WATCH DESIGN CHANGES AS YOU MAKE 'EM

FIVE: 1994

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Woofer Tester

Speaker Builder
THE LOUDSPEAKER JOURNAL

THE SYSTEM III
An Elegant Four-way Design
D. E. STENTON

CONQUERING IMPEDA
Speaker Drivers Aren't Resisto
VICTOR STAGGS

LINEAR ARRA
Searching for a Better Spatial Recording
PHILIP WITHAM

DICK CAMPBELL TRIES SOME RECIPES

USING BISCUITS FOR GREAT CORNER JOINTS

Good News

■ MIDRANGE DRIVER

ProSystems announces a new midrange 5¼″ compression driver designed for high-level studio monitoring and sound reinforcement. The APS-50DM has a frequency response of 1.3–6kHz (±1dB), and, through the use of a copper-clad aluminum voice coil and fully symmetrical magnetic field, claims extremely low second- and third-order distortion. It will handle 40W RMS/300W peak, while delivering the high efficiency (110dB 1W/1m) that today's more sophisticated crossovers require. ProSystems, 65–36th Street, Wheeling, WV 26003, (304) 233-2223, FAX (304) 233-2258.

Reader Service #101

➲ TWO UPGRADES

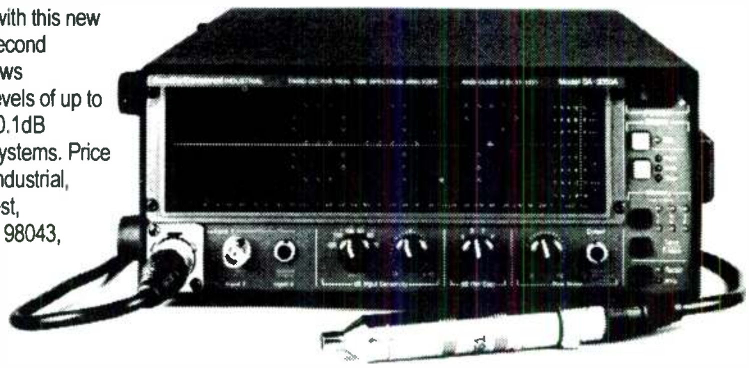
AudioControl Industrial announces two upgrades to its SA-3050A real-time analyzer. A new software update increases the resolution of the display to one-tenth dB for more accurate SPL measurements. The SA-3050A can be upgraded (factory-installed only) with this new feature for $100. The second upgrade (SPL-170) allows measurement of SPL levels of up to 170dB, as well as the 0.1dB display, for car stereo systems. Price is $795. AudioControl Industrial, 22410 70th Avenue West, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.

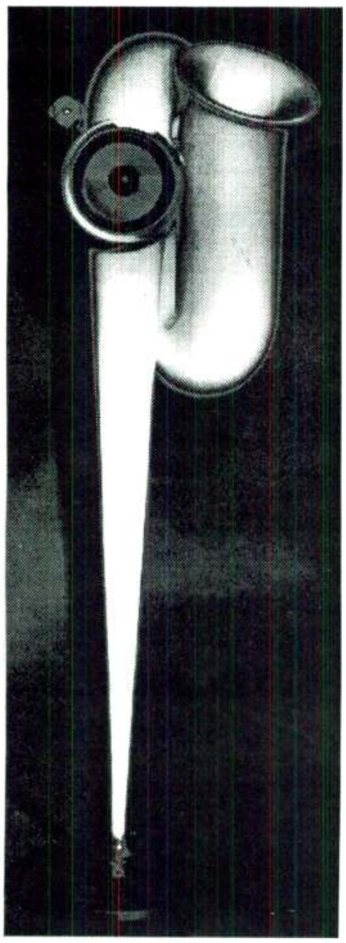
Reader Service #104

■ SWAN SOUND

Swans Speaker Systems introduces The Baton, a new loudspeaker that promises large-system features in an affordable package. The Baton (38″ tall) is a 5Ω system with 90dB sensitivity and has a frequency response of 48Hz–18.5kHz. Swans Speaker Systems, RR#1, Mason Rd., Charlottetown, Prince Edward Island, CANADA C1A 7J6, (902) 569-5520, FAX (902) 569-5123.

Reader Service #105





⊂ STANDING TALL

B&W announces the availability of its Emphasis Loudspeaker through its US dealers on a limited, special-order basis. The speaker's unique design employs one 6.5″ Kevlar-cone bass/midrange unit and one 1″ fluid-cooled metal-dome tweeter, creating an 8Ω system of 87dB SPL, 1W/1m sensitivity, which is suitable for use with amplifiers of 50–120W per channel. Response is 45Hz–25kHz (±3dB), with very low distortion. Emphasis ($9,000/pr) stands 54″ tall and weighs 77lbs. B&W, PO Box 8, 54 Concord St., N. Reading, MA 01864, (800) 370-3740, (508) 664-2870, FAX (508) 664-4109

Rea

■ SUPER-DUPER

North Creek Music Systems announces a user-installable upgrade kit for its NHT Super Zero loudspeakers. The Super-Duper upgrade ($79/pr.) features a complete passive crossover replacement, which is constructed with 14 AWG music coil inductors, Ohmite mil spec resistors, and Sprague-Dearborn high frequency metallized polypropylene film capacitors. Also included is internal wiring replacement and internal damping replacement. North Creek Music Systems, Route 8, PO Box 500, Speculator, NY 12164, (518) 548-3623.

Reader Service #102

■ HORN DRIVER

The Powerpoint is a new topology for horn drivers that promises a wider frequency response (1.0–32kHz), lower distortion, wider dynamics, and more usable clean high frequency.
The 425 model horn driver is available in either bolt-on or screw-on versions.
Recommended crossover point is 1.0kHz at 18dB per octave. For more information, contact ProSystems, 65–36th Street, Wheeling, WV 26003, (800) 258-8550, (304) 233-2223, FAX (304) 233-2258.

Reader Service #108

■ TWEETER TECHNOLOGY

Audax of France has launched its next gene
10mm tweeters: Models TM010A1 and TM
feature a neodymium motor structure and
design (29.5mm diameter). The A1 uses a
polycarbonate diaphragm and is designed
for the automotive market, while the A7
features a gold diaphragm and fits into a
special mounting pre-installed in the spea
cabinet, and the tweeter snaps into plac
Polydax Speaker Corporation (10 Upton
Wilmington, MA 01887, (508) 658-0700
FAX (508) 658-0703) for more informat

Reader Service #106

Speaker Builder (US ISSN 0199-7920) is published every six weeks (eight times a year), at $32 per year, $58 for two years; Canada add $8 per year; overseas rates $50 one year, $90 two years; by Audio Amateur Publications, Inc., Edward T. Dell, Jr., President, at 305 Union Street, PO Box 494, Peterborough, NH 03458-0494. Second-class postage paid at Peterborough, NH and an additional mailing office.
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← ***Reader Service #36***

About This Issue

Are you eager to tackle a challenging construction project? Designer **D.E. Stenton** offers the ambitious enthusiast who can overcome the construction difficulties—the design requires patience and care—the opportunity to be rewarded with unrivalled sound reproduction. His System III Loudspeaker, which demonstrates how properly applied engineering principles can result in outstanding sound, combines a low-frequency enclosure with lo-mid, mid, and high-frequency drivers as open-backed types. System III is an innovative project for the home builder.

Sometimes a little theory is good for the soul of the hands-on speaker builder; it promotes a better understanding of loudspeaker design and behavior. In "Exploring Loudspeaker Impedance," **Victor Staggs** presents some heavy, but quite useful, technical matter. You'll learn about impedance curve theory; at the same time, you'll achieve computer-optimized design results without an optimization program. All in all, this academic article is quite practical.

We're all familiar with cycling, but have you ever taken a ride with **Bill Waslo**'s fast-moving IMP program? His latest version updates your measurement results on the fly; that is, now you can watch the frequency response, for example, change while you make adjustments on the crossover. These instant results help you fine-tune a system more efficiently.

Does today's stereo sound leave you [redacted]fulfilled? Do you wish for more from [redacted]d reproduction? If so, then you're not [redacted]e. **Philip Witham** offers a significant [redacted]native that has piqued the curiosity of [redacted]y *SB* readers. To find out what the hub- [redacted] is about, turn to page 43 to read how [redacted]am—buoyed by reader feedback and [redacted]stions—plans to implement his radical [redacted]r-Array project.

[redacted]ating good corner joints is usually the [redacted]ritical part of any woodworking project [redacted]always a chore, unless you take advan- [redacted] woodcraftsman **Bob Wayland**'s tech- [redacted] By following his tips in this issue, [redacted]oon be creating strong, attractive cor- [redacted]s in your speaker enclosures.

[redacted] to keep you up-to-date on the latest [redacted]orld of audio, we offer reviews of [redacted] accessories: C & S Audio Labs' [redacted]Tester software and noted author [redacted]kason's new *Recipes* book.

Speaker Builder

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

–John Stuart Mill

Speaker Builder is published eight times [redacted]

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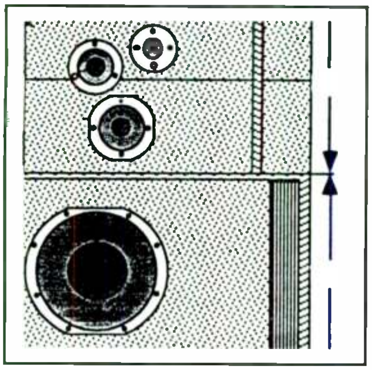
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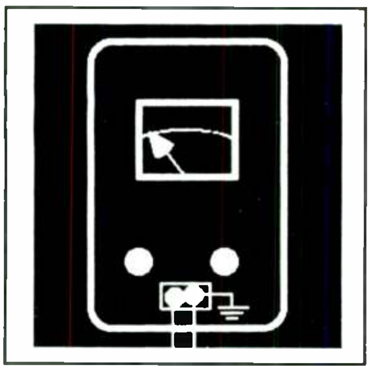
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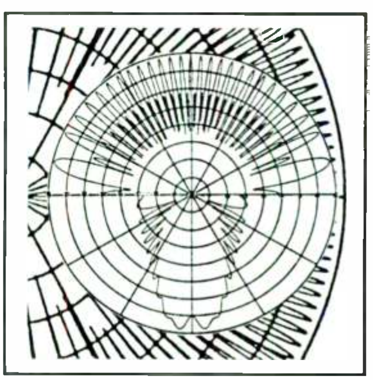
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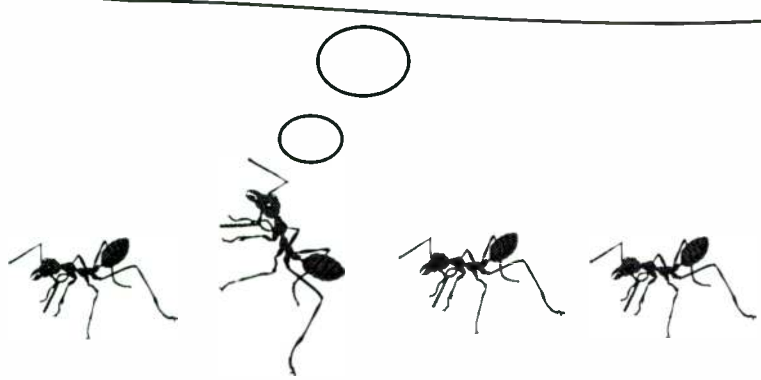
DEPARTMENTS

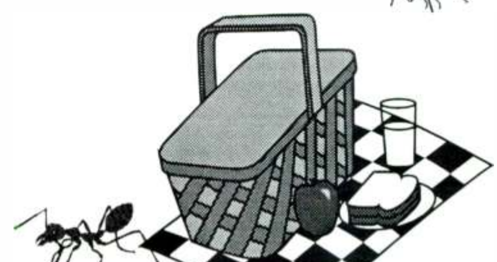
Madisound's
Picnic Basket Pickings

Madisound; P.O. Box 44283, Madison WI 53744; T:608-831-3433 Fax: 608-831-3771

QTY	Description	Price Each
11,000	Wesco 1.5 mfd Mylar Capacitor, 10%, 100V, 23mm x 10mm x 6mm, Axial 55mm long leads	10 for $2
3,100	Elpac 2.16 mfd Mylar capacitor, 5%, 200V, 30mm x 14mm x 9mm, Axial 30mm long leads	10 for $3
17,000	Hitachi 4.7 mfd Mylar Capacitor, 10%, 100V, 30mm x 12mm x 6mm, Axial 13mm long leads	10 for $3
1,000	KSC 75 mfd Bipolar Electrolytic Capacitor, 10%, 100V, 34mm x 16mm Ø, Axial 46mm long leads	10 for $9
850	Tecate 500 mfd Bipolar Electrolytic Capacitor, 10%, 100V, 51mm x 25mm Ø, Axial 50mm leads	$3.50 each
2,100	.2 mH Air Core Inductor, 19 awg wire on a plastic bobbin; 1 1/2" Ø x 1 1/8" T	10 for $6
1,000	2.0 mH Air Core Inductor, 19 awg, bobbinless, 2" Ø x 1" T	$1.50 each
4,700	10 Ω Sand Cast Resistor, 5%, 5W, 22mm x 9mm x 9mm, Axial 40mm leads	10 for $1.50
85	Motorola KSN 1025 1.8KHz Piezo Horn 2 x 6, 1.8KHz to 30 KHz, 92dB, 75W, 187mm x 79mm	$5.50 each
240	Seas 25TF H456 1" Damped textile dome tweeter; 104mm flange, Fs 1000Hz, 6Ω, 91.5dB, 50W	$12 each
60	MB Quart MCD-19E 3/4" Titanium dome tweeter, 95mm sq. flange, Fs 2430 Hz, 91dB, 8Ω, 100W	$24 each
210 pair	Europa 23 Surface Mount tweeter module, 6dB filter, 1" dome, 4Ω, 70mm T x 60mm D x 68mm W	$15 per pair
270	Vifa D25TG-55-06 1" Polymide dome tweeter, liquid cooled, 91dB, Fs 1500 Hz, 6Ω, 100W, 4" flange	$10.50 each
250	Sledgehammer CX-33 Electronic Autosound X-over, 2 or 3-way, 12dB, Level controls, phase switch	$50 each
28	Focal 4K111 Kevlar cone 4.5" midrange, 8Ω, rubber surround, cast frame, 110mm square, Fs 70 Hz, 91 dB, 35W, F3 200 Hz in 1 ltr sealed, exceptional clarity and smooth response to 5KHz	$60 each
12	Focal 5N412 Neoflex cone 5.25" woofer, 8Ω, rubber surround, cast frame, 130mm Square, 88dB, Fs 49 Hz, Qms 3.28, Qes .32, Qts .29, Vas 10 ltrs, 40mm VC, 5mm X-max, smooth response to 3.2KHz	$55 each
42	Vifa M21WP-00-08 Carbon/Paper cone 8" woofer, 8Ω, rubber surround, cast frame, Fs 32 Hz, Qms 2.46, Qes .43, Qts .36, Vas 80ltrs, 100W, 88dB, 40mm VC, 5mm X-max; 1.7cf, F3 40 Hz, 2" Ø x 2.5"	$30 each
30	Vifa P21WO-00-08 Poly cone 8" woofer, 8Ω, rubber suround, cast frame, Fs 25 Hz, Qms 1.9, Qes .34, Qts .29, Vas 148 ltrs, 70W, 91dB, 40mm VC, 3mm X-max, 1.5cf, F3 42 Hz, 2" Ø x 3" long	$25 each
43	Vifa M21WG-00-08 Paper cone 8" woofer, 8Ω, Foam surround, cast frame, Fs 40 Hz, Qms 4.4, Qes .73, Qts .63, Vas 86 ltrs, 70W, 91dB, 25mm VC, 3mm X-max, F3 50 Hz in 1-2cf sealed enclosure	$20 each
40	Scanspeak 18W/8544-05 Kevlar cone 8" woofer, 8Ω, Foam surround, cast frame, Fs 40 Hz, Qms 2.19, Qes .43, Qts .36, Vas 26 ltrs, F3 50 Hz in 1/2cf 2" Ø x 6.5" L or 79 Hz in 10 ltrs sealed	$66 each
52	Gefco 10" Passive radiator with rubber surround, poly cone, spider attached.	$16 each
68	Gefco 12" Passive radiator with rubber surround, poly cone, spider attached.	$20 each
295	Peerless 831857 CC line 12" woofer, 8Ω, rubber surround, 46oz magnet, 9mm X-max, 89.3dB, Fs 24Hz, Qms 3.72, Qes .49, Qts .44, Vas 210 ltrs, 220W, F3 25 Hz / 6cf 4" Ø x 4.6" or F3 39Hz in 3cf S	$70 each

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 or bill you to the exact shipping charge. We accept Mastercard or Visa on mail and phone orders.





Offer Expires
Sept. 30, 1994

Reader Service #1

Editorial

Another Road to Oz

Newcomers to speaker technology are often, like Dorothy, swept up out of Kansas into a swirling funnel of anticipation, looking for some kind of magical road that leads to loudspeaker Oz. Like Dorothy, the company is either rather poor or nonexistent.

A lot of readers seem to feel like the Lion, timorous, uncertain, and with a deep need for not only courage but large inputs of data. This is scarcely surprising. Most of us, having experienced a moment of something near revelation the first time we heard a genuinely good, wide-range loudspeaker system being driven by an equally competent set of components, are set on a quest for much more of the same. Something deeply primitive in each of us can be reached by beautiful full-bodied music—live or recorded. For a great many of us the appetite is built in and is rarely ever fully satisfied.

The frustration, however, is palpable. Most of us have budget limitations which keep the kinds of hardware we audition in the mall or at the stereo emporium permanently out of reach. The discovery that we might build something which gives us a permanent and affordable source for abundant, unlimited quantities of beautiful sound becomes a passion, a kind of Grail, in a very short time span.

Intellectually, we all realize, if we consider the matter rationally, that no quick, painless routes to significant achievement exist. The nature of the appetite, however, almost certainly makes rationality amid a new love affair with sound practically impossible.

A few of you write to the editors regularly asking for help. It's usually of two kinds: The first goes something like, "I have heard a great commercial speaker and I'd like to build my own version of it, but I'm not sure how to begin. Please tell me what drivers I should buy and which crossover I should choose." The question looks simple enough, but it asks for a great deal, in both senses of those two words. If we could supply what you think you are asking, the result would stunt your development as a speaker builder, probably forever. The amount of research time required from a competent designer would be far too expensive for us to afford.

The second sort of request we receive probably assumes that the first scenario is impractical so the request is framed differently. These nascent speaker builders ask for a buddy system. Since speaker building is a relatively small avocation, candidates to fullfill your need for a buddy are rather limited. The loneliness of those on this quest is certainly real. The number of clubs which fill several columns in each issue of *Speaker Builder* are testimony to that fact. It is certainly true that both misery, and fun, love company. Most enterprises are better done when someone else can be part of them.

With the latter need in mind, we would be happy to serve as a dating service for speaker builders who need a helping hand, provided that some of you are willing to offer your hand for a possible cooperative enterprise. Safeguards for such proposed liaisons are a must, of course. The requester's phone number would be supplied to the responder and a series of phone calls would allow the responder to evaluate not only the proposed project but the proposed relationship as well. A no-fault decision option by the responder would be necessary in every case.

We have put together questionnaires for both requesters and assisters. If you'd like a copy, send us an envelope with your name and address on it with a loose stamp or a postal coupon if you live outside the US. Address them to the magazine, marked either "HELP" or "HELPER."

A few practical considerations are in order for those requesting help. First, choose a simple project if you have never built a speaker system before. Building audio equipment is not a goal. It is a journey. Good reproduced sound is a quest for a Grail which will recede each time you think it is within your grasp. The world is full of clichés about this advice. But it is vital to your success to heed it.

Second, for both the helper and those needing help, the goal for the beginner is to learn how to do the project—not just to get the project done. No one can grow or become truly proficient at these disciplines without homework. Take the time to work through learning what you need to know. The assignments you decide on should be practical and specific, not generalized. Speaker builders do not need algebra in general, they need it for a specific task, which is always the best way to learn a skill.

If a club dedicated to audio enthusiasms is within commuting distance, I encourage you to seriously consider becoming a part of it. I would hope that clubs, whose agendas differ widely, would always be willing to foster audio construction interests as part of their programs and provide a way for like-minded builders of any sort of gear to get together.

One further piece of general advice may be appropriate for your personal journey to Oz. I believe it impossible to overemphasize the importance of reading, thoroughly, any subject-related catalog you can find. They are not only a resource listing, but also an educational tool. They merit very serious consideration and careful reading. You will be surprised at how much you can learn about not only what is new, but the basics of how tools, devices, machines, parts, and whatever work.

If you are feeling like you are being swirled upward in a thrilling but scary, unfamiliar tornado of excitement and desire for better-sounding speakers, we can try to help with your adventure in the ways I have tried to suggest here. It all begins with staying in touch. —E.T.D.

THE SYSTEM III LOUDSPEAKER

By D.E. Stenton

The past 20 years demonstrate how loudspeaker systems can be effectively engineered to significantly improve home stereo sound. Although new speaker materials have played an important role, properly engineered system designs are key to the success of the industry. Interestingly, these engineering principles are hardly new.

After reviewing numerous research papers and books—both old and new—and evaluating many design and construction techniques, I developed the full-range SYSTEM III Loudspeaker, which I designed to meet the highest standards of audio performance and listening pleasure (*Photo 1*).

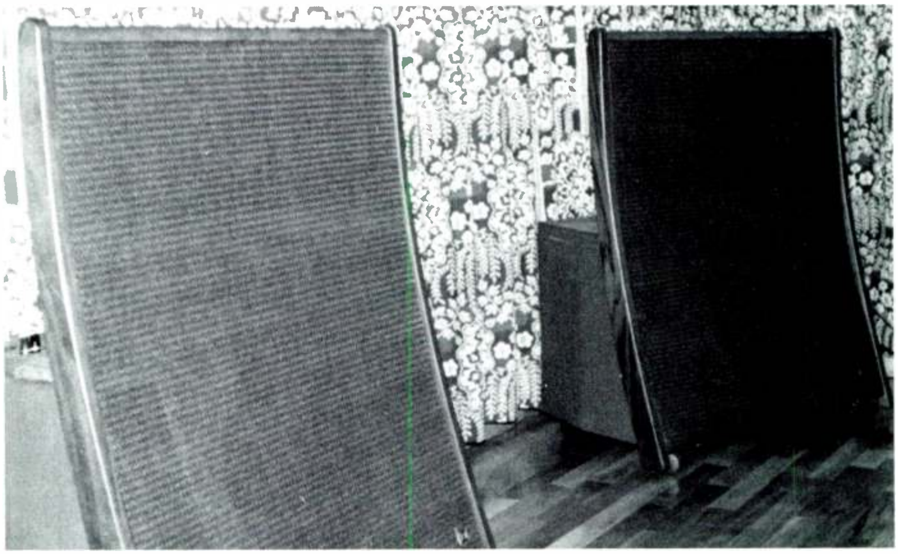


PHOTO 1: The SYSTEM III Loudspeakers are uniquely designed to enhance listening pleasure.

