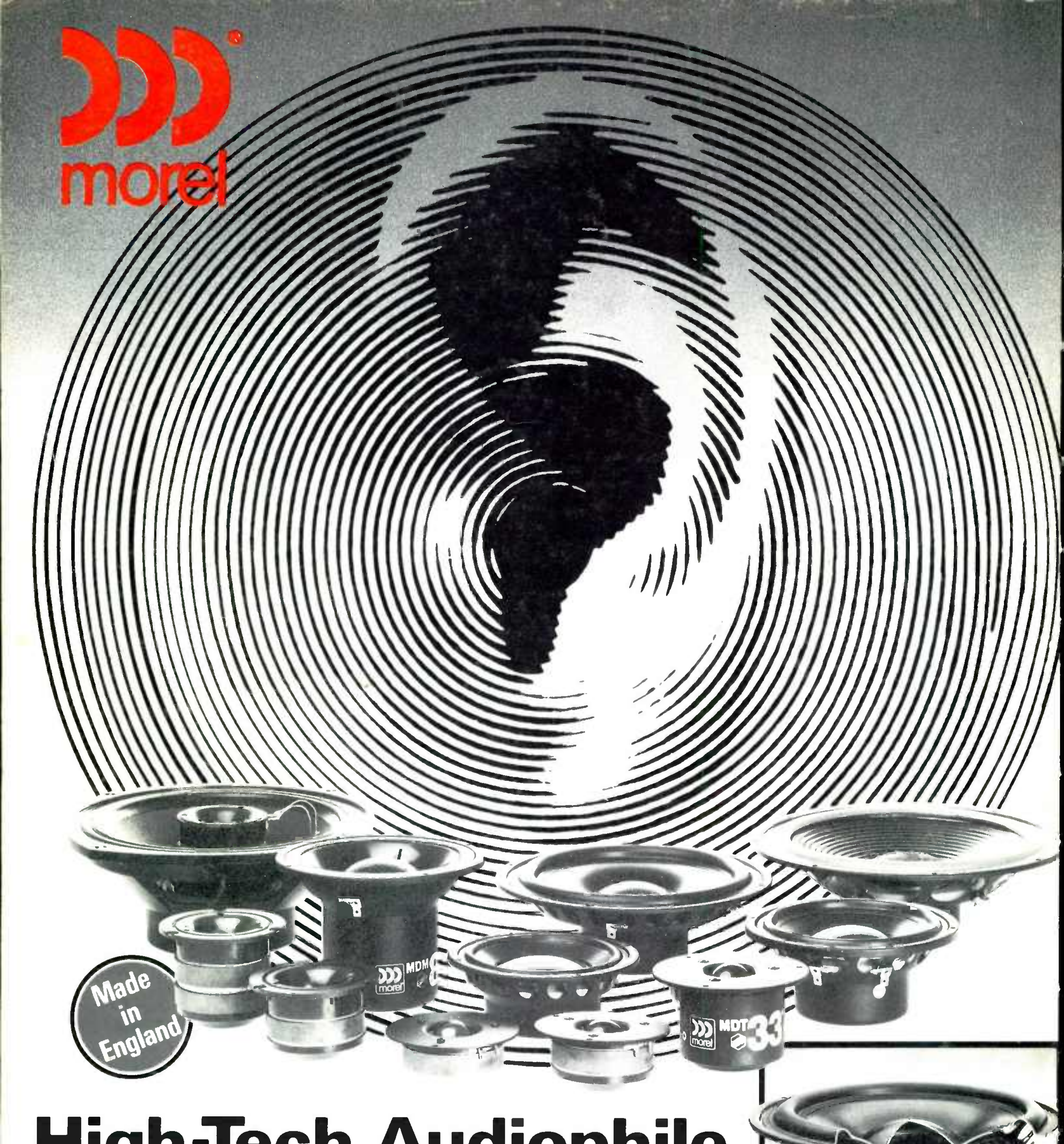
OLYDAX WOOFER/MIDRANGE

polydax

is audiophile quality 6-1/2" oofer/midrange has a long throw ice coil designed especially for low quency reproduction. Polydax's iovative Velbex rubber surround aterial allows for free cone movement d will outlast conventional foam rrounds. Parts Express has a large ection of Polydax woofers starting $24.90.

CALL FOR A FREE CATALOG 1-800-338-0531

Parts Express Int'l. Inc. 340 E. First St. Dayton, Ohio 45402



Typical Double Magnet Woofer Cross Section

High-Tech Audiophile Loudspeakers

For Further Information Please Contact:

morel acoustics USA
414 Harvard Street
Brookline, MA 02146
Tel: (617) 277-6663
FAX: (617) 277-2415

Fast Reply #GG142