MEET THE SYSTEM

The SYSTEM III Loudspeaker consists of four drivers: a low-frequency, a midrange, a midrange bridging, and a high-frequency type. An active crossover divides electrical signals applied to the system into low- and high-pass frequencies at about 300Hz.

The output of each crossover section is sent to two power amplifiers: one to drive the low-frequency speaker and the other to drive the upper-frequency range, which consists of two midrange drivers and one high-frequency driver. Each receives a portion of the sound spectrum from a special passive crossover network, which I carefully designed to provide a three-stage filter rather than the more conventional two-stage high-/low-pass filter. The third element in this passive filter network is a "bridging" circuit, together with the driver, which maintains both phase and amplitude linearity in the critical region around crossover frequency.

ABOUT THE AUTHOR

Donald E. Stenton is the undergraduate laboratory supervisor in the Department of Physics and Astronomy at the University of Victoria. He received a BS in math and physics from the University of British Columbia in 1962. Although he has maintained a broad interest in all fields of physics, sound systems have been a long-term investment in both time and effort, beginning with his development of a computer program in 1972 which used Thiele's parameters to solve bass-reflex enclosure designs. He is a member of both the AAPT (American Association of Physics Teachers) and the AES (Audio Engineering Society).

I'll describe my SYSTEM III in three parts: design characteristics, electronic networks, and construction.

PART 1, DESIGN

VENTED VS. SEALED

Of the many types of enclosure design choices for reproducing bass frequencies, vented-box systems offer the best tradeoff between bandwidth and efficiency and size and cost. Choosing a large driver for reasonable low-frequency sound pressure levels (SPLs) assumes a large cabinet.[1] If cabinet size is important to you, then you can usually make a vented enclosure (as opposed to a transmission-line or folded-horn) considerably smaller for a given cutoff frequency.

In a vented enclosure, the amount of acoustical power radiated at low frequencies depends on the radiation output from both the driver and the port, whereas the acoustical output of an infinite-baffle (sealed) model relies almost entirely on the motion of the driver cone.

Figure 1, which illustrates several advantages of a vented unit over a sealed one, shows a graph of the relative cone excursion at frequencies above and below the system cutoff frequency for constant power input. The three Butterworth filter response curves illustrated on this graph are: a second-order sealed system (B_2), and a fourth-order (B_4) and a sixth-order (B_6) vented-box alignment. Considering the speaker's response, the order of the alignment defines the approximate attenuation rate of acoustical output below the system's resonant frequency: i.e.,

- second-order (B_2) rate of attenuation 12dB/octave
- fourth-order (B_4) rate of attenuation 24dB/octave
- sixth-order (B_6) rate of attenuation 36dB/octave

(Note that f_3 equals f_B for this graph.)

This graph illustrates the following:

1. Above ($3 \times f_3$) the vent becomes inoperative, resulting in similar cone excursions for both systems.
2. The sealed system has its maximum cone excursion at f_3, while the vented version has its minimum at this point (its maximum occurs at $1.45 \times f_3$).
3. The vented maximum excursion in the passband is only 0.35 times that of the sealed system, so its f_3 cutoff frequency can be reduced to 0.6 times that of the sealed system for a given speaker without additional cone excursion.

4. Cone excursions in filtered systems (see B_6 alignment) at subsonic frequencies have less amplitude than those without ancillary filters.

Therefore, a vented enclosure provides greater acoustical output at the lowest frequencies with less resonance distortion than a similarly sized sealed box.

GREAT MOMENTS IN HISTORY

In 1961 A.N. Thiele first suggested applying transfer functions to determine speaker/amplifier/vented-box acoustic response variations with frequency. Since then many authors, including Benson, Ashley, Keele, Small, and others, have reviewed and developed that theme.[2, 3, 4] Using transfer functions probably presents no problem to audio experts, but to the uninitiated, their use may be puzzling.

Assuming that the acoustic efficiency of the vented box was low, Thiele demonstrated that a fourth-order high-pass filter function could represent the response. The speaker's behavior in a vented enclosure could therefore be represented by the transfer function

$$G_H(s)=\frac{s^4T_0^4}{s^4T_0^4+\alpha_1 s^3T_0^3+\alpha_2 s^2T_0^2+\alpha_3 sT_0+\alpha_4} \qquad (1)$$

where s is the complex variable, $\sigma + j\omega$; T_0 is the nominal filter time constant; and the four coefficients, $\alpha 1$, $\alpha 2$, $\alpha 3$, $\alpha 4$, determine the response curve's shape. The system can be designed for maximal flatness with a response below cutoff at 24dB/octave.

This brilliant analysis let theorists use analytic expressions for box volume, vent operation, damping factors, tuning ratios, and so on. Independently, Small determined that the measured and calculated values in these expressions were in agreement to within ±0.25dB.[4]

Within the limits imposed by this assumption, the system's electrical/mechanical/acoustical characteristics can be determined, and the coefficients expressed, in terms of these quantities. To understand the relationship of the variables within the transfer function, I've developed "The Synthesis of a Practical Sixth-Order Butterworth Response" in *Appendix 1(see pp. 24-25).*

These results show that a speaker's low-frequency performance can be adequately defined by several parameters: free air resonance (f_S), the resonant frequency of the enclosure (f_B), the compliance ratio of the speaker suspension to the volume of air in the enclosure (C_{AS}/C_{AB}), and the total Q of the driver including all system resistances at the resonant frequency (Q_T).

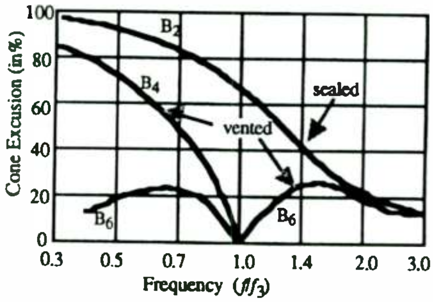


FIGURE 1: Percent cone excursion versus normalized frequency.

TURNING THE TABLES

You can design a low-frequency vented enclosure in one of several ways: for example, choosing a suitable alignment, substituting the correct driver parameters into the alignment equation, and solving for enclosure volume. However, a simpler way is to determine the appropriate system characteristics from an alignment table, as in *Table 1.*

Of course, if this method is to be useful, either the manufacturer must supply the desired parameters, or you must determine them for a given driver. The important parameters are: free air resonance, f_S; electrical Q_E of the driver at f_S; acoustical Q_A at f_S; DC resistance of voice coil, R_E; and ratio of acoustic compliance of driver suspension to free air in the enclosure, C_{AS}/C_{AB}.

The last parameter, C_{AS}/C_{AB}, measures the ratio of the volume of air having the same acoustical compliance as the driver to the net internal volume of the enclosure. It is defined as V_{AS}/V_B. If you intend to select a suitable alignment for a specific driver characteristic, it is more practical to invert this ratio and work with the inverse ratio of volume, V_B/V_{AS}, rather than the compliance ratio.

In the second part of his paper, Thiele produced a table describing 28 filter alignments, varying from third-order quasi-Butterworth to sixth-order Chebyshev functions.[2] Each parameter, when substituted into the correct transfer function, gave the designer an accurate way to predict the system's response.

I have slightly modified Thiele's original table (*Table 1*, Summary of Loudspeaker Alignment Data). To better represent the relative change in enclosure volume within each alignment group, I designated the compliance ratio in column 8 as V_B/V_{AS}, rather than its inverse ratio. I selected several alignments from *Table 1* and show them, evaluated, in *Table 2*.

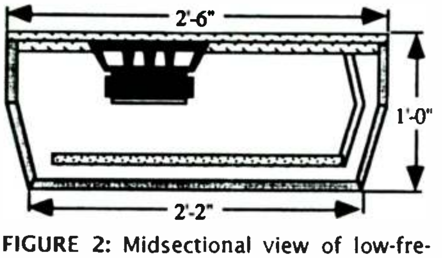


FIGURE 2: Midsectional view of low-frequency enclosure.

SMALLER IS BETTER

In designing my system, I gave top priority to two factors: (1) the volume of the enclosure should be as small as possible, and (2) the value of the cutoff frequency should be about 20Hz. At first glance at *Table 2*, alignment 5 seems a good choice since the resonant frequency, f_B, is tuned to free air resonance, f_S.

Alignment 15, however, offers similar Butterworth filter alignment characteristics *plus* smaller size, determined from the ratio of the V_B/V_{AS}, shown in *Table 1*. In this alignment, the compliance ratio C_{AS}/C_{AB} has a value of 0.366. Using this value, you can determine a net enclosure volume almost 50% less than in alignment 5, obviously a significant reduction, but not without a penalty.

Notice that Q_T in alignment 15 in *Table 2* is about 20% less than in alignment 5. Equation 22 shows how the system Q_T relates to the electrical Q_E, acoustical Q_A, amplifier output resistance, and driver resistance. In practice the electrical Q_E is the most dominant factor; therefore, Q_T can only be achieved successfully with adequate electrical damping. (Thiele and others have suggested using feedback in an

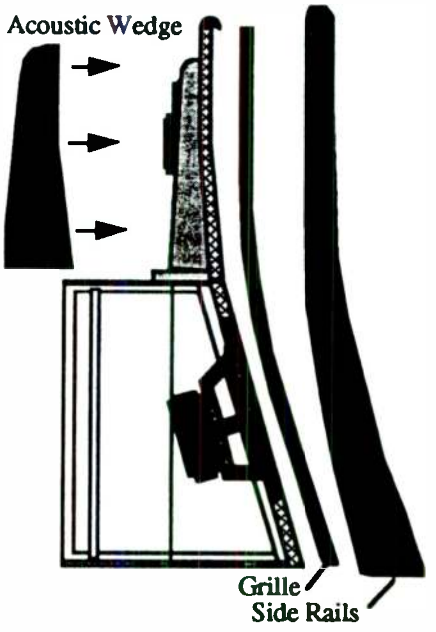


FIGURE 3: Side view of SYSTEM III Loudspeaker.

amplifier circuit as an alternative way to lower the system Q.)

To achieve proper damping the speaker in alignment 15 must have a smaller electrical Q_E value; i.e., a much larger magnet than would normally be acceptable in alignment 5. Unfortunately, manufacturers of drivers greater than 12" rarely provide a large enough magnet to satisfy many of these alignments.

CALCULATIONS

My bass driver meets the above electrical and mechanical requirements. It is a 12", low-impedance woofer, suitable for many possible alignments. Three characteristics, f_S, V_{AS}, and Q_E—often referred to as the Thiele system parameters—when substituted into equations 12, 13, 16, and 17, yield the system parameters shown in *Table 2*. However, these computations do not take into account the enclosure losses suggested in *Appendix 1*.

Using equations 16 and 17, you can obtain corrected values for Q_T and V_{AS}/V_{AB}. The results suggest a 5% Q_T increase and a 20% increase for net volume, V_B. These quantities now become

$Q_T = 0.312$ and $V_B = 4{,}400$ in^3

compared to the original values of

$Q_T = 0.299$ and $V_B = 3{,}660$ in^3.

However, measuring the system Q_T does not always produce the same value as that specified in the alignment tables (or calculated quantities). Since the system Q is not an adjustable parameter, but depends predominantly on the electrical Q_E, which, in turn, relies on the driver's magnet and voice-coil construction, it is necessary to make an adjustment to R_G, the amplifier resistance. This parameter is related to the electrical and mechanical Q in equation (22).

To ensure correct damping, a small resistance, which will effectively increase the total system Q, can be introduced in series with R_G. This method will, however, lower power to the low-frequency driver. Since the amplifier's internal resistance is often about 1/100Ω, any further resistance to the circuit will be significant. I used a 0.5W resistor (*Fig. 7*).

DUCT WORK

Since the enclosure must be tuned to the driver's free air resonance, you must install a vent, or duct. Normally, you would secure a cylindrical tube to the front baffle and project it towards the rear. However, this was impossible in my system for several reasons. As an alternative, I added a ducted port as part of the wall and back.

One of the first port design considerations in a reflex enclosure is size, i.e., the area of the port S_V. Small suggested that the minimum area is related to the enclosure's resonant frequency and the driver's peak displacement volume, V_D.[4] A value for the port area can be determined by

$$S_V \le 0.025 f_B V_D \quad (2)$$

where V_D is in^3. Areas less than those in equation (1) can often cause vent turbulence and audible wind noise at the port's output. In my design I used a 16 in^2 area, which proved satisfactory.

TABLE 1

	Alignment Details				Box Design				Ancillary Circuits			
	No.	Type	K	Ripple	f_3/f_s	f_b/f_s	V_b/V_{as}	Q_t	f_{aux}/f_s	X_{aux}	Peak	f_{pk}/f_s
Quasi	1	QB3	–	–	2.68	2.00	0.095	0.180	–	–	–	–
Third	2	QB3	–	–	2.28	1.73	0.134	0.209	–	–	–	–
Order	3	QB3	–	–	1.77	1.42	0.224	0.259	–	–	–	–
	4	QB3	–	–	1.45	1.23	0.339	0.303	–	–	–	–
Fourth	5	B4	1.0	–	1.00	1.00	0.707	0.383	–	–	–	–
Order	6	C4	.8	–	.867	.927	0.945	0.415	–	–	–	–
	7	C4	.6	.13	.729	.829	1.372	0.466	–	–	–	–
	8	C4	.6	.25	.641	.757	1.790	0.518	–	–	–	–
	9	C4	.6	.55	.600	.716	2.062	0.557	–	–	–	–
	91/2	C4	.6	1.52	.520	.638	2.600	0.625	–	–	–	–
Fifth	10	B5	1.0	–	1.00	1.00	1.000	0.447	1.000	–	–	–
Order	11	C5	.7	–	.852	.912	1.715	0.545	1.218	–	–	–
	12	C5	.4	.25	.724	.814	3.663	0.810	1.810	–	–	–
	13	C5	.4	.5	.704	.798	4.405	0.924	2.06	–	–	–
	14	C5	.3	1.0	.685	.781	5.236	1.102	2.47	–	–	–
Sixth	15	B6	1.0	–	1.00	1.00	0.366	0.299	1.000	0.518	+6.0	1.07
Order	16	C6	.8	–	.850	.979	0.429	0.317	0.858	0.420	+7.7	0.90
Class	17	C6	.6	–	.689	.931	0.552	0.348	0.712	0.318	+10.1	0.73
I	18	C6	.5	–	.620	.888	0.662	0.371	0.639	0.265	+11.6	0.65
	19	C6	.4	0.1	.554	.841	0.800	0.399	0.576	0.222	+13.2	0.58
Sixth	20	B6	1.0	–	1.00	1.00	1.000	0.408	1.000	1.414	–	–
Order	21	C6	.8	–	.844	.885	1.385	0.431	0.928	1.250	+0.2	1.99
Class	22	C6	.6	–	.677	.738	2.000	0.461	0.819	1.029	+1.1	1.18
II	23	C6	.5	–	.592	.656	2.415	0.484	0.752	0.895	+1.9	0.96
	24	C6	.4	0.1	.520	.584	2.832	0.513	0.681	0.766	+3.0	0.81
Sixth	25	C6	.3	0.1	.404	.461	3.623	0.616	0.553	0.518	+6.0	0.59
Order	26	B6	1.0	–	1.00	1.00	1.366	0.518	1.000	1.931	–	–
Class	27	C6	.3	0.6	.778	.854	9.091	1.503	2.12	1.414	–	–
III	28	QB3	–	–	.952	.971	0.529	0.328	1.028	–	+6.0	0.0

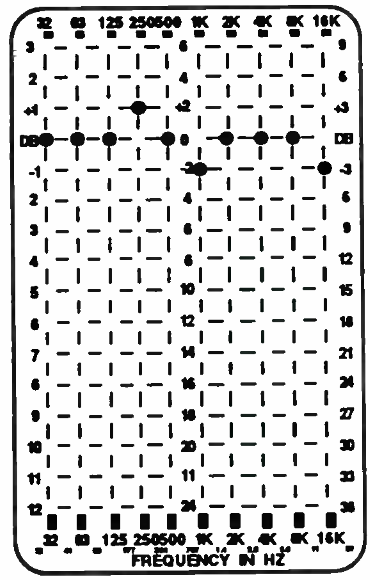


FIGURE 4: Audio spectrum analyzer frequency response in 2dB steps.

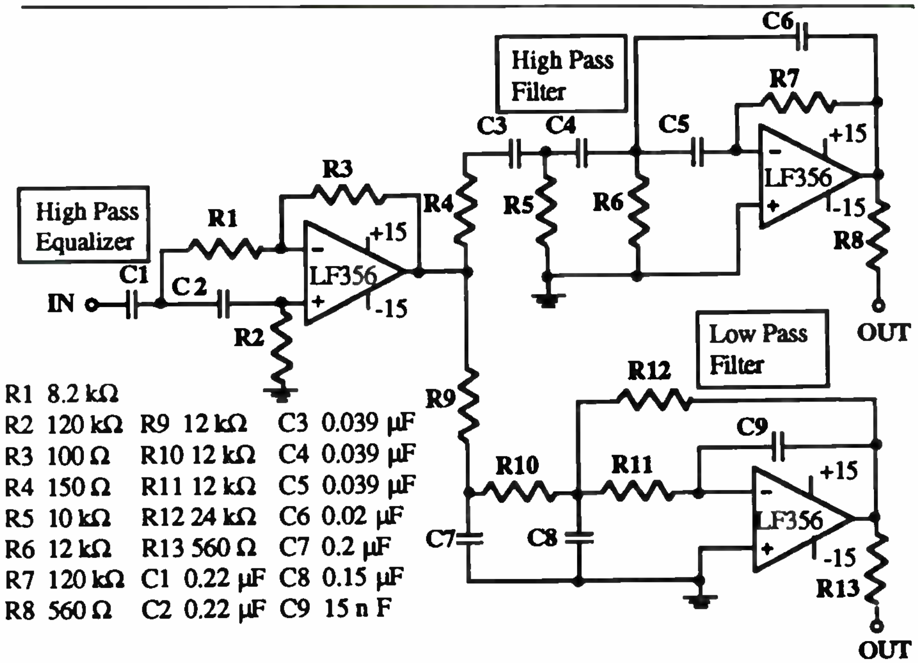


FIGURE 5: High-pass equalizer and high-pass/low-pass active filter circuit.

Once you determine the port's area, use the following equation to calculate the length of the duct:

$$\frac{L_V}{S_V}=\frac{1.82\times10^8}{\omega_B^2 V_B} \quad (3)$$

where L_V = length of the vent in inches and S_V = area of the vent (in^2).

Since the duct terminates in a semi-flanged surface, you must include an end correction, which is determined by

$$\left(\frac{L_V}{S_V}\right)_{end}=\frac{0.77}{\sqrt{S_V}} \quad (4)$$

The length of the duct—approximately 33"—is calculated using equations (3) and (4). To accommodate this length in such a small enclosure, I installed a panel along the wall and back as shown in *Fig. 2*, and in more detail in *Fig. 8*.

SPEAKER TWEAKER

Alignment 15 belongs with the Class 1 sixth-order high-pass filter functions and can be represented as the product of two functions—a fourth-order polynomial and a second-order polynomial. Since vented-unit acoustics have an equivalent fourth-order high-pass characteristic, the remaining second-order polynomial may be specified as a suitable electronic filter (described in Part 2).

To ensure that the parameters agree with the constants specified by the fourth-order portion of the transfer function and that the overall system response follows the theoretical model, it is often necessary to alter one of the speaker's parameters, i.e., R_E.

MIDFREQUENCY REPRODUCTION

At midfrequencies, where the sound wavelength is much less than the cone diameter, and the cone no longer vibrates as a rigid piston, the sound field tends to be directional. When this occurs, the most radiation energy is emitted along the speaker's axis. Further increasing frequency usually produces a narrow beam of sound, with "lobes" of sound radiation on either side of the main beam.

Conventionally constructed, single-cone speakers become increasingly directional with increased frequency.[5] For good imaging and clarity, however, speakers must achieve radiation uniformity, i.e., a consistent distribution of the sound intensity as a function of frequency—both direct and reflected sound, which the listener receives. Notably, less directional speakers produce more reverberant sound in a listening room and provide a more spacious effect.

Normally, midrange drivers are housed in the same enclosure as the bass driver, but not the woofer, since these drivers often suffer intermodulation distortion due to pressure changes caused by the bass driver. To solve this problem, midrange speakers often have their own enclosure, which, unfortunately, may cause a cavity in which midfrequency standing waves occur. These cavity standing waves can be re-radiated through the cone, causing considerable "coloration" of the original sound.

DON'T BECOME BAFFLED

Some systems feature an "open" baffle, which lacks an enclosure for the midrange sound reproducer. The midrange drivers and high-frequency unit are mounted so that the back of each diaphragm is surrounded by acoustic-absorbing material, and each speaker effectively radiates into free space in both the back- and front-side spaces.

Since an open baffle unit radiates acoustical patterns in both directions—front and rear—and since these two signals are equal in amplitude, but opposite in phase, there can be cancellation—destructive interference of the front and rear wave patterns. However, cancellation can only approach

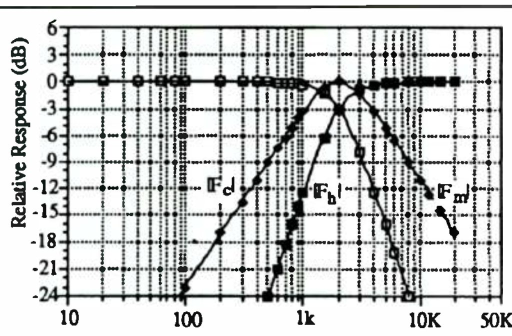


FIGURE 6: Response characteristic of the low-pass, high-pass and "bridging" crossover network.

TABLE 2

SELECTED ALIGNMENTS FROM *TABLE 1*.

No.	Order	f_3 (Hz)	f_B (Hz)	V_B (in.^{3})	Q_T	Aux. Filter
4	QB3	32	27	3,390	0.303	no
5	B4	22	22	7,070	0.383	no
6	C4	19	20	9,480	0.415	no
15	B6	22	22	3,660	0.299	yes
16	C6	19	21.5	4,290	0.317	yes

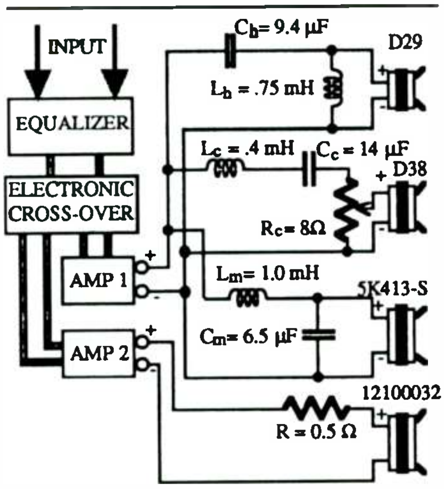


FIGURE 7: Passive crossover circuit used with mid-/center-/high-frequency drivers.

completion on each side of the baffle on a line perpendicular to its principle axis.[6]

Consider a noncircular baffle where the distance from the speaker center to the nearest edge is *d*. When the sound wavelength, λ, is much greater than the distance, 2*d*, from the front side of the cone around the edge to the back side, then the compressed air on one side can easily fill the air vacancy on the other. The net sound intensity will be reduced and continue to decrease at 6dB/octave below a frequency, f_D given by $f_D = c/2\pi d$, where c is the speed of sound in air.

Using a large baffle and avoiding very low crossover frequencies is usually sufficient to prevent the inefficiencies of dipole-type radiation. For midrange and treble speakers this application is practical, but to obtain reasonable bass response, the baffle size becomes too large for most applications.

Tests indicate that the bipolar radiation in the midfrequency range of a large open baffle can create a clear, well-defined sound image with a sense of spaciousness not often heard from midrange speakers in sealed enclosures. An example of this construction type is illustrated in *Fig. 3*.

Using two or more drivers for a full sound spectrum can introduce several more problems, including the relationship between the length of the acoustical path to the listener along the acoustic axis of each driver. Differences will introduce phase distortion, which affects transient performance.

Since each driver's natural phase response varies with frequency, there exists a relationship between each speaker's acoustic center and the group time delay. At low frequencies, the center can be substantially behind the speaker on the baffle.

AN OLD AUDIO MAXIM

You can compensate for group delay by positioning individual drivers on a curved surface, as illustrated in *Fig. 3*.[7] The acoustic center of the high-frequency driver is just behind the midrange driver, whose acoustic center is behind the low-frequency driver. The path length from the acoustic axis of each driver to the listener is therefore approximately the same.

Unfortunately, this is a relatively insignificant cure for phase distortion, since phase errors due to driver placement are small compared to those caused by incorrect crossover design.

It has often been said that crossover networks can be "heard." Hansen and Madsen demonstrated that phase distortion does affect transient response.[8] Crossover designs which compromise on rate of attenuation, lack phase coherence, or introduce uneven response in the passband are unacceptable in today's systems.

CORRECT CROSSOVER

In an *ideal* crossover, the vector sum of the voltages to each driver equals the input voltage for the accurate transfer of amplitude and phase of the originating signal. Passive filters with a 6dB/octave slope response provide such a constant voltage crossover network, as long as the drivers maintain a flat frequency response over a four-octave range, which, in practice, is difficult to achieve.

Increasing the crossover filter attenuation to 12dB/octave reduces this requirement to a two-octave range, but introduces another problem—a null at crossover. A third-order network, while providing a constant total signal amplitude, introduces a complete phase reversal at crossover.

A HOLE PROBLEM

When using even-order filters as effective crossovers, designers have adopted several

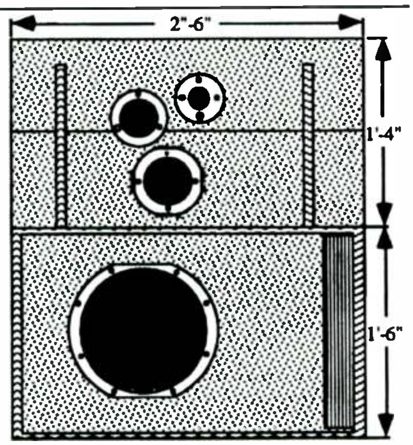


FIGURE 8: Front view showing driver locations.

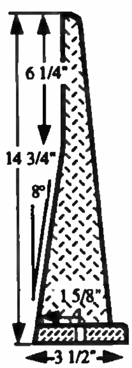


FIGURE 9: Side view of baffle supports.

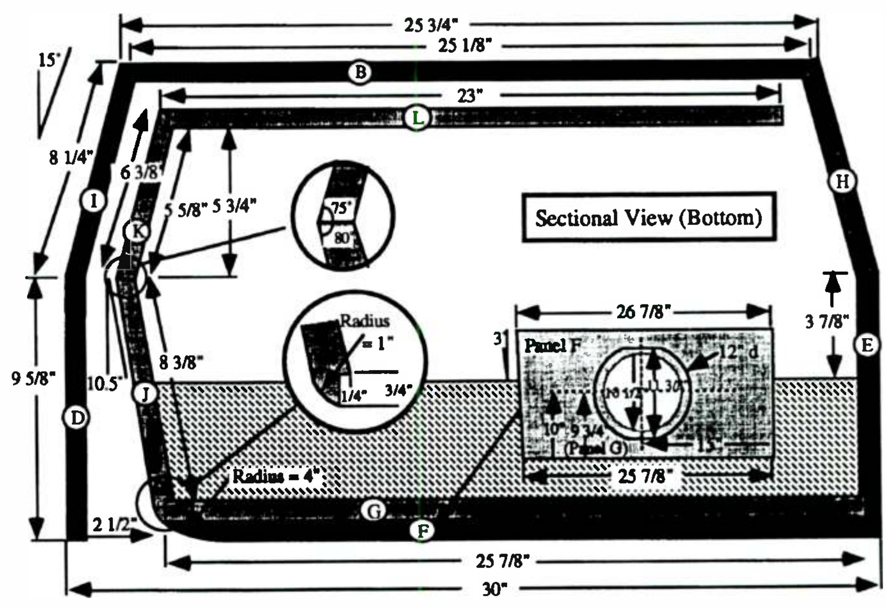


FIGURE 10: Low-frequency enclosure showing cross-sectional view of cabinet bottom.

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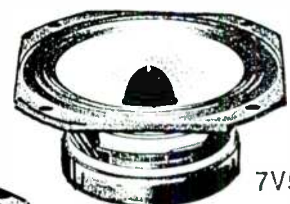
- **4" FOCAL midrange:** 40111, cast frame, curvilinear profile impregnated paper cone, rubber surround, super clean impulse, ideal from 250 Hz and up it goes to a real 10 KHz! Easy job on crossover.
- **4" FOCAL midbass:** 4C212, cast frame, semi exponential coated paper cone, negative rubber surround, sharp impulse here also, and real muscle too! Will work in 2-way + sub. Nice roll-off around 2-3 KHz.
- **6" FOCAL midbass:** 60211S, new low profile 6" frame, latex damped paper cone, Neoprene surround, 4 layers VC. Ideal for potent 2-way, easy to cross over.
- **7" FOCAL High efficiency midrange/midbass:** 7V513, cast frame, Polyglass cone with phase plug, coated foam surround. Edge-wound flat copper wire VC on Kapton former, massive magnet structure. An explosive high power midrange, more than 93 dB/W/m. Detailed and elegant it is the rare find for tube amp. officionados!

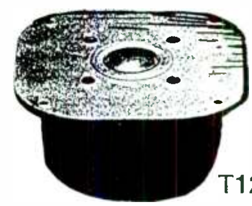
4C111

6C211S



7V513

4C212



T120ZLT

	4C111	4C212	6C211S	7V513	T120ZLT
DCR	6.16	6.78	3.10	6.83	6.0
Fs	81.72	72.80	41.37	46.81	–
Qes	0.482	0.58	0.389	0.250	–
Qms	2.851	2.33	6.114	2.91	–
Qts	0.412	0.47	0.362	0.28	–
Vas (L)	4.14	4.82	29.92	33.81	–
L (mH)	0.24	0.48	0.5	0.55	–
dB/W/m	88.56	86.8	89.2	93.25	93.5
VC length	7.2	12.4	11.5	13	–
Gap height	5	4	4	6	–
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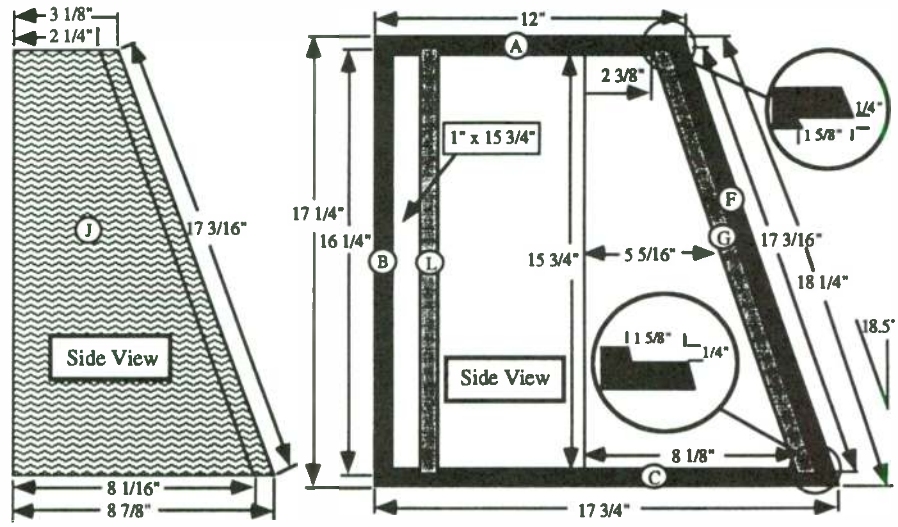

FIGURE 11: Low-frequency enclosure showing side view of cabinet design.

methods to help compensate for this annoying null in the amplitude response, often referred to as a "hole." One often-used method reverses the phase of one of the two drivers, resulting in constant voltage transfer. Other designers have turned to asymmetrical designs in either the high- or low-pass sections.[9,10]

None of these methods provides an ideal solution. A more successful way is to introduce a compensating electroacoustical signal whose transfer fills in the acoustical hole.[11] This involves adding a series inductance and capacitance to the existing high- and low-pass filter network and introducing another driver to the network. I specify the transfer characteristics of this circuit in the next section.

I included midfrequency, high-frequency, and bridging drivers because each provides a range of frequencies meeting the design criteria. The linearity of their response curves over the required high-pass region, the low distortion, high power handling, and wide dispersion of the audio sound spectrum are required for overall high performance.

A list of speakers you can use in an open-baffle design is shown in *Table 3*. I tested those models preceded by an asterisk (*).

An audio spectrum response test of the complete system is shown in *Fig. 4*, where the display resolution of the spectrum analyzer has been set to the 2dB per step scale. This measurement represents the system response at one meter from the center of the baffle with the grille removed.

PART 2, ELECTRONIC NETWORKS

DISTORTION MITIGATION

As I mentioned, you must add a second-order high-pass equalization filter to the existing fourth-order vented-enclosure alignment to achieve the correct sixth-order system alignment defined by equation (11). The behavior of this filter is shown in *Fig. 16*.

To obtain the values for the two time constants T_1 and T_2, shown in equations (24) and (25), use a unity gain amplifier and select the same value for each capacitance so that the ratio of C2 to C1 equals 1. You can determine the peak frequency value, f_{PK}/f_S in alignment 15 in *Table 1*. You can now determine the value of the resistance components from equations (26) and (27).

The attenuation of the lowest frequencies by the high-pass equalizer circuit significantly reduces the effects of Doppler distortion (due to subsonic modulation) and intermodulation distortion. This additional filter not only provides the proper equalization for low-frequency response, but also adds 12dB/octave rolloff to the normal 24dB/octave rate, producing an overall effective rate of 36dB/octave. This increased attenuation reduces the net cone excursion to less than 20% of maximum at approximately 15Hz.

Appendix 1 includes a complete analysis of the transfer function describing this filter. The parameters specified in the transfer function are in *Table 1*, alignment 15. You'll note that the equalization filter has a peak frequency to speaker resonant frequency ratio, f_{PK}/f_S, of 1.07, indicating that the maximum peak should occur at 23.5Hz, with a peak lift of 6dB. Component values to produce these characteristics are listed in *Fig. 5*.

THE BEST ACTIVE DESIGN

Biamplification generally produces lower midrange distortion.[12] This type of distortion often results from bass transient clipping, since low-frequency signals have much higher transient amplitudes than high frequencies.[13] Separating the audio spectrum into high- and low-frequency bands can improve sound quality. Since the design criteria require an equalization filter, it is both economical and convenient to include a low-pass and high-pass active filter on the same circuit board to achieve the necessary frequency division.

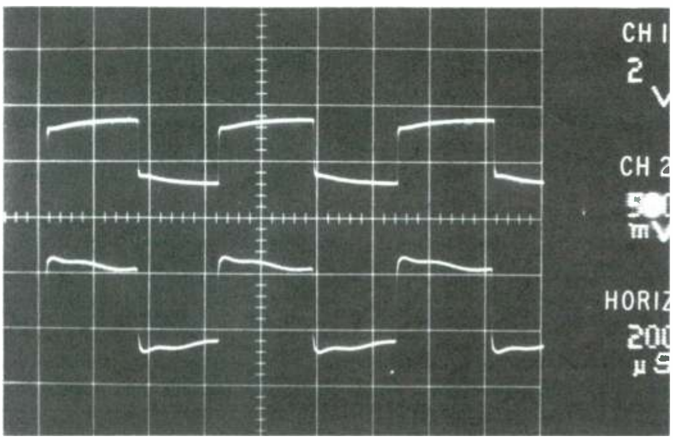
PHOTO 2: Square-wave response with bridging driver.

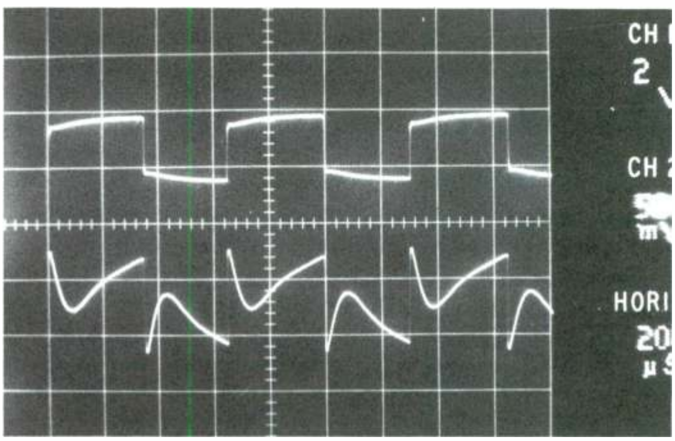
PHOTO 3: Square-wave response without bridging driver.

The crossover design which best exhibits maximally flat magnitude response features a flat voltage and power frequency response with a gradual change in phase across the band and offers a sharp cutoff of −18dB/octave. This is a third-order Butterworth filter[14] (*Fig. 5*), with an attenuation rate of 18dB/octave, a 20kW input impedance, and a crossover frequency of 300Hz. The circuit board layout is shown in *Fig. 14*. The conventionally designed power supply requires ±15V to supply each operational amplifier.

Notice that each operational amplifier in the *Fig. 5* circuit is used in the inverting mode. Adding the high-pass equalizer stage to each filter provides another phase inversion to keep the output signals in phase with the input signals.

The phase characteristics of the active crossover result in a gradual phase shift from 0 to −360 as the frequency increases through the filter, reaching −180 at ω_0. This gradual phase shift is not audible and does not contain a magnitude ripple. Both voltage and power have a flat frequency response.

PASSIVELY SPEAKING

A graph of the passive crossover network response curves is shown in *Fig. 6*. Each LC (inductance-capacitance) network is a second-order filter centered at 1.9kHz. The low- and high-pass sections have a 12dB/octave slope, whereas the bridging section has a slope of 6dB/octave. The bridging speaker therefore must reproduce a range of frequencies two octaves above and below 1.9kHz center.

The midrange driver must have nearly flat response through two octaves *above* crossover frequency, and the tweeter must have nearly flat response for two octaves *below* crossover frequency. Since this criterion exceeds the capabilities of most drivers, the solution is to increase the attenuation of the crossover filters in their stopband region. Using a higher-order Butterworth filter such as a second-order 12dB/octave constant resistance crossover network will solve the problem.

ANOTHER HOLE

Two speakers operating in phase, fed from a second-order network, does not, however, fulfill the need for constant voltage transfer. The amplitude response has a null at the crossover frequency, f_0, which can be seen from the total transfer function in the frequency plane:

$$\frac{V_M + V_H}{V_I} = \frac{s^2 + \omega_0^2}{s^2 + \sqrt{2}\omega_0 s + \omega_0^2} \quad (5)$$

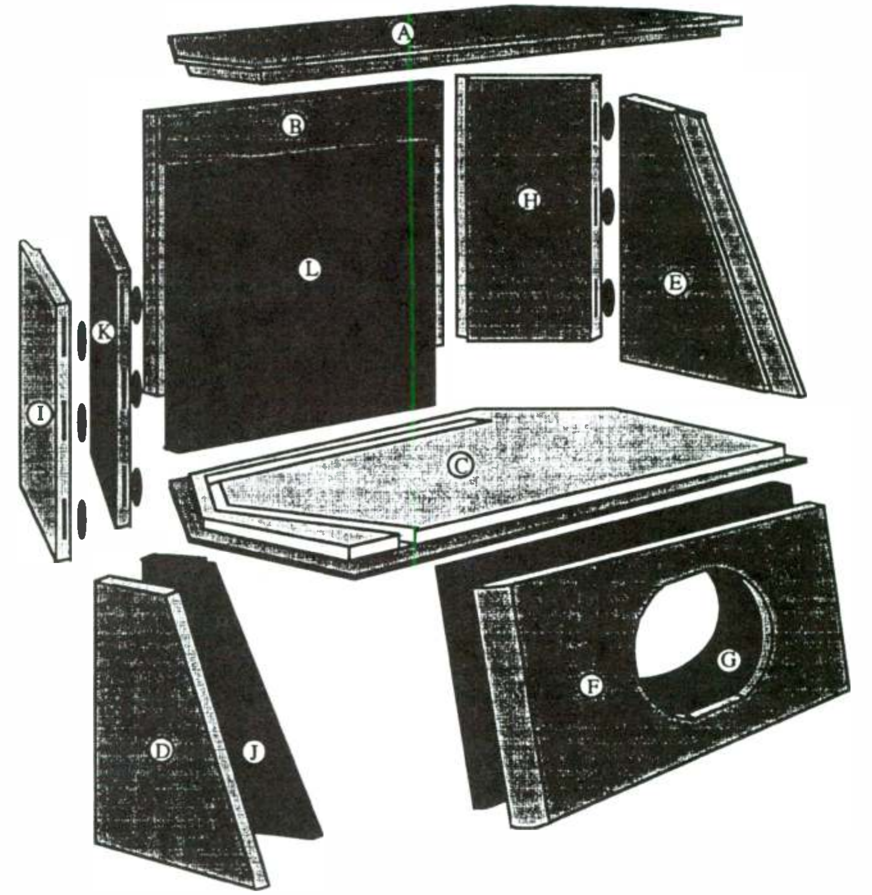

FIGURE 12: Exploded view of panels used in the low-frequency enclosure.

where V_M is the voltage at the low-frequency driver terminals, V_H is the voltage at the high-frequency terminals, and V_I is the applied input voltage.

At the crossover frequency, $s = j\omega_0$, so that

$$\frac{V_M + V_H}{V_I} = 0 \quad (6)$$

which indicates the presence of a hole in the frequency response at the crossover frequency. This hole can be "heard," i.e., the sound in this region is noticeably missing from the full audio spectrum.[6]

BRIDGE CIRCUIT

Instead of modifying the components in the crossover network, another way to achieve constant voltage transfer is adding an electroacoustic signal, whose transfer provides a constant total transfer function. This circuit, called a "bridging" network, has the following transfer characteristic:

$$\frac{V_M}{V_I} = \frac{\sqrt{2}\omega_0 s}{s^2 + \sqrt{2}\omega_0 s + \omega_0^2} \quad (7)$$

where V_M represents the output voltage across the bridging driver. This transfer function may be synthesized to yield the component values shown in *Fig. 7*.

A comparison of this filter's frequency response with and without the bridging network is best illustrated by the oscilloscope readings in *Photos 2* and *3*. The upper trace in each photo represents a square-wave signal measured at the input to the crossover circuit. The lower trace measures the sum of the output voltages at each filter section. In *Photo 2*, the output voltage includes the signal from the bridging network, whereas, in *Photo 3*, the output voltage comes from the more conventional high-pass/low-pass 12dB/octave crossover network.

Notice that the leading edge of the wave form in *Photo 2* suggests that the high-frequency components are accurately represented and predicts an excellent transient performance of this filter network. The lower trace in *Photo 3*, on the other hand, indicates severe problems with the amplitude and phase relationships of the harmonics.

PART 3, CONSTRUCTION

BUILDING THE BOX

After defining the volume and speaker characteristics to reproduce the theoretical

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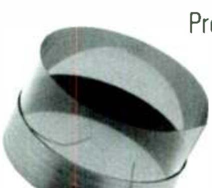

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DUPONT

low-frequency limit, it's time to design an enclosure and midrange baffle which satisfies the model's criteria.

Since 300Hz (the lower cutoff frequency) is desirable for the midrange driver, the area of the front baffle must be large, i.e., the width must be greater than 15″. Therefore, to be reasonably safe, I made the baffle 30″ wide by 16″ high. The low-frequency cabinet is 18″ high, and the overall structure is 34″ tall by 30″ wide (*Fig. 8*).

The cabinet's depth depends on the area of the port (16 in^2) and the length of the duct (33″) from the front surface, which features a 19° reverse slope from the bottom to the top of the cabinet. To enclose this duct along one side and the rear of the cabinet and accommodate the sloping front panel, I designed the enclosure about 18″ deep at its base and 12″ at the top. For precise external dimensions, refer to *Figs. 10* and *11*.

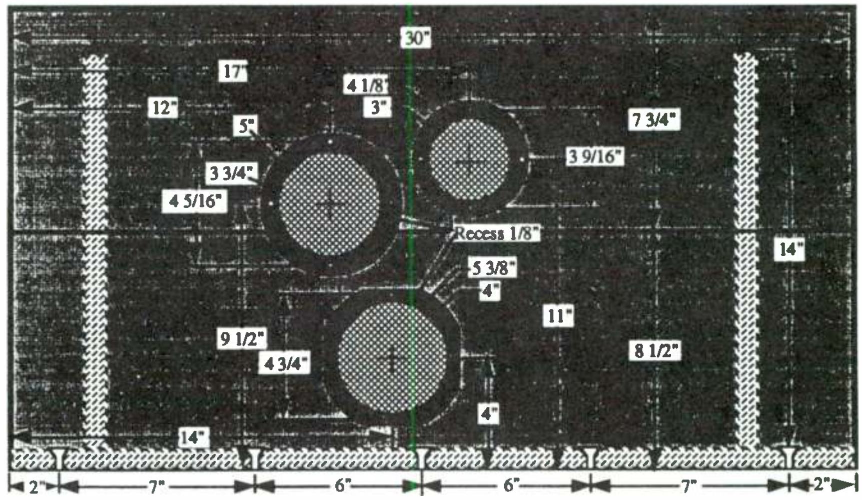

FIGURE 13: Front panel baffle showing location of three drivers.

STYLE

The front surface of the baffle is concave when viewed along a vertical plane. The enclosure is sloped towards the back at approximately 19°, shown in *Fig. 11*; the mid-section is sloped at an 8° angle, shown in Fig. 9; and the baffle is mounted in a vertical plane (*Fig. 9*).

Although this style can cause some difficulties in construction, it does have some benefits. A sloping front panel minimizes the group delay problem. Designing an enclosure with nonparallel sides can help reduce annoying low-frequency reflections and possible standing waves. To further assist in the reduction of unwanted sounds, I doubled the thickness of the driver-supported front panel to reduce the possibility of the front baffle re-emitting internally reflected sounds.

I offset panels I and H at a 15° angle with respect to panels D and E (*Fig. 10*). The width of the back of the box is therefore smaller than the front dimension. This gives the impression that the unit is relatively shallow when viewed from the front.

The diagrams, which illustrate the location of the drivers with respect to the port (*Figs. 8* and *13*), depict only the left speaker. The right one is simply the mirror image of the drawings.

TABLE 3

RECOMMENDED LOUDSPEAKER COMPONENTS (all are 8Ω impedance).

(1) Mid-Frequency drivers

Mfrer	Model	Size	Response	Output
FOCAL	5N313/8	5″	200Hz–8kHz	90dB
*FOCAL	5N413/8	5″	200Hz–5kHz	91dB
ScanSpeak	13M/8636	5¼″	0.2Hz–10kHz	89dB
ScanSpeak	13M/8621	5¼″	0.2Hz–10kHz	90dB
ETON	4-203/25/8	4½″	0.2Hz–5kHz	92dB
KEF	B110	5″	0.2Hz–5kHz	88dB

(2) Bridging drivers

SEAS	76MF/8	3″	0.4Hz–4kHz	91dB
MB-Electronics	MCD51M	2″	0.7Hz–10kHz	88dB
LPG	PMK 51/130R	2″	0.9Hz–10kHz	93dB
MOREL	MDM-75/8	3″	0.3Hz–5kHz	90dB
Dynaudio	D75-AF	3″	0.3Hz–5kHz	88dB
*ScanSpeak	D3806	1½″	1Hz–13.5kHz	91dB

(3) High-Frequency drivers

FOCAL	T120K/8	1″	2Hz–20kHz	91dB
*ScanSpeak	D2905	1″	1.5Hz–22kHz	90dB
CERATEC	C2-22	1″	1.5Hz–22kHz	88dB
MB-Electronics	MCD-25M	1″	2Hz–20kHz	91dB
MOREL	MDT-33/8	1″	1.2Hz–20kHz	91dB
KEF	T27	1″	2Hz–22kHz	89dB

DUCTED PORT

Although the cross-sectional area of the port is relatively small (1″ × 15¾″), it exits into almost three times the area. This semiflanged surface is different than the more conventional round tube.

At the rather low resonant frequency of this design, the air volume velocity at the port output can be very high. A small port area results in high velocities of particle motion at high signal levels, which can lead to peculiar sounds. Test results indicate that a tapered port limits the turbulent effects of the air at the port exit and produces a more satisfying result.

ASSEMBLY

I recommend you use Medite high-density particleboard. With the exception of the front panel, which I doubled in thickness, all construction materials are ¾″ thick. You can use wood screws wherever two surfaces are to be glued, although I strongly suggest dry-wall screws. You can use #20 biscuits to help secure the butt-jointed panels.

To assemble the structure, you must install the internal duct before assembling the outer panels (*Fig. 12*). Using three #20 biscuits, you should first glue together parts J and K. Insert these panels into C and glue into place. You can then install duct panel L onto C, and then glue and screw it to panels K and C. You should

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now temporarily install top panel A and keep it in place until the glue between panels J, K, L, and C has set.

After panels J, K, and L are firmly in place, remove panel A. With #20 biscuits you can now glue together panels D, I, H, and E and mount them to bottom panel C. Glue and screw panel A to the side panels and duct panels, temporarily using the back panel to provide right-angle support for the cabinet while the glue joints are setting.

Once the glue has set, you should remove the back panel and apply a small bead of caulking compound to all joints. Finally, glue in place the back panel B and front panels G and F. However, before gluing together the two front panels, cut the two mounting holes in panels F and G for the driver. Although you could use a sanding disc to produce the required curve on the two front panels (R = 1″ for panel G, R= 4″ for panel F) before they are glued in place, you could also postpone this procedure until you've completely assembled the structure.

Feel free to insert several wooden braces between the top and bottom panels to reduce flexing and vibrations caused by changes in internal air pressure. Add them after you have completely assembled the cabinet, but before you install the driver. In addition, secure three 2″ thick battens of fiberglass wool to the inside of the bottom, back, and side panels. Use #10 bolts rather than wood screws to install the driver.

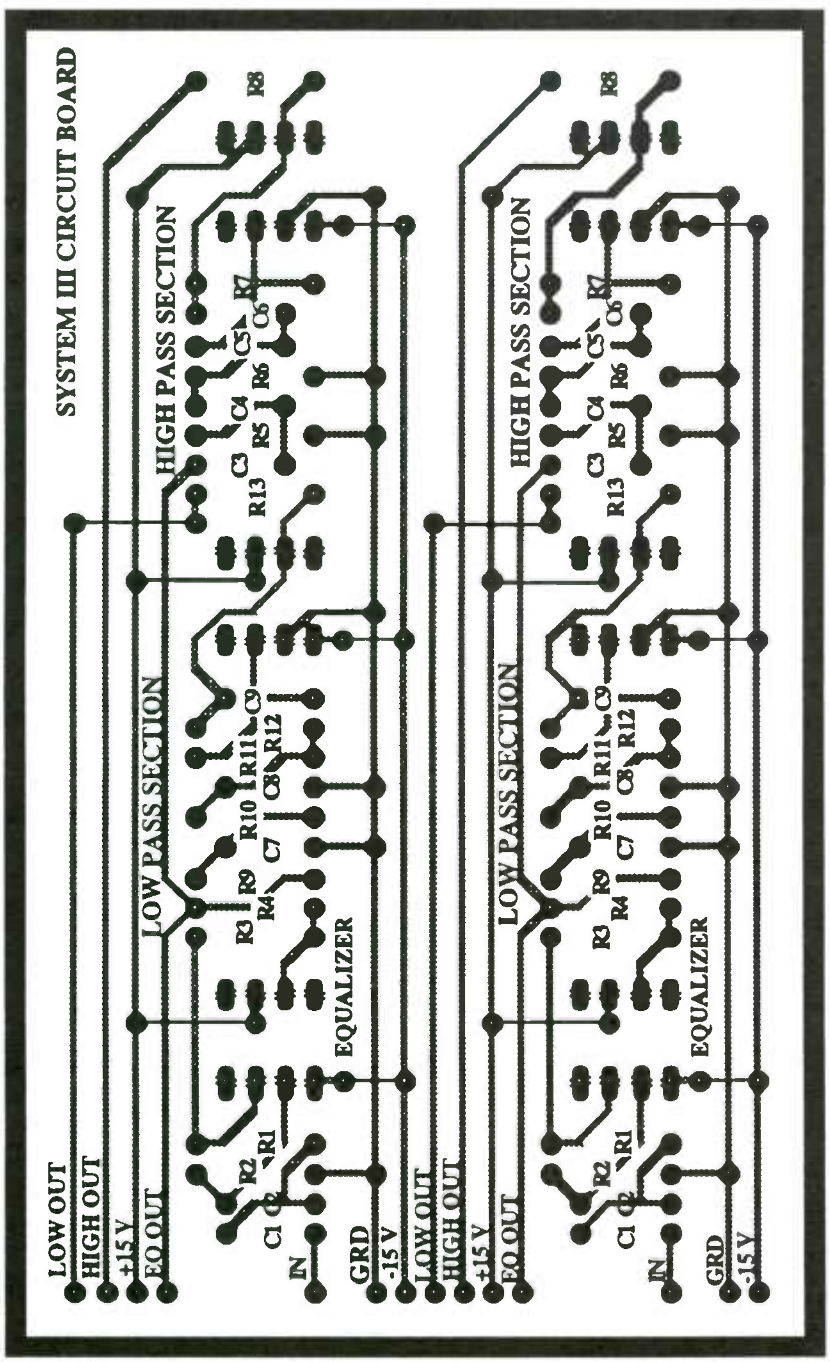

FIGURE 14: Active equalization filter and passive crossover circuit board layout using 2:1 scale.

BAFFLE ASSEMBLY

The top of the bass cabinet supports the upper baffle boards, on which are mounted the midrange and high-frequency drivers. This baffle is constructed of two 30″ boards; one is 8½″ wide and is joined with glue and #20 biscuits to a second, 7½″ wide. Mount the two boards to two baffle support brackets. When assembled, form an 8° angle with respect to each other. This presents a concave contour to the baffle when viewed from the front.

Driver location is shown in *Fig. 13*. The midrange driver occupies the lower position in the baffle, and the tweeter is near the top. The bridging driver is mounted approximately midway between the mid-frequency and high-frequency unit. You'll need to use a router to complete the cutouts and the recess to support the speaker mounting flange for the high- and mid-frequency drivers. The location of the bridging driver, on the other hand, requires you to use a template to assist in routing the proper depth of the recess.

The speaker model numbers are indicated by an asterisk in *Table 3*. Mounted on the rear is the passive crossover network. I conveniently installed the inductors and capacitors for the network in an aluminum die-cast enclosure fastened with screws to the rear of the baffle.

GRILLE ASSEMBLY

The front surface of the completed system contains a grille cloth mounted on a thin (1/8″) plywood board, with appropriate cutouts for the four drivers. Two clips mounted to the bottom of the cabinet and a 1″ diameter grooved cap fastened to the top of the baffle hold the cloth in place. First, install the grille panel in the top grooved cap, then tighten the clips on the bottom to place the entire grille panel under tension. The effectiveness of this design gives the enclosure the appearance of a smooth concave surface from top to bottom.

FINISHING TOUCHES

To enhance the cabinet's appearance, I attached two wooden rails to the sides. I cut one-inch solid rosewood into the shape shown in *Fig. 3*, and then fastened it to the sides of the baffle and lower enclosure using right-angle brackets. Notice that

Text continued on page 26.

NHT 1259

NOMINAL IMPEDANCE	4 ohm
MUSIC POWER	300 W
FREQUENCY RANGE	19-1000 Hz
SENSITIVITY (2.83 V)	90 dB
EFFECTIVE CONE AREA	507 cm2
VOICE COIL RESISTANCE	3.4 ohm
VOICE COIL INDUCTANCE	1.2 mH
VOICE COIL DIAMETER	50 mm
VOICE COIL HEIGHT	34 mm
AIR GAP HEIGHT	8 mm
XMAX	13 mm
FREE AIR RESONANCE	19 Hz
MOVING MASS (MMS)	121 g
FORCE FACTOR (BI)	9.6 TM
MAGNET WEIGHT	59 oz
QMS	2.6
QES	.56
QTS	.46
VAS	190 LTRS
KRM	28.89
KXM	26.67
ERM	.584
EXM	.647
Recomended sealed box volume	70-100 LTRS

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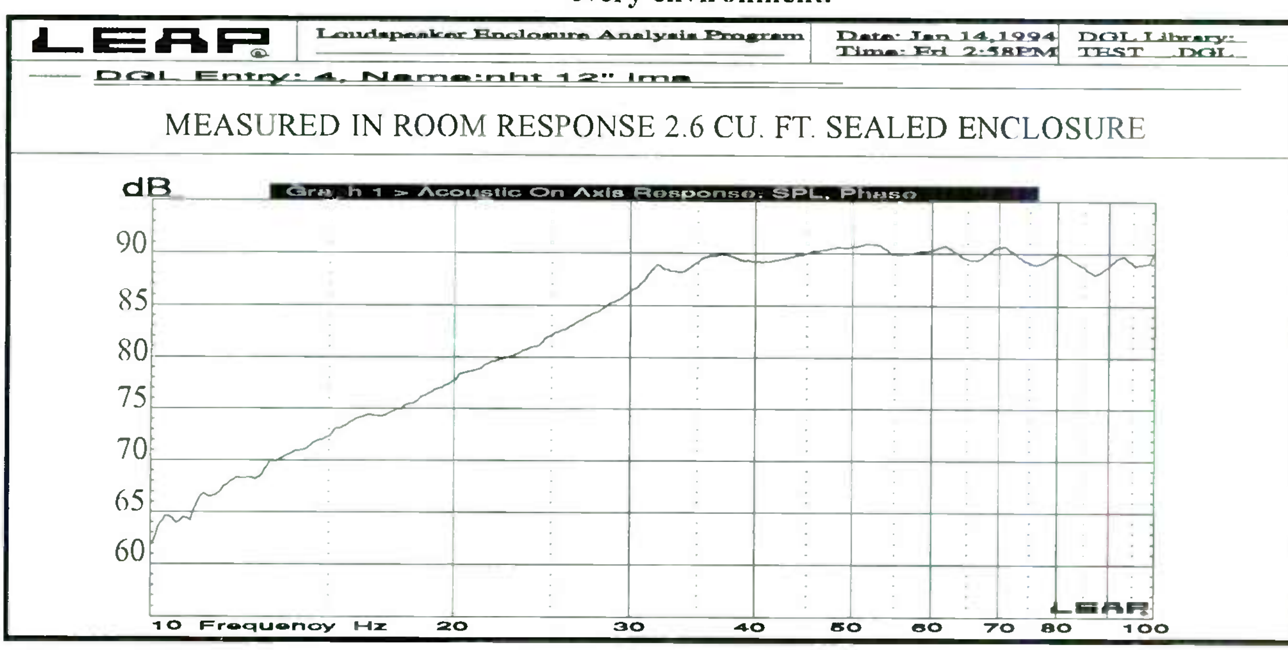

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Reader Service #31

APPENDIX 1 (to "System III Loudspeaker")

SYNTHESIS OF A PRACTICAL SIXTH-ORDER BUTTERWORTH RESPONSE

In a sixth-order high-pass filter, the transfer function contains six negative poles represented as three second-order polynomials, with low, medium, and high damping factors. Since a driver in a vented enclosure has fourth-order high-pass filter characteristics, you can multiply any pair of the above second-order factors to get the required box/vent/speaker properties, and the remaining second-order factor will specify the filter requirements. The system satisfies the overall sixth-order filter characteristic.

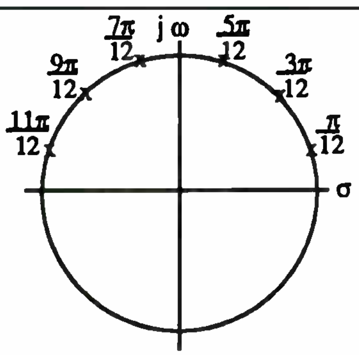

FIGURE 15: Locus of the poles.

POLAR EXPLORATION

Therefore, three families, or classes, of sixth-order Butterworth designs, each with its own electrical filter, compliance, and tuning ratios, are possible. If the lowest of these is assigned to an auxiliary electrical filter and the other two are combined to form the fourth-order expression, descriptive of the enclosure/speaker/vent system, it is possible to design an enclosure whose volume can be set to a minimum size. Provided the amplifier can produce the extra power required by the underdamped filter, and the voice coil can handle the increased power dissipation, the desired low-frequency cutoff can be achieved with a cabinet of modest volume.

Textbook equations showing pole locations of a sixth-order Butterworth low-pass filter are:

$$\sigma_N = \pm \sin\frac{(2n-1)\pi}{2n} \qquad (8)$$

$$\omega_N = \cos\frac{(2n-1)\pi}{2n} \qquad (9)$$

where n = 1, 2, ...2n, and n = 6 (the order of the filter).

In the complex plane, we define the complex frequency variable as $s = \sigma_N + j\omega_N$. Since σ_N is the real coordinate and ω_N the imaginary one, the filter poles are determined by:

$$\sigma_N = \pm\left[\sin\frac{\pi}{12},\frac{3\pi}{12},\frac{5\pi}{12},\frac{7\pi}{12},\frac{9\pi}{12},\frac{11\pi}{12}\right]$$

$$\omega_N = +\left[\cos\frac{\pi}{12},\frac{3\pi}{12},\frac{5\pi}{12},\frac{7\pi}{12},\frac{9\pi}{12},\frac{11\pi}{12}\right]$$

and are located on the unit circle (the positive poles are ignored for discussion of filters) as shown in *Fig. 15.*

The zeros of the sixth-order Butterworth polynomial are:

$-0.2588 \pm j0.9659$,
$-0.7071 \pm j0.7071$,
$-0.9659 \pm j0.2588$.

These represent the real and conjugate poles of the filter and reflex system.

Thiele showed that the pole locations can be distributed between the filter and the reflex system in three ways, yielding three different sets of parameters which describe the same overall system response curve. These "classes" of filters can therefore be described by a polynomial which is the product of the three second-order terms:

$$(s^2+0.517s+1)(s^2+1.414s+1)(s^2+1.932s+1) \qquad (10)$$

EXPLAINING LOSSES

Any pair of these second-order expressions may form the fourth-order term governing system acoustics, and the remaining expression will define the characteristics required of the auxiliary second-order electrical filter. Selecting the polynomial whose coefficient represents an underdamped second-order filter as the electrical portion (Thiele's Class I Filter), you can then combine the transfer function to form the product of two terms, the electrical and acoustic filters. The transfer function of the filter is therefore:

$$(s^2 + 0.517s + 1)(s^4 + 3.346s^3 + 4.7432s^2 + 3.346s + 1) \qquad (11)$$

The fourth-order system response function in equation (1) contained coefficients $\alpha 1$ to $\alpha 4$, which determined the behavior of the filter response. You can now equate these coefficients, $\alpha 1$, $\alpha 2$, $\alpha 3$, and $\alpha 4$, from equation (1) to the values shown in equation (11), i.e.,

$\alpha_1 = 3.346$
$\alpha_2 = 4.732$

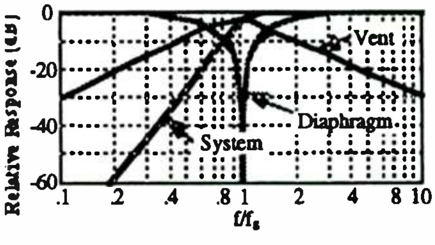

FIGURE 16: Frequency response of vented enclosure.

$\alpha_3 = 3.346$
$\alpha_4 = 1$

Thiele determined that these coefficients are related to the system parameters in the following way:

$$\omega_B/\omega_s = (\alpha_1/\alpha_3) \qquad (12)$$

$$T_0\omega_s = (\alpha_1/\alpha_3)^{1/2} \qquad (13)$$

$$Q_T = 1/(\alpha_1\alpha_3)^{1/2} \qquad (14)$$

$$C_{AS}/C_{AB} = (\alpha_1\alpha_2\alpha_3 - \alpha_1^2 - \alpha_2^2)/\alpha_1^2 \qquad (15)$$

where:

$w_S = 2\pi f_S$ resonant frequency of speaker
$w_B = 2\pi f_B$ resonant frequency of cabinet
Q_T total Q of driver at f_S

and:

$C_{AS}/C_{AB} = V_{AS}/V_B$
C_{AS}/C_{AB} ratio of suspension compliance to box compliance
V_{AS}/V_B ratio of speaker compliance to equivalent box volume

GRAPHING THE RESULTS

Note, however, equations (12) through (15) assume a perfect enclosure: one without losses. The driver diaphragm response shows a null at the vented-box resonant frequency. The action of the driver and vent above and below the resonant frequency is shown in the idealized response curves of *Fig. 16.* The depth of the null at f/f_S is proportional to the total enclosure loss, Q_B.

Small introduced a more rigorous set of equations to explain system losses.[2] He considered three kinds in the enclosure-vent resonant circuit at f_B: absorption, leakage, and vent losses. Since absorption losses are only significant above f_B and vent losses are significant below f_B, then the most significant loss must be related to Q_L. Leakage losses usually give Q_L values between 5 and 20.

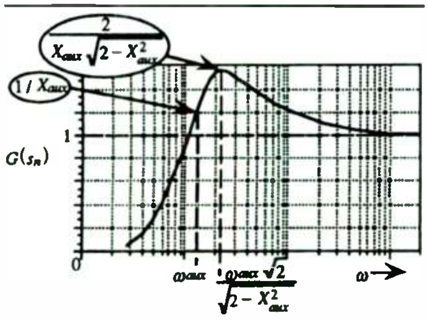

FIGURE 17: Ancillary parameters shown on the high-pass equalization curve.

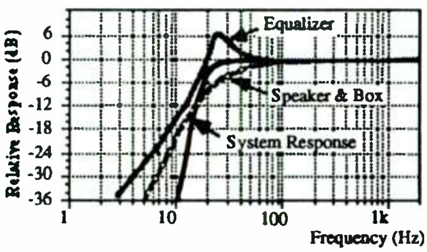

FIGURE 18: Resultant amplitude-frequency response curve.

Imperfections in construction affect the system Q_T and the compliance of the air in the enclosure. Small derived a set of alignment parameters similar to Thiele's which included leakage losses where

$$Q_T = hQ_L/(\alpha_3 h^{1/2} Q_L - 1) \quad (16)$$

and:

$$\frac{C_{AS}}{C_{AB}} = \alpha_2 h - h^2 - 1 - \left(\frac{1}{Q_L^2}\right)(\alpha_3 h^{1/2} Q_L - 1) \quad (17)$$

where $h = \alpha_3/\alpha_1$.

Notice that upon the substitution of $Q_L = \infty$ into equations (16) and (17), both these equations will reduce to Thiele's equations, (14) and (15). In experimental enclosure designs, Q_L values due to air leak losses are about 7.

Substituting the accepted value of $Q_L = 7$ and the coefficients α_1 to α_4 into equations (12), (13), (16), and (17), you obtain the following alignment parameters:

$$\omega_B/\omega_S = 1 \quad (18)$$

$$f_0/f_S = 1f/ \quad (19)$$

$$Q_T = 0.312A \quad (20)$$

$$C_{AS}/C_{AB} = 2.274 \quad (21)$$

The total Q_T relates to several measurable quantities, i.e., the electrical damping due to Q_E, the acoustical resistance due to Q_A of the speaker suspension, the DC resistance of the voice coil, and the amplifier resistance. The total Q can be expressed as the following equation:

$$\frac{1}{Q_T} = \frac{1}{Q_A} + \frac{1}{Q_E(1+R_G/R_E)} \quad (22)$$

Referring now to the electrical filter, the second-order factor, $(s^2 + 0.5176s + 1)$, may be normalized to the high-pass filter function,

$$G(s_n) = \frac{s_n^2}{s_N^2 + X_{AUX}\left(\frac{\omega_{AUX}}{\omega_S}\right)s_N + \left(\frac{\omega_{AUX}}{\omega_S}\right)^2} \quad (23)$$

where $X_{AUX} = 1/Q_{AUX}$ is a constant which is the reciprocal of the filter Q. Sallen and Key[14] have derived the time constants, T_1 and T_2, which define the response curve for this active high-pass filter:

$$T_1 = \frac{X_{AUX}}{2+\gamma}\left[1 \pm \frac{\sqrt{1-4(1+\gamma)(1-K)}}{X_{AUX}^2}\right] \quad (24)$$

and

$$T_2 = 1/T_1 \quad (25)$$

where $\gamma = C_2/C_1$,
K = amplifier gain.

Using equations (24) and (25), you can determine this filter's time constant knowing the damping factor, X_{AUX}; the capacitance ratio, C2/C1; and the gain of the amplifier, K. Using the values for the time constants, T_1 and T_2, and the peak frequency, f_{PK}, you can now determine the final components—resistors R1 and R2—to be designed into the circuit. The value for each time constant is given by

$$T_1 = \omega_0 R_1 C_1 \quad (26)$$

and

$$T_2 = \omega_0 R_2 C_2 \quad (27)$$

The resistance and capacitance values in the equalization section of the electronic filter circuit are shown in *Fig. 4*. They have been determined using a damping factor $X_{AUX} = 0.5176$, and a peak frequency f_{PK} = 23.5Hz. Both γ and K have been assigned a value of one. The specific location of the various constants are shown in *Fig. 17*. This graph represents the function described in equation (23).

To help you visualize the effect of adding equalization, I've included a graph showing the output response of the electronic circuit and the speaker/box combination in *Fig. 18*. The network, which acts as a high-pass filter, shows a peak frequency around 23Hz and a rolloff rate of approximately 12dB/octave.

The speaker/box response curve, on the other hand, shows a more severe rolloff rate approaching 24dB/octave and significant attenuation at the nominal cutoff frequency. However, curve three, shown as the system response curve, illustrates the desired response curve when the previous two responses are added together. Note the very steep rolloff of 36dB/octave for the final response.

Although this graph illustrates the general form of the equalization curve, the actual values for the peak frequency, f_{PK}, and its magnitude in decibels can be recorded from *Table 1*.

SUMMARY

Once you determine the second-order ancillary equalization filter, you can express the sixth-order transfer function, defined in equation (11), as the product of a second-order electronic filter and a fourth-order acoustical mechanical filter. The resultant output of this function is simply the sum of an electronic equalization filter and the speaker response in a specific enclosure volume.

GLOSSARY OF SYMBOLS

- C_{AB} - acoustic compliance of air in enclosure
- C_{AS} - acoustic compliance of driver suspension
- f_{AUX} - cutoff frequency of auxiliary filter
- f_B - resonant frequency of vented enclosure
- f_{PK} - frequency at peak of auxiliary filter
- f_S - resonant frequency of driver
- f_3 - half power (-3dB) frequency of loudspeaker system
- h - system tuning ratio (f_B/f_S)
- L_V - length of vent
- Q_A - Q of driver at f_S (includes non-electrical resistances)
- Q_B - Q of the enclosure box loss
- Q_E - Q of driver at f_S (includes electrical resistance R_E)
- Q_L - Q of the enclosure vent loss
- Q_T - total Q of driver at f_S (includes system resistances)
- R_G - output resistance of source (amplifier)
- R_E - DC resistance of driver voice coil
- s - complex frequency variable ($\sigma + j\omega$)
- S_D - effective surface area of driver diaphragm
- S_V - cross-sectional area of vent
- T_0 - nominal time constant of filter
- V_{AS} - volume of air = acoustic compliance of suspension
- V_B - net internal volume of enclosure
- V_D - driver displacement volume
- X_{AUX} - damping constant of the auxiliary filter ($1/Q_{AUX}$)
- ω_0 - the nominal cutoff angular frequency
- ω_S - angular resonant frequency of loudspeaker

Continued from page 22

each rail—tapered from 1½" wide at the top to 3" wide at the bottom—has the same contour as that of the baffle.

Since the upper baffle is likely to provide a significant amount of reflected sound from a wall near the rear of the speaker, I suggest you add some acoustical damping to the rear of the baffle. *Figure 3* shows the location of the acoustical wedge, which is approximately 4" thick and consists of several layers of foam and fiberglass sheets glued together. The wedge has been shaped to the contour of the rear of the baffle and is therefore thicker at the bottom than the top. It is 22½" long and fits between the baffle supports, which are shown in *Fig. 9.*

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EXPLORING LOUDSPEAKER IMPEDANCE

By Victor Staggs

One of the pitfalls of loudspeaker system design is that a typical driver's input impedance does not look like a simple resistor. Thus, textbook dividing network topologies will not work as you wish, because the driver pushes and pulls current through the network. In addition, the network does not appear to the driver as a perfect voltage source, as would an amplifier. Consequently, the network interacts with any nonresistive component of the driver impedance.

If we use a network optimization computer program, the real driver load is manageable. Even in this case, however, the resulting "optimal" network will still perform poorly if its topology is sensitive to the driver load. An explanation of loudspeaker impedance curve theory and practice will help you accommodate it in dividing network design. I'll also describe techniques that will permit you to make successful designs using either optimization software or your programmable calculator.

As a bonus, I will present some advanced methods for modeling and compensating voice coil inductance. Some of this material has not been published in an American journal before. I therefore urge you to stay with the familiar material until you get to the new ideas.

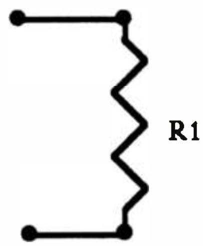

FIGURE 1: Simple model of voice coil impedance.

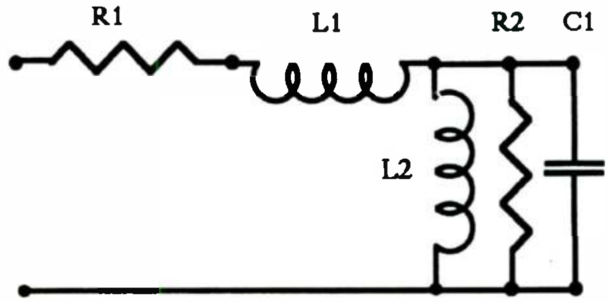

FIGURE 2: Conventional model of voice coil impedance.

WHAT'S WRONG HERE?

Early speaker design articles, and some current books, represent a loudspeaker as a simple resistor in a typical dividing network schematic. While this might be adequate to introduce the concept of dividing networks, simple resistor models are decidedly inadequate for the level of accuracy required in today's speaker design.

Some dividing network topologies are overly sensitive to the actual load presented to them, in particular the high-pass filter feeding a midrange or a tweeter. The pass element feeding the driver may be a capacitor, which causes the driver to ring badly at its fundamental frequency unless further steps are taken.[1] The capacitive output impedance also causes ringing with the voice coil inductance. If the textbook network topology is used, then the actual driver's input impedance can be modified by adding extra components across it, so that it looks like a resistor.

Figure 1 is a resistor, the simplest representation of a driver's input impedance. We will eventually see how to make a real loudspeaker look like this.

THE REAL PICTURE

Figure 2 shows the impedance model of a typical coil-driven, direct-radiator driver, either a cone or a dome speaker. Note the large number of components compared to *Fig. 1*. This model (without inductor L1) has long been used to describe the effect of the voice coil moving in the field of the permanent magnet, in addition to including the coil's simple ohmic resistance.

L1, the inductance of the voice coil windings, is ignored at the low frequencies considered by Thiele and Small.[2,3] The voice coil impedance circuit model is also provided by Beranek, Locanthi, and MacLachlan.[4-6] As it turns out, even L1's inclusion is a simplification when the effects of eddy-current losses in the magnet structure pole pieces are taken into account. For now, though, let's try to understand the reason for all those parts in *Fig. 2*.

Resistor R1 is obviously the voice coil wire's DC resistance. You can measure it

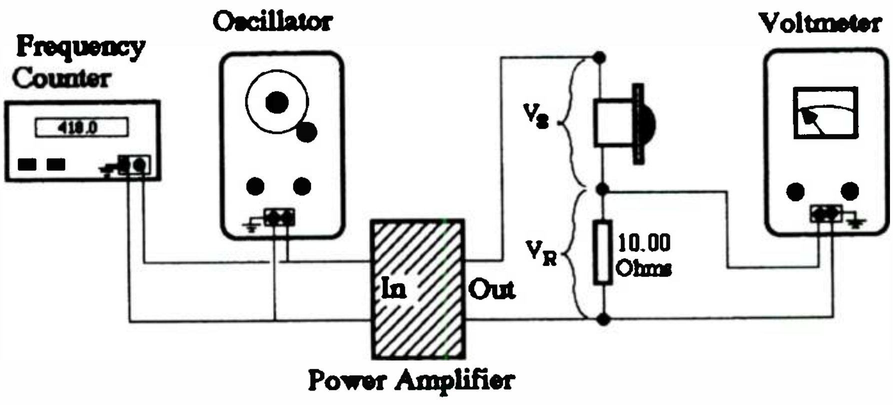

FIGURE 3: Impedance measurement setup.

ABOUT THE AUTHOR

Victor Staggs has recently received a PhD in communication theory and is pursuing work in this field. He learned electronics at Berkeley during the transition from tubes to transistors, and has worked at space physics, nuclear radiation effects, and mathematical physics. He has been most dissatisfied with the sound of loudspeakers among all audio devices, and has devoted most of his hobby time to these creatures.

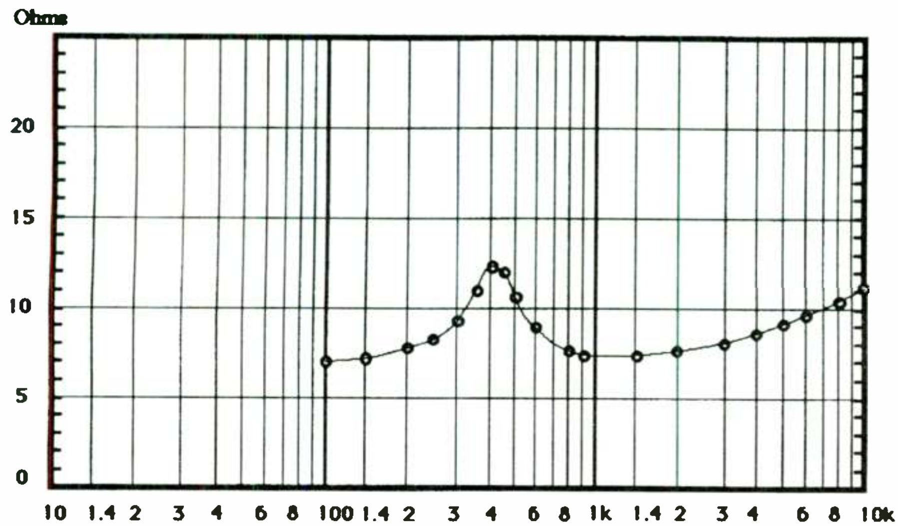

FIGURE 4: Modulus of voice coil impedance versus frequency for Philips AD0211/SQ8.

with an ohmmeter if it is accurate in the region between 3–8Ω. Since the impedances of L1 and L2 are zero at DC, they will not influence the measurement of R1. Furthermore, L1 is usually ignored when measuring R2, L2, and C1 in the remainder of the circuit, since they occur at "low enough" frequencies. L1 actually does influence the measurement of these last three values, so some correction must be made by the fastidious.

R2, L2, and C1 are the reflected mechanical impedances. When the speaker diaphragm moves, it causes the voice coil to cut through lines of magnetic flux in the gap formed by the pole pieces. A back electromotive force (emf) is set up across the coil terminals that is given by E = BLV, where BL is the product of the magnetic (B) field in the gap and the length of the wire in the gap, and V is the coil velocity. At very low frequencies, the diaphragm's motion is controlled by the compliance. For a constant amplitude of input current, the excursion of the cone motion will be constant, but the velocity will decrease with frequency. If the excursion is given by:

$$x = I_{OEXP}(i\omega t)BLC_{MS}$$

then the velocity is its derivative with respect to time, namely:

$$V = i\omega I_{OEXP}(i\omega t)BLC_{MS}$$

Hence, the back emf will increase with frequency, making the diaphragm compliance look like an electrical inductance as seen at the voice coil terminals. The quantity C_{MS} is the mechanical compliance of the diaphragm in its enclosure.

Dividing out the current, I_{OEXP} $(i\omega t)$, from BLV to obtain the voltage-to-current ratio, you obtain the reflected reactance:

$$X_M = i\omega(BL)^2C_{MS}$$

This tells us the electrical inductance due to the diaphragm's net mechanical compliance in terms of this compliance and the BL product:

$$L2 = (BL)^2 C_{MS}$$

Above the diaphragm's fundamental resonance frequency, the motion is mass controlled. For a constant amplitude of input current, the diaphragm's acceleration is constant with driving frequency. Then:

$$a = I_{OEXP}\ (i\omega t)\ BL/M_S$$

and the velocity is the first integral of this, namely:

$$V = I_{OEXP}\ (i\omega t)\ BL/(M_S\ i\omega)$$

Here, M_S is the diaphragm's mechanical mass, including the effect of air-mass loading. The reflected impedance of the diaphragm mass therefore appears as a capacitance at the voice coil terminals, since the back emf will decrease with frequency.

Again, dividing the voltage (BLV) by the current to get impedance, we find that the electrical capacitance, in terms of the total diaphragm mass and the BL product, is given by:

$$C1 = M_S/(BL)^2$$

Resistor R2 is that part of the back emf which is due to mechanical losses in the suspension, to the possible use of ferrofluid damping and the presence of acoustical resistances under the dust cap, or to leakage through a porous dust cap. Some center poles are vented resistively through the back of the driver, either into a damped rear volume or into the speaker box itself. R2's value in terms of the total mechanical resistance is:

$$R2 = (BL)^2/R_M$$

Putting the pieces together, the only circuit which will be inductive below the fundamental resonance, capacitive above it, and resistive right at it, is the parallel combination of L2, C1, and R2 shown in *Fig. 2*.

Thiele and Small provide a prescription for finding all the values except L1. Many manufacturers now include this value in their data sheets, as the voice coil inductance L_{VC}, so you can use it for design purposes. Its use is really an approximation, however, as I will later show.

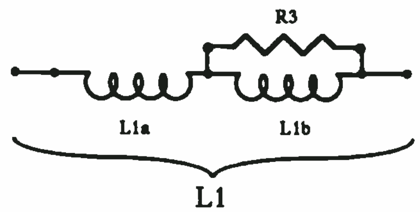

FIGURE 5: Extended model of voice coil inductance.

MEASURING INPUT IMPEDANCE

Not everyone trusts the manufacturer's published voice coil impedance curve. First, it is in graphical form and reading precise numbers from it is difficult. Second, some manufacturers erroneously or for a cost saving print the measurements from another driver, believing them to be close enough. You will wish to know whether your samples are electrically similar to the published curves as a check against hidden flaws. Ample motivation exists for you to measure your own impedance curves.

Figure 3 is a typical impedance measuring setup requiring an oscillator, a frequency meter, a voltmeter, and a 10Ω precision reference resistor. You measure both V_R and V_S at the list of frequencies desired, and then solve for Z_{VC} using the following equation. Since you are ignoring phase here, all you really know is the magnitude of what is theoretically a complex quantity

FORMULAS FOR VOICE COIL IMPEDANCE OF MODEL IN FIGURE 2

Using the assumption that L1 is a simple inductor, the voice coil input impedance consists of the sum of the impedances of R1, L1, and Z_P, the parallel network of L2, R2, and C1. Compute the impedance as follows:

First, calculate the complex impedance Z_P:

$$Z_P(\omega) = \frac{R2\ i\omega/(\omega_0 Q_M)}{1 + i\omega/(\omega_0 Q_M) + (i\omega/\omega_0)^2} \quad (A1)$$

Then compute Z_{VC} by adding in the series impedances:

$$Z_{VC}(\omega) = R1 + i\omega L1 + Z_P(\omega) \quad (A2)$$

The apparent voice coil inductance L_T at radian frequency ω is obtained from the formula:

$$L_T(\omega) = \mathrm{Im}[Z_{VC}(\omega)]/\omega \quad (A3)$$

Equation A3 is valid only in those frequency ranges where the voice coil input impedance is increasing with frequency.

In these formulas, Im[] means to take the imaginary part of the quantity in brackets; $\omega = 2\pi f$, where f is the frequency desired; and $\omega_O = 2\pi f_S$, where f_S is the resonance frequency of the driver in its enclosure.

When using the extended model of the inductance L1, including the effects of eddy-current losses in the pole pieces, one substitutes the following formula for the term iωL1 in Equation A2.

$$Z_{L1}(\omega) = i\omega L1_A + \frac{i\omega L1_B R3}{i\omega L1_B + R3} \quad (A4)$$

IN MEMORIAM

Mr. Bart Locanthi, one of the early contributors to the subject of this article, passed away in early January 1994. I could not begin to list Bart's audio accomplishments here, but I wish to contribute a sidelight about his relationship to this particular article.

In the late 1980s I attended an AES convention in Los Angeles and asked a question of one of the speakers (who covered an aspect of the measured impedance curve) from the floor. At the end of the session, Bart and some other audio luminaries gathered around me for an exchange of opinions.

The speaker had used a simplified model of the impedance curve to derive some properties, but by that time I was already using nonlinear optimization to model the entire impedance curve without simplifications. Bart took an interest in my knowledge of the inductance part.

A few years later, after I graduated from U.C., I visited him in his office in Pasadena. He admitted that he had never solved the problem of how to model the effect of eddy-current losses, nor how to design an effectively perfect zobel network that included them. I later sent him a copy of the manuscript for this article.

To admit that he had not solved this problem, and to learn about my (and others') solution to it, showed Bart to have a desire to learn and an open mind, both admirable qualities. May we all have them.

with a magnitude and a phase. R's value is either 10Ω or its accurately measured value.

$$\left|Z_{VC}(\omega)\right| = \frac{RV_S}{V_R} \qquad (1)$$

Note in *Fig. 3* that V_S is not taken with reference to ground, so you must use a voltmeter with a floating ground. If you use an AC-powered voltmeter, and one side of AC is grounded to the chassis, then the measurement must be taken twice: once to find V_R as shown and once to find V_S after reversing the resistor and speaker positions to put one side onto ground. The oscillator and voltmeter grounds will thus be identical for all the measurements.

You should consider the frequency counter a necessity, as the dials on an oscillator are unlikely to be very accurate. In addition, it will be difficult to place the dial at the identical frequencies if the AC-grounded voltmeter is used and the list of frequencies is scanned twice.

A typical measured impedance curve, obtained for the Philips AD0211/SQ8 is shown in *Fig. 4*. This 2" dome midrange driver, no longer in production, has a nominal 8Ω input impedance.

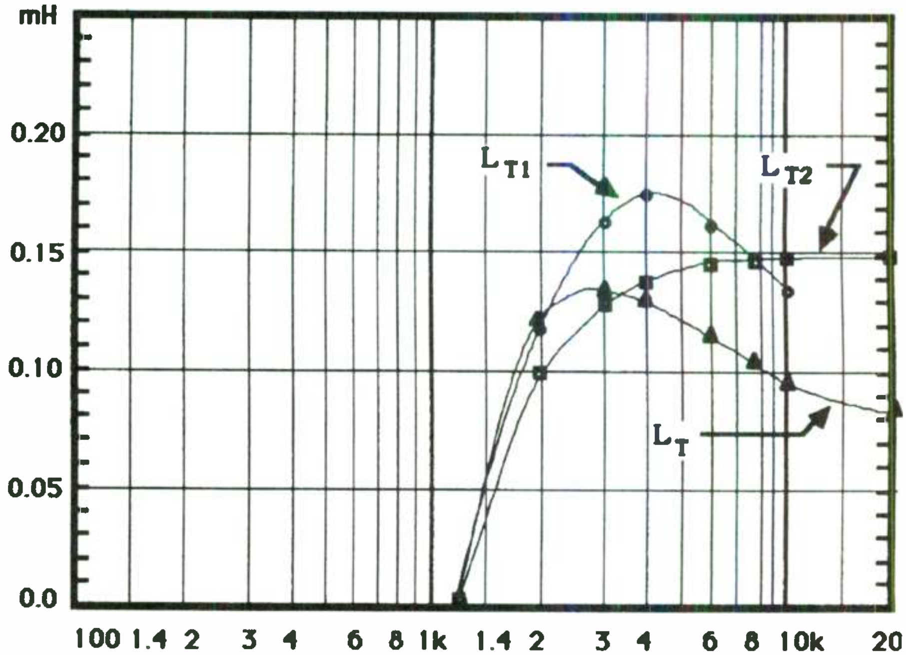

FIGURE 6: Apparent inductance versus frequency, Philips AD0211/SQ8.

With the impedance measuring setup, you can apply the T/S method of finding the resonance frequency f_S, the mechanical quality factor Q_M, and the DC resistance R_E. You will use these to obtain the values of R1 = R_E, R2 = R_{ES}, L2 = L_{CET}, and C1 = C_{MEC}, simplifying their notation slightly. The following equations give the impedance equivalent circuit values directly in terms of the resonance frequency in radians per second, $\omega_O = 2\pi f_S$, and the measured Q_M.

$$R2=R(\omega_0)-R1;\quad L2=\frac{R2}{(\omega_0 Q_M)};\quad C1=\frac{Q_M}{(\omega_0 R2)} \qquad (2)$$

I have derived a simplified formula for finding L1 using the measured impedance curve. You must first find the frequency at which the input impedance attains its minimum value, above the peak due to the fundamental resonance. This minimum is the result of another resonance between L1 and C1, at a frequency where the L2 reactance is small but cannot be ignored.

First, compute the parallel impedance of the circuit composed of L2, R2, and C1:

$$X_P=\frac{R_2 i\omega_M/(\omega_0 Q_M)}{1+i\omega_M/(\omega_0 Q_M)+(i\omega_M/\omega_0)^2}$$

At the radian frequency ω_M, the minimum impedance is resistive, so you must cancel out the imaginary part of X_P by an equally imaginary reactance from L1, which will have the opposite sign. We can therefore write:

$$ImXL1 = \omega_M L1 = -ImX_P$$

Solving for L1, we obtain:

$$L1_M=\frac{-ImX_P}{\omega_M} \qquad (3)$$

The imaginary part of X_P, ImX, is obtained from a calculator that works in complex arithmetic. This value of L1 is referred to as $L1_M$ because it applies at the frequency ω_M.

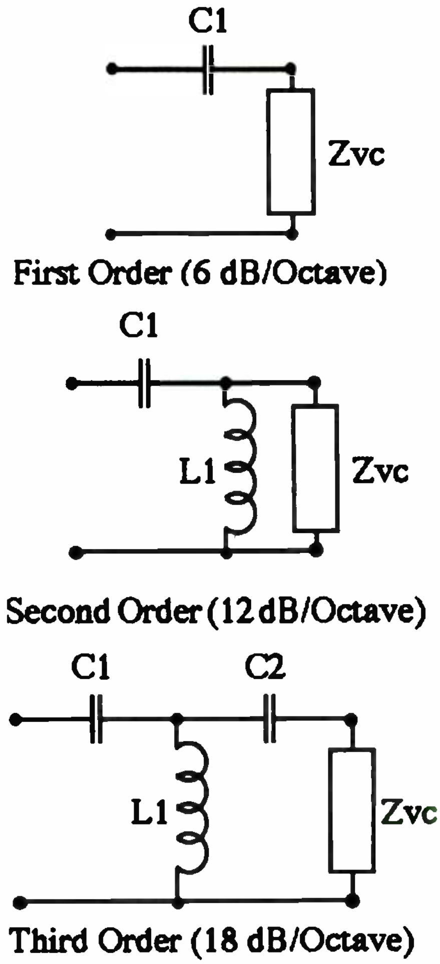

FIGURE 7: High-pass networks.

Note that the component L1 is inaccessible to direct measurement, since we cannot insert real probes into a theoretical equivalent circuit. We must find L1 by indirect means, with help from a calculator. The value of inductance measured directly at the voice coil terminals—for instance, by a complex impedance bridge—will be variable with frequency and only indirectly related to L1. I will later refer to this apparent inductance measured at the terminals as L_T, which will be a positive quantity only when the impedance curve is increasing with frequency.

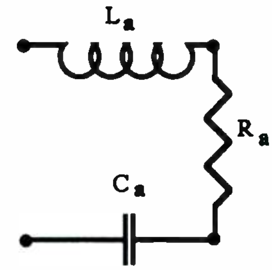

FIGURE 8: Impedance compensation network for fundamental resonance.

IMPEDANCE CURVE INTERPRETED

With your own measurements, the voice coil circuit parameters are known. Lacking these, you can use the manufacturer's curve to find the approximate parameters, provided the numbers can be scaled off the curve to apply the T/S method. Manufacturers sometimes provide the parameters. Certain obvious facts can be gleaned from the curves.

For example, you will know when a driver is bona fide and not defective. In case there is doubt as to whether or not the unit is ferrofluid damped, you can examine the height of the resonance peak. If the mea-

Continued on page 35

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Robert M. Bullock III, Robert White

NEW THIS ISSUE

Bullock & White's venerable all-star has now been improved even more! This worldwide best seller's regular capabilities and features have included sealed, vented, and passive radiator designs; four concurrent design capability; on-line help; 8088 through Pentium support; no coprocessor required, but recommended; DOS real-mode application; auto-detect CGA, EGA, VGA display; pull-down menu; inclusion of atmospheric effects on design; port length calculations; inclusion of box, port, and absorption losses in analysis; filter-assisted alignments; equalized alignments through 8th order; and automatic file save/recall. Graphing capabilities in BoxModel have included maximum and relative SPL; voice coil impedance magnitude and phase; acoustic phase response; vent air speed; driver piston excursion; and passive radiator excursion.

Now, transient response and group delay have been added as new graphs. In addition, other new features in BoxModel include series and parallel compound and isobaric designs; graphics printer dumps, support for over 300 printers; generation of optimized flat alignments; display of coefficients of transfer function; and redesigned user interface. Original article from *Speaker Builder* 6/90; reprint included. IBM, 1 × 3½" DS/DD or 1 × 5¼" DS/HD—PLEASE SPECIFY DISK SIZE. Also available:

SOF-MOD4BXGD	BoxModel 3.0 demo (usable as credit toward later purchase of full package)—PLEASE SPECIFY DISK SIZE	**$5.00**
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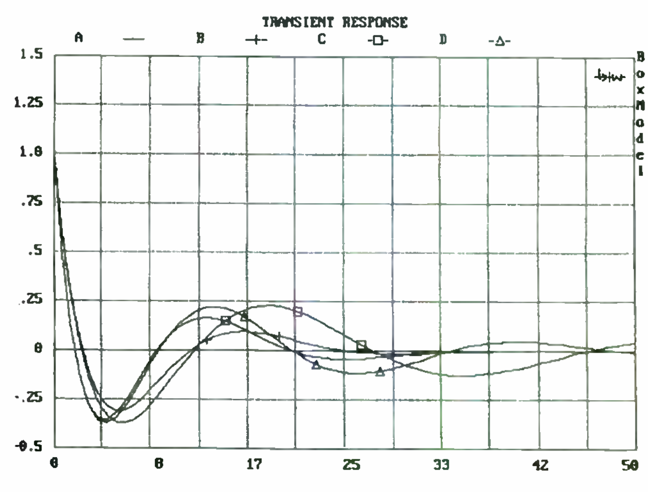

WHAT'S A SYNTHESIZER? — BKHL7

Jon F. Eiche — **$5.95**

A whole new jargon of musical technology has sprung up—terms such as "digital synthesis," "sampling," "RAM cartridges," and "MIDI." This book was written to explain, simply and briefly, the whats, hows, and whys of the new musical technology. Although it doesn't cover absolutely everything, it does provide you with a basic grasp of the most important equipment and ideas, and prepares you to ask informed questions. 1987, 61pp., 6 × 9, softbound.

WHAT'S A SAMPLER? — BKHL6

Freff; K. Kimball Holland, editor — **$5.95**

You've heard the results in recordings, movie scores, and TV commercial jingles: sounds so realistic, you'd swear they were the real thing—yet they're not. They're being reproduced by electronic musical instruments. Welcome to digital sampling, the land of "aliasing" and "munchkinization." This book provides the basics of how to make and use samples, as well as definitions and explanations of words such as these. Whether you're serious about sampling or merely curious, this book will be of benefit. 1989, 42pp., 6 × 9, softbound.

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AUDIOCAD PRO 4.2 SOF-AUD1B3G SOFTWARE $299.00

Michael Bernhard Uibel

NEW FROM GERMANY! AudioCad Pro 4.2 is a loudspeaker development package—with English dialogue and manual included—specifically designed for the professional user. The most popular such program in Germany, it contains a wide range of simulation and evaluation routines for the development of loudspeaker boxes and passive crossovers. Features include a dBase-compatible databank with over 800 drivers already loaded; measurement databank for storing amplitude, impedance, electrical, and acoustical phase patterns; manual data input option; measuring system interfaces for AMSPC, AudioLab, AudioTestBoard, DAAS3, DSA, IMP, LMS, MEPEG, and MLSSA; response simulation using T/S parameters; enclosure simulation for infinite baffle, bass reflex, and single and double vented bandpass; TL, tractrix, and exponential horn calculation without simulation; graphical enclosure construction and parts lists; crossover simulation for 2- to 4-ways; graphical editing of circuit diagram; bass, treble, and bandpass calculation for Bessel, Butterworth, Chebychev, Linkwitz, or Compromise; crossover optimization; export functions for all simulations; support for 9- and 24-pin, inkjet, and laser printers; on-screen help; mouse-controlled user interface; and Windows compatibility. Requires PC-AT (286, 386, 486, Pentium) with 640K main memory; 2.5Mb free hard-drive space; VGA; MS-compatible mouse; and MS-DOS 3.0+ or compatible. 1 × 3½″ DS/HD. Further information available upon request. Also available:

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OMP/MF MOSFET POWER AMPLIFIER MODULES

BK Electronics

NEW FROM ENGLAND! Now available for North America's 110V, the highly renowned OMP/MF Series II of MOSFET power amp modules offers no-compromise, high-quality audio power at exceptionally realistic prices. All modules are supplied as one complete built and tested unit. These amps are used internationally by professionals in leisure, disco, public address, and studio applications, to name but a few, where they are well known for their quality, reliability, and price/power/performance ratios. In additional to often being the choice of professionals, this series is equally well suited to the domestic applications of discerning audiophiles or the needs of industrial or academic researchers. Exceptional clarity and realism are the hallmarks of this sophisticated design, resulting in huge sales for these units throughout Europe (almost 1000 sold in Germany alone). In order to ensure reliability for the state-of-the-art MOSFET output devices, generous power supplies are employed, incorporating a toroidal transformer. Glass fiber printed circuit boards are utilized, mounted on a a solid aluminum chassis with extensive heatsinking. An onboard low-voltage supply and drive circuit are also provided to operate an optional (see next column) LED VU meter.

There are currently five models available, offering a choice of output powers from 100Wrms up to a staggering 1000Wrms. There are two versions of each module available, offering a choice of input sensitivity and bandwidth (PLEASE SPECIFY): STANDARD, with 500mV input sensitivity, 100kHz bandwidth; or PEC (Professional Equipment Compatible), with 775mV input sensitivity, 50kHz bandwidth. Further information available on request. Purchasing options available:

KM-14A/100S	OMP/MF100 with 115/105Wrms output into 4-/8-Ohm loads, Standard sensitivity	$139.95
KM-14A/100P	As above, with PEC sensitivity	$139.95
KM-14A/200S	OMP/MF200 with 215/150Wrms output	$219.95
KM-14A/200P	As above, with PEC sensitivity	$219.95
KM-14A/300S	OMP/MF300 with 310/220Wrms output	$269.95
KM-14A/300P	As above, with PEC sensitivity	$269.95
KM-14A/450S	OMP/MF450 with 455/305Wrms output	$399.95
KM-14A/450P	As above, with PEC sensitivity	$399.95
KM-14A/1000S	OMP/MF1000 with 1050/725Wrms output	$699.95
KM-14A/1000P	As above, with PEC sensitivity	$699.95

SUPER ACCURATE ADD-ON VU METER KM-15A $16.95

BK Electronics

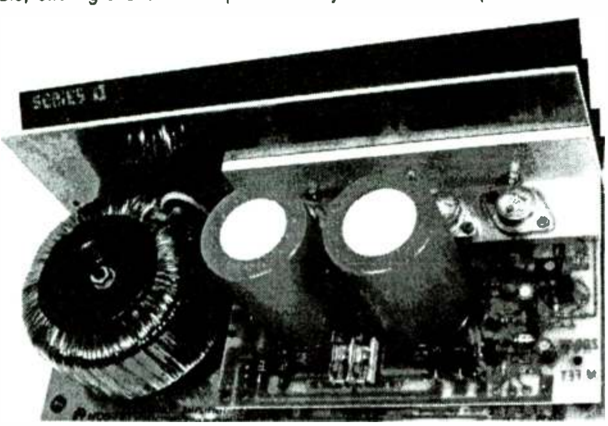
Compatible with the OMP/MF series MOSFET power amp modules above, this device from England has a very accurate visual display employing eleven display LEDs—seven green and four red—plus an additional LED on/off indicator. Sophisticated logic control circuits for very fast rise and decay times. Tough, molded plastic case with tinted acrylic front, easily connected to amp modules with only three wires. Tolerance: within +/-1dB. Input impedance: 10k. Size: 83.5mm × 27mm × 45mm. Power supply required: 12V-13VDC @ 40mA. Frequency response: 20Hz to 100kHz. Sensitivity: 775mV - 0dB. Dynamic range: -20dB to +3dB (23dB).

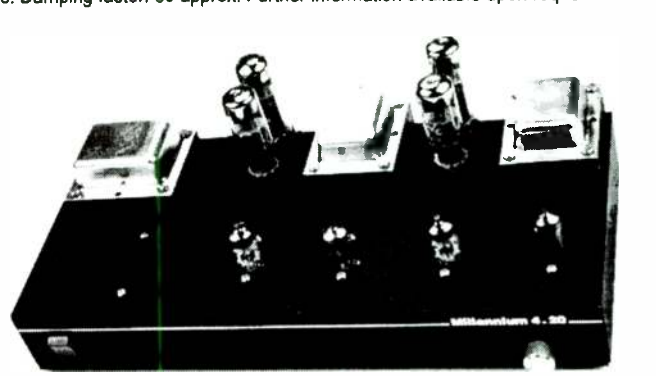
NEW MILLENNIUM 4-20 20W TUBE POWER AMP KIT KM-16 $399.00

Mike Holmes/Maplin Electronics

NEW FROM ENGLAND! This classy new product from the famous folks at Maplin is just one more example of why their kits are known throughout Europe for quality and care in preparation. Now configured for North America's 110V, the Millennium 4-20 is a stereo amp, complete with power supply, which uses a non-hybrid, thoroughly traditional four-tube design typical of the high quality power amps of the '50s and '60s. It offers an output of up to 20Wrms into 8 Ohms, operating in Class AB1, and has a total distortion figure of around 0.4%, with an input sensitivity of 220mV. Using 1 × EF86, 1 × ECC83, and 2 × EL34, each channel features a push-pull power output stage preceded by a phase-splitting stage, along with a front end stage which can incorporate a negative feedback loop to set the gain and control the overall circuit. Simplified construction using PCBs. Great for home stereo systems, small-scale public address systems, or musical instrument amplification, this beauty's sound will draw raves—and you built it yourself!

Specifications: Type—Class AB1 "Ultralinear." Maximum output power—20Wrms. Gain—30dB. Input sensitivity—220mV for 20W output. Frequency response—25Hz to 30kHz +/-0.5dB @ 20W; -3dB @ 75kHz @ 20W; <10Hz to <40kHz +/-0.5dB @ 1W. Risetime (1kHz square wave)—4 microsecs. Overshoot and ringing (1kHz square wave)—approx. 10%. Phase shift error—20 degrees @ 20kHz. Signal-to-noise ratio—89dB. Output noise (input grounded), hum—<3mV peak; white noise—<2mV peak. Harmonic distortion—0.05% (0.1% @ 27W). Intermod distortion—0.7% of carrier (1% @ 27W) [may vary slightly]. Beat-note distortion—0.25% (0.3% @ 27W) [may vary slightly]. Output impedance—<0.2 Ohms. Damping factor: 50 approx. Further information available upon request.





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Continued from page 32

sured acoustic frequency response is available, you can use this as a further test of existing ferrofluid damping, since the output at the resonance frequency will be much reduced compared to that of a driver without such damping. (In the usual direct-radiator driver, Q_M is much higher than Q_E, due to low losses in the suspension; with a ferrofluid-damped driver, Q_M is about as low or lower than Q_E.) The rise of impedance with frequency will provide some measure of the driver's usefulness at high frequencies. Four-layer voice coils tend to have more inductance than two-layer ones; they also weigh more and reduce the high-frequency output.

While the measurement setup is still warm, plot the measured impedance curve *(Fig. 4)* and see whether some frequency ranges need further measurement. Small bulges sometimes appear in the curve which are not predicted by the T/S model. Some stamped-frame drivers have the magnet structure mounted on the back of the basket in such a way that the basket's entire back surface can flex.

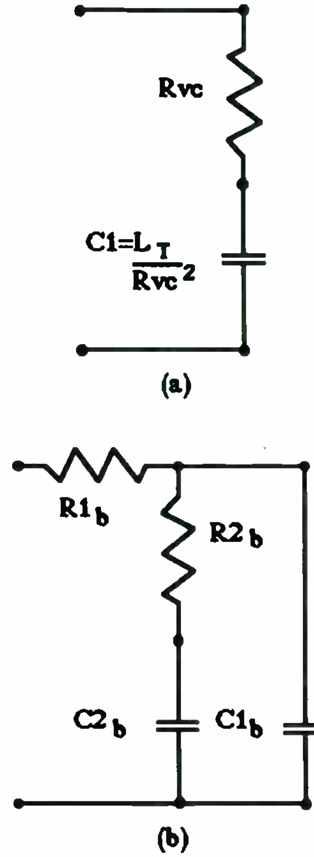

FIGURE 9a: Original zobel network for simple voice coil inductance compensation. FIGURE 9b: Zobel network including effects of eddy-current losses in pole pieces.

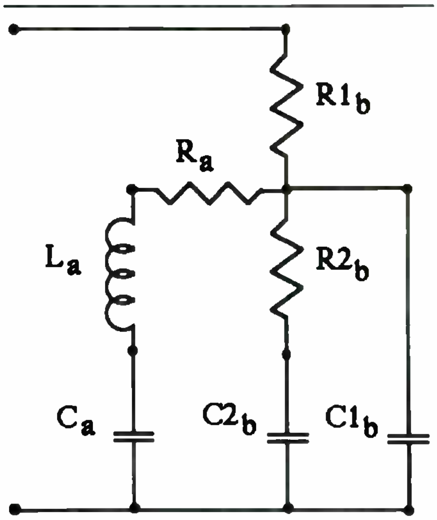

FIGURE 10: Zobel network for entire impedance curve of driver.

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Note: Some of the above preprints have been printed in the JAES as articles. Preprints may be ordered from the Audio Engineering Society, 60 E. 42nd St., Rm. 2520, New York, NY 10165-0075 at $5 each, postpaid.

At some frequency, the magnet mass will resonate on this sheet metal compliance in antiphase to the voice coil, and this will cause a measurable bump in the impedance curve. You may wish to coat the inside basket spokes with damping material, and to brush some into the crack between the magnet assembly and the basket's rear surface. If the bump occurs at a high frequency, it may be caused by diaphragm or surround resonance. A significant bump may mean a large peak or dip exists in the driver's acoustical output.

You must closely examine the manufacturer's acoustical measurement to determine the driver's suitability, and you may contemplate making your own acoustical measurements. The advantage of doing this at discrete frequencies is that no smoothing is involved as with swept or impulse testing, so potentially you can learn more from sine-wave testing.

APPARENT INDUCTANCE SIMPLIFIED

A crude idea of the apparent inductance L_T can be gathered from the manufacturer's impedance curve or from your own measured one. You would present this inductance to an external circuit, such as a dividing network, and use its value when neutralizing the driver's inductance with a zobel network.

As a first approximation, assume that the input impedance (above resonance) is given by Equation (4), where the resistance is taken as that at the minimum above the fundamental resonance, and will be assumed not to vary with frequency. The inductance can then be found from Equation (5). We call this approximate value L_{T1}. Here, $R_M = [Z_{VC}(\omega_M)]$, the magnitude of the voice-coil impedance at ω_M.

$$Z_{VC}(\omega) = R_M + i\omega L_{T1}(\omega) \qquad (4)$$

$$L_{T1}(\omega) = \frac{\sqrt{|Z_{VC}(\omega)|^2 - R_M^2}}{\omega} \qquad (5)$$

The result of this approximation shows the terminal inductance to be a function of frequency. At high frequencies, the real L_T—and perhaps this simplified one—ultimately decreases with frequency. This is the externally observable effect of eddy-current losses in magnet structure pole pieces. Actually, the resistance also varies with frequency, so we have made a simplifying assumption.

The values of L_{T1} obtained from Equation (5) will agree with true values obtained from a bridge measurement up to frequencies about $2 \times \omega_M$. This accuracy will not be quite good enough to allow L_{T1} to be used at the crossover frequency, which, depending upon the driver, may be much higher than ω_M.

A second approximation to L_T is to find $L1_M$ from Equation (3), use this in *Fig. 2*'s impedance model, and then evaluate the expression:

$$L_{T2}(\omega) = \frac{l_M Z_{VC}(\omega)}{\omega} \qquad (6)$$

This approximation will agree with ideal values up to a frequency slightly higher than L_{T1}.

AN EXTENDED MODEL

At higher frequencies, eddy-current losses in the pole pieces act as though they shunt part of L1 with a resistor of unknown value, which reduces L1's back emf considered alone. We therefore adopt a model that consists of two subinductors in series—L1a and L1b—which add up to L1, with L1b shunted by the unknown resistor R3 *(Fig. 5)*.

Substituting the extended model for L1 into *Fig. 2*'s circuit diagram and computing the new input impedance curve is simple. The formulas are given in the sidebar. It is also easy enough, though somewhat tedious, to try different values of L1a, L1b, and R3, and to replot the impedance curve. Good starting values for L1a and L1b should add up to $L1_M$, since this is what they sum to at ω_M.

This cut-and-try method is an effective substitute for computerized nonlinear optimization, and predicting an impedance curve that closely matches the measured one doesn't take long. When a match occurs, you will know the complete voice coil model, published by Thiele in an Australian journal.[7,8]

Eugene Zaustinsky published an AES preprint about this model.[9] It can be inserted into PSPICE or one of the other programs that computes impedances from lumped circuits. You do not need to use a calculator or write special coding to do the calculation.

Vanderkooy provides the physical theory of eddy-current losses as they affect voice coil inductance.[10] Wright gives a functional fit to the phenomenon, but requires you to measure the inductance and resistance separately with a phase sensitive bridge.[11] This

Skaaning Loudspeakers

These loudspeakers are a perfect blend of accurate performance, robust construction, and elegant industrial design. The SK170 is a mid bass driver for use in a sealed system, a woofer for small vented systems, or a bandpass subwoofer. Applications include recording monitors, compact hi power audio.

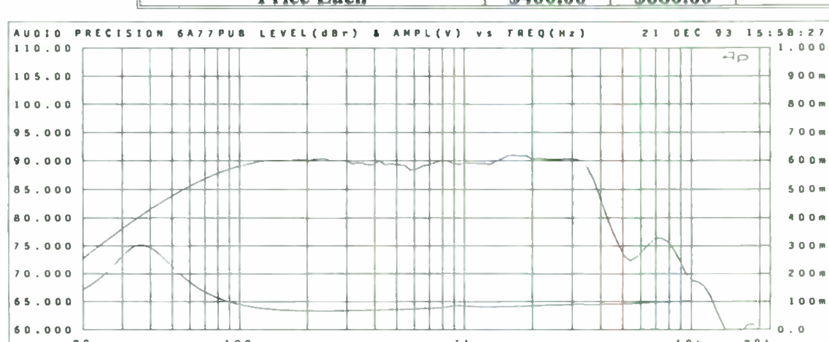
The SK300 combines the dynamic range of professional stage woofers and the accuracy of hi-fi woofers. It is designed for use as a subwoofer in Audio, Home Theatre, and Professional applications. It is also available as a Dual Voice Coil unit at 4 ohms per coil: SK300-344DVC.

To show the awesome capabilities of Skaaning Loudspeakers, Madisound has developed the Mimesis system for exhibition at a recent Stereophile show. The Mimesis has extremely wide dynamic range, clean transients, and unequalized bass output extending below 25 Hz in a sealed cabinet.

Ejvind Skaaning was the founder and guiding force behind the well respected Danish companies of Scanspeak and Dynaudio. Mr. Skaaning's engineering skill and innovative designs are fully expressed in his new line of loudspeakers.

Technical Data	Symbol	SK170-308	SK300-304	Unit
Nominal Impedance	Z	8	4	Ω
Resonance Frequency	Fs	37	18.38	Hz
Power Handling Nominal	P	200	350	W
Sensitivity (1W/1m)	E	90	91	dB
Voice coil Diameter	Ø	77	77	mm
DC Resistance	Re	5.49	3.4	Ω
Voice Coil Inductance	Lbm	.243	.557	mH
Voice Coil Length	h	20	30	mm
Former		Aluminum		
Wire		Aluminum		
Number of Layers	n	2	2	-
Basket		210mm Ø Die Cast Aluminum	350mm Ø Die Cast Aluminum	
Cone Material		Mineral Filled Polypropylene		
Surround Material		Rubber		
Magnet Size		170mm OD x 20mm H, 77mm ID	170mm OD x 24mmH, 77mm ID	
Force Factor	BL	7.4737	9.5235	NA^{-1}
Height of Magnet Gap	He	10	10	mm
Linear Excursion peak	Xmax	5	10	mm
Suspension Compliance	Cms	1161.44	1052.72	μmN^{-1}
Mechanical Q Factor	Qms	1.704	1.722	-
Electrical Q Factor	Qes	0.365	0.308	-
Total Q Factor	Qts	0.300	0.262	-
Moving Mass	Mms	15.98	71.24	g
Effective Piston Area	S	0.017	0.0515	m^2
Equivalent Air Volume	Vas	47.66	396.48	Ltrs
Leap Motor Constants	Krm	160.172	4.037	mΩ
	Kxm	448.881	31.023	mH
	Erm	0.308	0.720	
	Exm	0.140	0.540	
Price Each		$400.00	$680.00	

SK170-308



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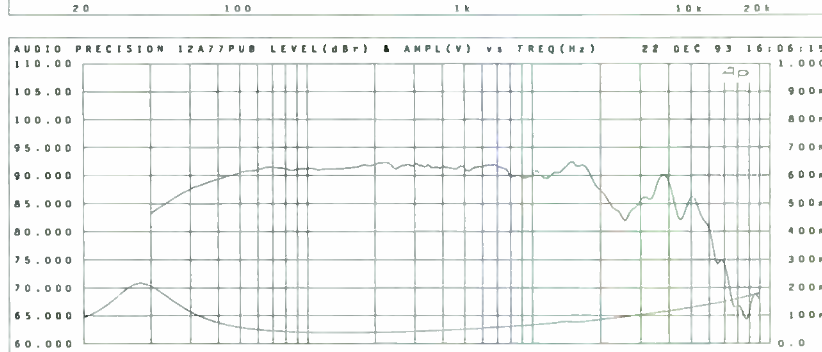

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entails more equipment than merely an oscillator and a voltmeter. Elliott describes very accurate voice coil impedance measurements to demonstrate eddy-current losses.[12]

Wright's approach is superior in that it will work over a very wide frequency range. When modeling a large woofer, Thiele's circuit may need an additional splitting of the remaining simple inductor into a smaller unit plus another shunted by a resistor at high frequencies.

Figure 6 shows the approximations of L_T for the Philips dome midrange unit whose impedance curve is shown in *Fig. 4.* The values of apparent inductance L_{T1} and L_{T2} are included.

The third curve uses a computer-optimized fit to the model in *Fig. 2*, using *Fig. 5*'s extended inductance model to predict L_T. Theory tells us that a close fit will model both phase and amplitude nearly perfectly, so we will use this curve as the standard of perfection. You can see that L_{T1} seriously overpredicts L_T above 2kHz and that L_{T2} yields a fair prediction of L_T up to about 4kHz, above which it overpredicts.

Only a driver with an apparent inductance L_{T2} would be perfectly compensated with a conventional zobel network, but real drivers behave like L_T. Using a simple, single-capacitor zobel network to compensate L_T will not be valid much above or below the crossover frequency.

If you have access to a computer program for designing dividing networks, use the complete voice coil impedance model to load it. Either the optimizer or an auxiliary program will compute the load impedance and store it for use during the optimality search. If you know that the optimal dividing network will be improved when some part of the speaker's complex input impedance is made to resemble a pure resistor, you can add this compensation before optimizing.

NEUTRALIZING THE PEAK

Figure 7 shows some common passive tweeter networks. The first- and third-order ones present the load with a capacitive source, which will not pass much current at the resonance frequency. Thus, there is little self-damping at the driver's resonance, with a resulting large amplitude peak.

Some commercial dividing network designs use a simple series LC network across the driver's terminals to provide a zero impedance source at its fundamental frequency. The disadvantage is that resonances at other frequencies are set up by the presence of this high-Q circuit, making its use a two-edged sword. Furthermore, it bends the frequency response away from the desired ideal curve below the crossover frequency.

A better tactic is to add a series resistor to the LC network so the driver's resonant peak is compensated to the DC value of resistance. This will leave the inductive rise at high frequencies, which you can either ignore or compensate. Usually, the subjective impression of sound is better from a neutralized network than from one with undamped resonances attempting to cancel each other out over a wide frequency range.

To derive unique values of R, L, and C that neutralize the driver input impedance at resonance, note the expression for the driver input impedance without L1 but with an unknown impedance in parallel with the driver terminals. Then simply set the net input impedance equal to the speaker's DC resistance, solve for the complex value of the compensating impedance, and guess which circuit it represents.

When this is done, the results are as given in Equation (7). These values of R, L, and C are for a series network that you install in parallel with the driver. You could call this a generalized zobel network for the driver's fundamental resonance. The circuit is shown in (*Fig. 8*).

$$R_A = R1\,(R1/R2 + 1);\ L_A = R1^2C1;\quad C_A = L2/R1^2 \quad (7)$$

Using this network allows the optimizing program to deal only with a load consisting of the driver's DC resistance, plus the voice coil inductance with eddy-current losses. We are not through neutralizing, however.

NULL THE INDUCTANCE

The well-known form of the zobel network is shown in *Fig. 9a.* It supposedly neutralizes the voice coil's apparent inductance, assuming that this is due to a simple inductor. Normally, you find the value of the driver's apparent inductance (L_T) at the crossover frequency and then neutralize that value. The inductance will be over-neutralized at frequencies above the crossover frequency and under-neutralized below it.

Figure 9b shows the extended form of the zobel network for neutralizing the voice coil inductance. It consists of two capacitors and two resistors.

The circuit values are:

$$R1_B = R1;\ R2_B = R1^2/R3;\ C1_B = L1_A/R1^2;\quad C2_B = L1_B/R1^2 \quad (8)$$

This circuit is applied in the same manner as the original zobel network, namely across the voice coil terminals without the other network for neutralizing the fundamental resonance.

If the inductance and fundamental resonance zobels are used together, they must be modified slightly and combined to satisfy theoretical requirements. *Figure 10* shows the combined network circuit. All the old values are used except for R_A, which now becomes:

$$R_A' = R_A - R_E = R_A - R1_B \quad (9)$$

The combined network, wired in parallel with the voice coil terminals, will create an input impedance that is R1 across the entire audio frequency range. The driver will then be referred to as fully neutralized.

DIVIDING NETWORK USE

The fully neutralized driver can be used with any dividing network topology, but is most helpful with the first- and third-order ones shown in *Fig. 7.* If you use an optimization program, the values of R_A, L_A, and C_A for neutralizing the fundamental resonance can be made free variables with starting values equal to the ones given here.

Alternatively, the driver can be used with the compensation network fixed and the appropriately neutralized impedance fed in as data, in order to simplify the optimization procedure. Depending on how they will interact with your dividing network, this approach can be used for either the voice coil inductance or the fundamental resonance or both.

Not all of you own a network optimization program. With this article, you have the option of using a fully neutralized driver. The textbook dividing network topologies, assuming a resistive load, will actually work. Their only departure from perfection will be that the drivers intrinsically have a frequency response that is not flat, and this will affect the summed acoustic response. You can compensate for this to some extent by choosing different crossover frequencies for the two drivers that cause their responses to overlap somewhat.

Careful experimenters can use a low-pass acoustic response to model the lower-frequency driver and a second-order high-pass response for the upper-frequency one. These will allow you to predict the acoustic output, a technique explained very well by Zaustinsky.[13] With a little patience in calculating the summed frequency response, you will arrive at the best value of crossover frequency overlap to use. This results in a quite decent frequency response without the use of a computer.

For reliable prediction of the summed response, you must model the amplitude and phase of the drivers' own acoustic responses, or obtain measurements of them.[14] The acoustic response of drivers far from the crossover frequency has a profound effect on

Continued on page 63

IMPCYCLING

By Bill Waslo

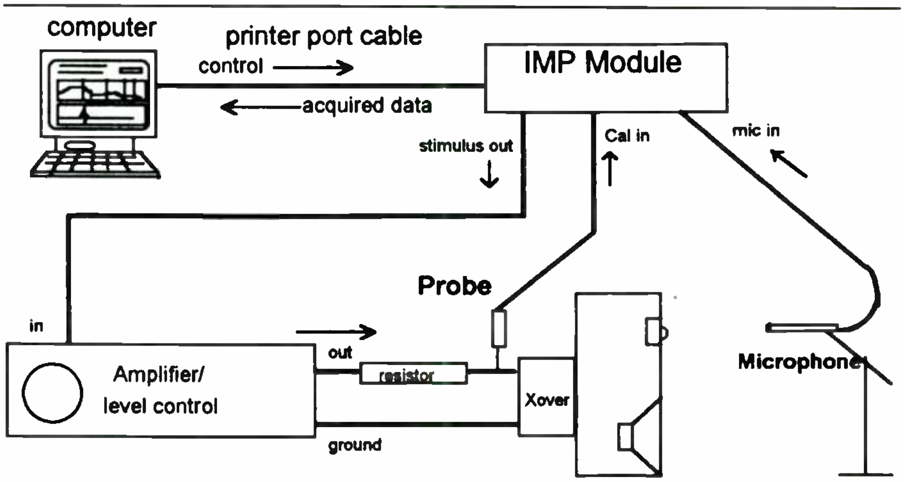

FIGURE 1: Setup for using [auto_Measure Acous] while adjusting the crossover components "live." The resistor protects the amp.

The ability to measure a speaker's or a system's performance is a tremendous asset in developing custom equipment. Without data from measuring gear such as warble tones-and-microphone, pink noise-and-RTA or FFT audio analyzers (such as, oh...IMP), you can only base your designs and procedures about the available components on theory and assumptions. But the "loop" is never closed: the effects of changes aren't quantified and there is no clear verification. It is extremely difficult and time-consuming to determine cause and effect relationships or verify a theory based solely on listening tests.

With good measuring gear, you can use an informed optimization process: measure components, analyze and design, assemble, measure again to verify the design, revise the design, implement changes, measure again, and so on. When the system is the way you wish, carefully listen to it to make sure that what you planned sounds as you expected; if not, revise your design requirements and optimize again. Good computer-assisted engineering (CAE) software such as CALSOD or LEAP reduces the number of tests by improving the match between designed and measured results.

A BETTER IDEA

Instead of making the measurements between changes, what if you could watch the results change *while* you adjust? For instance, wouldn't it be great to watch your speaker's frequency response vary while you work on the crossover? Or try different components in a Zobel network while you watch the impedance curve flatten? Or see how a tweeter's output blends with a midrange while you vary the tilt of the baffle? Well, read on.

I wish to thank Pete Folberth of Cincinnati, OH, for both the concept and the method of accomplishing the following idea. I call it "cycling," and it is included under the [auto_Measure Setup] menu of IMP/M (version 2.00 and higher). Since this was added just before software release, the documentation for it in IMP Help is sketchy at best; this article aims to fill in the gaps.

The idea is simple: A switch in the IMP software causes certain auto_Measure operations, to repeat continuously ("cycle") until you press the Esc key. To keep the latest result plot (for example, the frequency response curve of "auto_Measure Acous") on the screen while the next acquisition is being processed, all graphic updates except the last of each operation are suppressed. This not only avoids distracting you with the busy gyrations of the acquisition and the cal process, but it also considerably speeds up the process by eliminating time-consuming screen drawing.

The screen displays an ongoing series of sequential plots of only the finished data. (Note for IMP non-users: "auto_Measure" operations are canned measurement procedures the computer conducts after you have set up the basics, which include equipment connection, level settings, software settings, and sampling time window selection.)

On a fast computer (such as a 33MHz 486) at high sample rate, this cycling process is similar to watching high-resolution frequency response graphs in real time (updates every second or two). Also, using MLS with this method helps you obtain good immunity to noise without having to average. Using this process, I've adjusted crossovers for acoustic response while conversing on the telephone.

THE GOOD, THE BAD, AND THE LAZY

IMP-cycling has its good points and its bad points. Obviously, fine tuning a system, level-matching drivers, and adjusting other continuous parameters can be done with extreme efficiency. Armed with a bag of clip-leads, various capacitors, inductors and low-value resistors (and a basic understanding of filter operation), you can often patch together a working crossover in minutes, instead of hours.

Maybe more importantly, IMP-cycling is fun. Watching the effects of felt and foam being moved around on a baffle (for edge diffraction effects), for instance, is just fascinating.

But you can also become spoiled and lazy. You may arrive at a decent crossover network, for instance, entirely by trial and error. Without doing any theoretical work, you may miss a better approach or a design which involves a component value not in your bag-o'-crossover-parts. Then again,

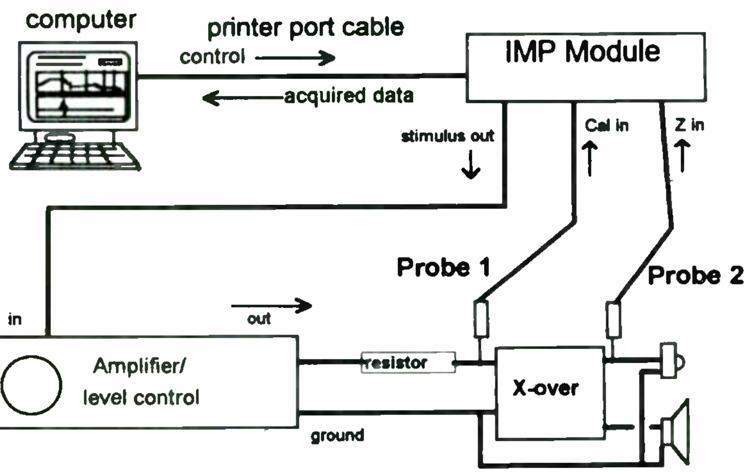

FIGURE 2: Setup for using [auto_Measure Elect] while adjusting crossover components "live." Hookup for measuring tweeter signal shown.

such free-form designs can make good starting points for later computer-based design optimizations.

You also must watch for out-of-band energy going into the drivers (particularly the tweeter), which is not always obvious when you're looking at the acoustic response alone. It's a good idea to also check the electrical signal into the tweeter (use [auto_Measure Elec]) before you play loud music through the system.

HELPFUL HINTS

Turning on cycling is easy. Just enter [* auto_Measure Setup cYcling On], which will affect any auto_Measure operations of types "Acous," "Elec," and "Impedance" ...until cycling is turned off again.

Here are some guidelines for best results when cycling:

- Keep the SIZE parameter as small as practical to minimize FFT times, and thus increase the update rate.
- Operation will be fastest when sMoothing (under the [Display Format] menu) is set to None. If smoothing is required, use as little as possible (i.e., tWelfth octave will be quicker than Sixth, which is quicker than Third, and so forth).
- If you can, acquire and declare a cal before you start, and then use it for following operations, rather than generating a new cal for each acquisition. To instruct the auto_Measure routines how to deal with cal, use [* auto_Measure Setup Cal_source]. For noncritical measurements and faster results, you can also elect not to use cal at all.
- If you are making quasi-anechoic acoustic measurements (editing to remove echoes), be sure to set up your time-editing markers, viewing the time domain via [F1], before you start the [auto_Measure Acous] process.
- As always when using auto_Measure, set all amplifier and IMP input levels beforehand.

• If you are developing a crossover and plan to look at the electrical signal provided to each driver, be sure to connect the drivers in your design so that one terminal of each is grounded. This is necessary because the IMP probes measure only ground-referenced voltages. *Never* connect the probe grounds to any nongrounded points (in fact, because probe grounds are redundant, you seldom need to connect them).

If you are adjusting electrical networks connected to your power amplifier output, you should insert a resistor (3Ω or more) between the power amplifier hot terminal and the network. This prevents shorting the amplifier or accidently connecting a nasty reactance directly across the amplifier output terminals. This is the normal setup for impedance measurements (such as adjusting Zobels) which use a series reference resistor, so in such cases the hookup is the same as usual.

If you wish to watch frequency response while you "mess" with the crossover—and a possibility of connecting bad impedances to the amplifier terminals exists—you should add the resistor to the setup. Use at least 3Ω and try to keep it below 10Ω (I recommend 5Ω). Note that you must connect the cal probe (#1) on the other side of the resistor (not at the amplifier output), as shown in *Figs. 1* and *2*. This connection prevents the resistor from affecting the measured data (IMP will correctly measure the change in the signal from input to output of the unit under test), yet will still protect the amplifier from errors.

One problem with using this resistor is that the true cal response (what the probe sees) may change as the speaker system impedance is altered by your crossover modifications. You can deal with this possibility in one of two ways: Generate a new cal with each acquisition or just take an initial cal (with the resistor in place, of course) and ignore variations. If you choose to temporarily overlook the variations, you can always get another cal at the end for a more precise look. Don't forget to take the resistor out before the listening test.

When adjusting driver levels using L-pads or the controls on an active crossover, you do not need to include the safety resistor, of course, since there is no risk to the amplifier. The same is true for adjusting an equalizer, although you may want to look in the IMP Help facility under [Transform Xfer] for another technique for adjusting equalizers.

A GOODIES GRAB BAG

If you often need to homebrew a quick crossover, I suggest you put together a kit of common crossover parts. You'll be surprised at how many people will want you to tweak a home-made or modified speaker.

Your kit should include a bag of 1μF film capacitors, which you can inexpensively obtain from many sources. These can be paralleled to cover a wide range. Get a bunch of 5 or 10μF caps for the larger values. Inductors can be reduced in size by removing turns (you can use IMP to measure the values, too), but you probably won't wish to reduce inductance by more than about 70% to keep the Q up. You should cover a good range by including a few each of 0.2mH, 0.4mH, 1mH, and 3mH values. Alternately, you can use one of Kim Girardin's multi-tapped inductors, as described in *SB* 3/94, "A Multitap Air Inductor," p. 16. You can also include an inexpensive selection of power resistors (1W or more) in the range of 0.33Ω to about 20Ω, which will cover most purposes.

Continued on page 64

Part 1

THE LINEAR-ARRAY CHRONICLES

By Philip Witham

In *SB* 3/94 ("The Linear-Array Sound System," p. 28) Philip Witham set forth a radical new approach to sound recording and reproduction. His plans for a Linear-Array system stirred readers' imaginations and elicited many suggestions and comments. Encouraged by this response, Witham has proceeded with his testing and experiments. This article—in the form of Witham's replies to reader feedback—updates his efforts and outlines his goals for implementing his design. *SB* will chronicle his progress, and we encourage further reader reaction to his unusual sound reproduction proposal.—Eds.]

SONAR SAMPLE

Mr. G.R. Koonce of Liverpool, NY, offered several useful comments about the Linear Array. Mr. Koonce is a recently retired submarine SONAR electronics engineer who "spent 33 years building SONAR systems." He says, "You can't build speakers for a hobby and design sonars for a living without wondering what would happen if sonar techniques were applied to a home sound system." His systems—with up to 1,000 channels in line, cylinder, sphere, and 2-D flat area arrays—make my proposal look rather pedestrian in comparison. Bandwidths range from "low audio frequencies to several megahertz," with problems such as Doppler shift (from the motion of the sub), varying speeds of sound (due to water temperature differences), EMP protection from nuclear attack or lightning, and intense electromagnetic noise problems, to name a few. I've extracted or paraphrased some of his ideas and comments in the following:

In sonar work my "image aliasing" is termed "false targets," and was not thought of in terms of Nyquist sampling theory. The results are the same, however, and he confirms my rough math on the needed array spacings. In general, you need less than 180° of phase shift between adjacent channels at a particular frequency, and array spacing of three or more per wavelength is preferred. He notes that on the "receive" (microphone) end, some sound will come from wall reflections at the recording site at or near "end-fire" angles (90°, or nearly directly to the sides of the mike array). He believes these reflections will be a problem for my proposed 1.125″ array spacing, where false targets (images) will be generated for frequencies over 6kHz.

I completely agree with these numbers, especially since I've run a simulation of the system in the program Mathcad (see below), and learned what Mr. Koonce knew a long time ago: The record situation is more critical than the playback one. If strong high-frequency sources are located close to the mike, the situation approaches the worst-case "end-fire," or 90° case. Eliminating this problem would require at least four times as many channels, which I believe is impractical. He agrees with me that careful attention to the recording setup, including using absorbent panels to tame particularly pesky wall reflections, should help.

More interesting is his comment that "the system will not perfectly reproduce the source information unless the receive array (mikes) and transmit array (speakers) have the same beam properties. Not only must the array spacings match, but the local directivities of the elements must match." He makes an important statement here, and I hope the operative word is "perfectly." In comparison, two-channel stereo is about 2% "perfect" (a meaningless number, completely fabricated) at reproducing sound wave fronts. We live with it, nonetheless.

Perhaps the microphone array must be composed of vertical ribbons with figure-eight polar patterns for best results. Or who knows what experiments will show us? I think that simple omni or cardioid mikes teamed with nearly any drivers in a linear array will immediately demonstrate a big

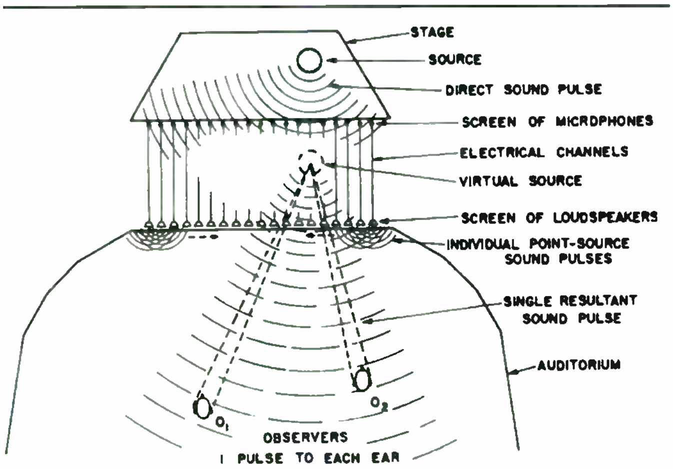

FIGURE 1: "Ideal stereophonic system. A very large number of very small microphones and loudspeakers would give a perfect reproduction of the original sound" (from *Journal of the SMPTE*).

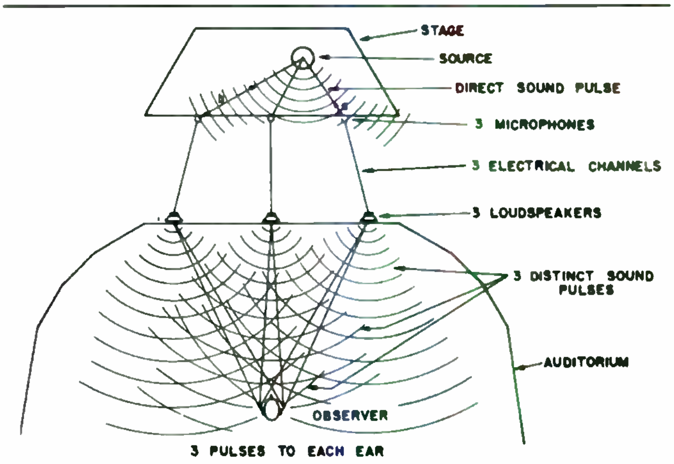

FIGURE 2: "Actual 3-channel stereophonic system. A practical stereophonic system gives a multiple reproduction of the original sound which the observer interprets as coming from a single source" (from *Journal of the SMPTE*).

improvement in imaging over stereo.

He also suggests trying a vertically interleaved array of small (3" or so) dynamic drivers instead of an electrostat, and also using the same drivers as microphone *and* speaker. I am afraid that last would sacrifice good sound quality (especially flat frequency response) for theoretically perfect imaging. But G.R. *has* convinced me to build a cheap and dirty proof-of-concept system using little dynamic drivers (more on this later).

'50s FLASHBACK

Thanks to Dennis Green of Detroit, MI, for providing some interesting references, which I tracked down at the local university library. He notes that William B. Snow describes a Linear Array in his paper, "Basic Principles of Stereophonic Sound," in the *Journal of the SMPTE*, Vol. 61, Nov. 1953 (see *Figs. 1* and *2*). This paper was included in the "Bibliography on Audio, Sept. 1970," distributed by Klipsch and Associates, Inc. The idea was immediately rejected as impractical, but cited as a good one, nonetheless.

Snow wrote: "It has become customary to describe stereophonic reproduction as follows: A screen consisting of an extremely large number of extremely small microphones is hung in front of the sound source. Each microphone is connected to a corresponding extremely small loudspeaker in a screen of loudspeakers hung before the audience. Then the sound projected at the audience will be a faithful copy of the original sound and an observer will hear the sound in true auditory perspective."

To summarize your points and the gist of these papers:

1. Three channels across the front is much better than two, especially in depth imaging.

2. Six-track discrete (Dolby SR-D 5.1 channel) is a marketing *fait accompli*, and a much more practical near-term goal than a Linear Array. I agree.

In my opinion, gnashing of teeth over data compression (Golden Ears versus significant improvements in spatial cues) is correct but unnecessary, since it will only be a few years before you can have 5.1 channels in an uncompressed (or only mildly compressed) format. *EE Times* magazine recently reported that Sony has doubled the bit capacity and rate for CDs by using both edges to encode separate bits. Their apparent intention is to record four discrete channels directly.

In any event, the continuing "digitization of everything" appears to be giving us a *format independence* for all types of recordings. Everything becomes just more data. You can see this major trend everywhere you look. Also, the time for some sort of surround sound is finally upon us. After reading up on M.A. Gerzon's Ambisonic system, I favor this as a more complete, adaptable, universal, effective, and efficient method than simply using three front channels and two "surrounds" and leaving the mike technique to the recording animals, er, industry. But the

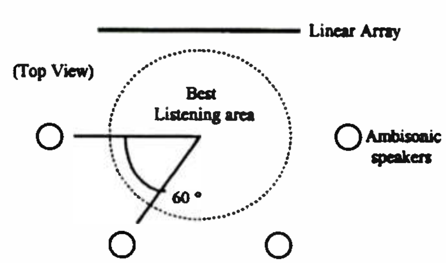

FIGURE 3: Possible configuration for the Witham Linear Array project.

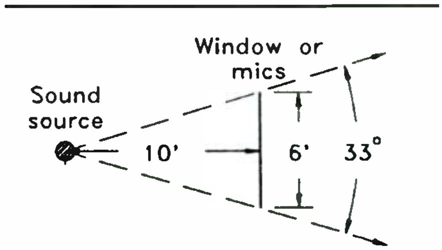

FIGURE 4: Test case 1 using a 5kHz sine sound source.

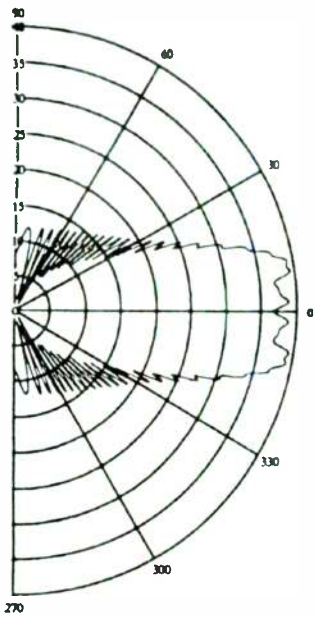

FIGURE 5: Relative sound level of test case 1.

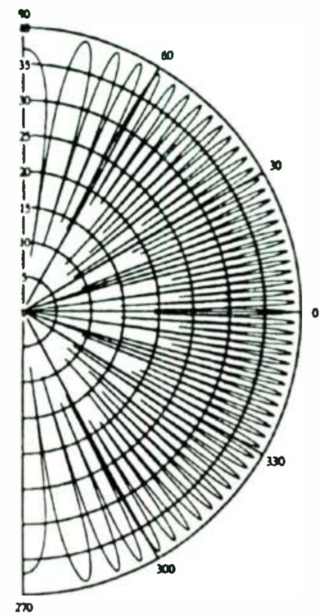

FIGURE 6: Omnidirectional speaker pattern.

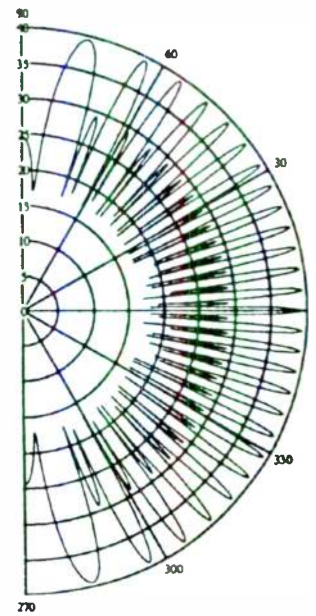

FIGURE 7: Measurement with center channel speaker added.

Dolby system has really "happened," and I'm all for it. You've convinced me to stop comparing a Linear Array to two-channel stereo, and start comparing it to three front channels from now on.

The papers you mentioned increase my dissatisfaction with stereo. Here is where I get off the bandwagon: The early references state that an array of enough channels would actually reproduce a sound field, and that two- *or* three-channel stereo cannot do this. They state, correctly, that stereo only creates an *illusion* of a 2D sound-stage in your listening room (and, incidentally, not an illusion of being in the recording location). The

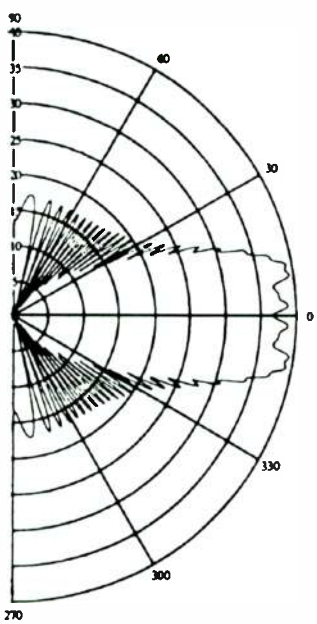

FIGURE 8: 48-channel Linear Array.

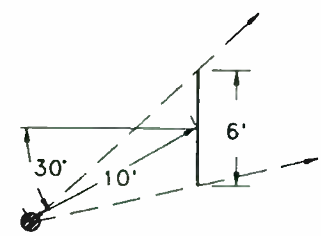

FIGURE 9: Test case 2 (30° off mike center).

ear does this magic for us, and made stereo practical in the '50s by fusing a few separate discrete sound sources, as you mentioned, into (more or less) one whole.

Snow states that there are fundamental differences between this *illusion* and the real thing, and that the ear uses two distinctly different mechanisms to hear binaural (natural) sound, and stereo (a fusing of separate sound sources that does not occur in nature). These papers state, simply, flat out, that while an array of lots of little mikes and corresponding speakers would be ideal, it is not practical. End of discussion.

This is true today as it was in 1953, when people struggled to broadcast just two channels—in one test mentioned, they transmitted one channel on AM radio, and the other on FM! They did not say that they even tried their "myriad loudspeakers of the screen." Someone, no doubt, did, and I am still searching for that paper, buried somewhere, with the results. Or maybe no one has tried it with *this* many channels spaced so closely. Some of the problems with two- or three-channel stereo are: The image shifts left-right with the listener rather than being fixed

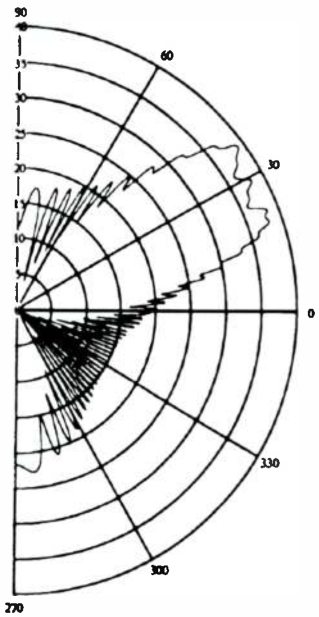

FIGURE 10: Test case 2 measurement results.

in space, and if you plot stage position versus apparent position at playback, you get a very warped plot—an image like looking through a melted lump of glass.

I have no illusions about the practicality of recording 16–100 channels now. I know it can be done, and for (high-end) consumer prices, but it would be a long uphill battle to uproot and deflect the recording industry. I just want an explanation for why our sound systems do not sound real, and I think the difference between this *illusion* of stereo and the *fact* of a truly reproduced sound field is a big part of the answer. And I intend to find out if this is so, by experiment.

I am often so enchanted by listening to real sounds that when I play my stereo system, I am truly disappointed...by something, which is not: distortion, frequency response, maximum sound level, wow, noise level, or mike technique, or anything we conventionally measure. We are rarely (and only momentarily) truly fooled into believing that *that sound* coming from the speakers is real.

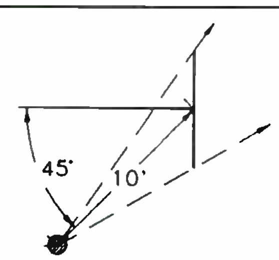

FIGURE 11: Test case 3 (45° off-center).

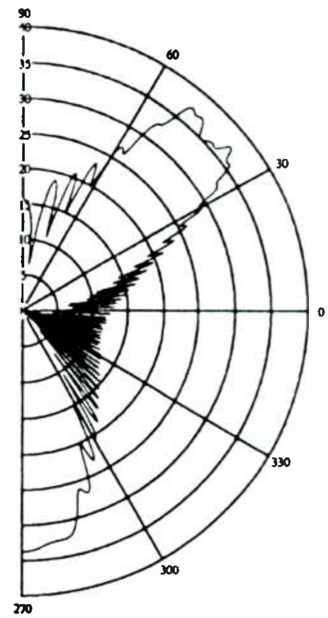

FIGURE 12: Test case 3 measurement results.

We suspend our disbelief to enjoy our recordings, and can almost imagine the musicians being there. We've been suspending our disbelief for so long that we ignore what we hear. Blindfolded, your half-deaf grandfather could correctly determine "real" or "Memorex" 90% of the time. If my feeling is correct, then we will know one direction of the future, even if we have to wait to get it. Sound interesting?

I'm going to experiment with a simple test, but with at least 48 channels. Sorry I gave the impression of being a Golden Ear with my preference for an ESL, but there are reasons why I think an ESL would physically work better here. Yes, I intend to duplicate Snow's stage-position versus image position experiment—or something like it. But there are probably better experiments to copy, done with more modern mike techniques, oriented towards home reproduction (Snow's goal was good movie theater reproduction). I'm interested in finding a good modern reference for this.

One last tidbit: 64 channels * 16 bits/sample * 48,000 samples/sec = 49.2 Mbits/sec. With 2:1 compression (very easy to do with no losses in this case), that's under 25 Mbits/sec, the rate of a consumer digital video recorder.

ESL ALTERNATIVE

Stanley Marquiss of Plymouth, CA, suggests that an electrostatic loudspeaker (ESL) is not the way to go, but that a large planar system is philosophically the best method for reproducing an orchestra, which is a rather large area sound source itself. He referred me to a system called "The Wall," measuring about 8′ × 16′ and displayed at the 1987 summer CES. He also believes that digital sampling (needed to make a 64-channel system practical) is a step backwards, and that "CD quality is a dreadful oxymoron."

In my opinion, regardless of digital sampling shortcomings, the audible benefits of having so many recorded channels outweigh many small problems we're concerned with today.

CHANNELS & PLACEMENT

I agree with Mr. Sommerwerck (of Bellevue, WA) on many of his points, with some exceptions. Omnidirectional mikes, as I understand, do pick up the "instantaneous pressure of the composite waveform at their location." Air is a linear medium for sound propagation (at the sound levels we deal with), and multiple sources are summed at any point to produce a total instantaneous pressure. This shouldn't be much different than the basis for phased array SONAR, which has been in use for decades.

Sound gives air an instantaneous velocity as well, but an omni mike does not sense that, as far as I know. Obviously, drawing wave fronts as lines is a gross oversimplification of real sound sources, which are not point sources, and are immersed in a reflecting/diffusing/absorbing environment. But, with the exception of the limitations mentioned previously, that should make no difference to the Linear Array concept.

While there are never *too many* channels in a phased array, there is a definite spacing limit where you don't *need* to fit the channels any closer for a given recording situation to eliminate audible aliasing. I roughly defined this limit in my article. If you have more channels to work with, they would be best used to build a wider system—perhaps a 2-D array—or to surround the listener. I judge 64 channels to be a practical limit for the near future.

I am fully in favor of adding three or more ambisonic channels in the Linear Array recordings. What's five or ten percent more bandwidth between friends? I'll lend you a few channels. I need to resolve such thorny issues as the best location for a sound field mike relative to the line mike, and the best combined speaker configuration. The combination is a way to concentrate bandwidth (recorded information) where it has the most benefit to the whole effect. (Don't be shy about the future, we will have massive bit rates at our disposal soon enough.) One possible configuration is shown in *Fig. 3*.

The front two ambisonic signals could possibly be added to the linear array signals with a DSP system simulating two virtual front ambisonic speakers using individual time delays to each array channel. Or, the front two channels could be ignored, given the linear array. In any event, I'm just dreaming here; I'm concentrating on prototyping the array.

Regarding shoehorning the required number of channels into a CD, I assume William is talking about three or so ambisonic channels. Compressing 64 channels into a CD is not practical. The consumer digital video recorder will allow a simple, lossless compression technique to be used.

William is correct that I should have used the term "temporal sampling" rather than "digital sampling" for my spatial sampling analogy. The Nyquist criterion does not say you need three samples per cycle, but merely *more than two*. How many depends on how good your output filter is and how quiet you want the aliasing. CDs do fine with 2.25 samples/cycle, for example. In the case of a Linear Array, a first-order output filter is effectively produced by the number of channels; that is, the maximum power generated by one channel is reduced by the total number of channels (correct me if I'm wrong here, Mr. Koonce!).

As far as beaming or poor dispersion from a large planar diaphragm, I should have said: "The problem will disappear if a Linear Array *System* is divided into enough channels," meaning, microphones, amps, *and* speaker channels.

His point about instrument timbre and the imaging of leading edge attacks is a good one. The attack usually has the highest content of high frequencies, and will be very demanding. I guess that the attack of most instruments has a broader spectrum and not usually a well-defined set of harmonics, which should help. Also, stereo has a much bigger problem with this—try feeding your two speakers a 6kHz sine wave and carrying a sound level meter around the room.

William also mentioned that several TEAC DA-88 8-channel digital recorders can be synched/ganged for this use. Could be a good choice for initial demos and such. I appreciate William's comments and his kind offer to help with the electronics design and assembly.

MARKETING APPROACH

Tom Holzel, an ex-Advent marketing manager from Concord, MA, offered several practical suggestions for marketing my novel design. At this time, however, I am not currently trying to produce a commercial product, which, with recording equipment, a library of program material, and home playback equipment, would take at least five to 15 years of hard work and investment.

Commercializing a sound system using array speakers with synthesized delays would probably be possible in the near-term. But it would probably remain an esoteric ($5,000+) high-end speaker. Its main advantage would be precise, adjustable dispersion and apparent "speaker" location. Just turn a knob... I imagine this would be appropriate for a video sound system, hidden behind the screen. Otherwise, why not just buy a nice pair of Martin Logans (curved ESLs) or Quads? Or some Apogees?

I am going to start with conventional speakers for an experiment, but with a couple of misgivings:

1.) Vertical dispersion will not be controlled. You can hear the difference between a cylindrical and a spherical wave front through room acoustic interactions (bouncing off the floor and ceiling) and through the apparent source location as you move. Also, the volume level changes more with distance from a point source than a vertical line source. I believe it all adds up to being able to hear the location of point source speakers more easily, which we don't want. Many commercially available speakers—including

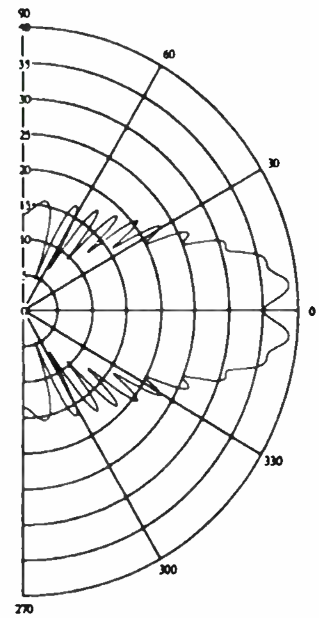

FIGURE 13: Test case 1 results at 2kHz.

tall ribbon speakers, vertical arrays of cone drivers, all ESLs, and other planar drivers—restrict the vertical dispersion angle. In my opinion, they sound better for it.

2.) The array will be sensitive to phase and amplitude differences between channels, and so the variation between different copies of the same driver model could scramble the imaging.

The point is to determine whether the Linear Array is a better way of recording sound than stereo or quad or Q-sound or Ambisonic, or what have you, and to give the system a fair chance by avoiding half-measures. Then we can discover what a minimum linear array system really is, and just what we're missing with stereo. We could promote a format standard, and think about commercializing it. Until then, it's a labor of love.

PROGRESS REPORT

G.R. Koonce convinced me to build a cheap experiment using dynamic drivers. I could not bring myself to adopt 3″ drivers interlaced vertically, which was his suggestion.

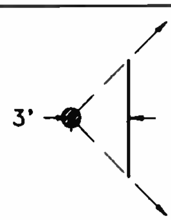

FIGURE 14: Test case 4.

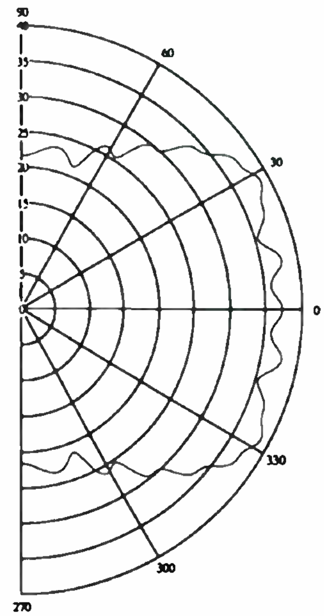

FIGURE 15: Test case 4 results.

The spacing is too wide, and the vertical spacing is unacceptable. I am using small, four-dollar 1.6″ Mylar® cone headphone-style models, instead, and two-dollar omni mike capsules. These little drivers have a 600Ω impedance, so I can drive them directly from the mike preamp. I designed the mike preamp, and built one complete signal chain: mike, preamp, and speaker element. Testing indicates the mike and preamp perform wonderfully for the job.

I built and tested an enclosure for the driver using PVC pipe stuffed with wool and plugged. Simply putting the driver in a hole in a large baffle (as a dipole) worked best. I plan to mount 48 of these in a six-foot line on an 8′ by 4′ sheet of Lexan®. The little driver has an annoying emphasis on the 5–20kHz range (+7dB), probably put there intentionally as someone's idea of "hi-fi."

Otherwise, it is flat ±1.5dB from 150Hz–4kHz (measured with one-third octave warble tones). If anyone knows a better 600Ω, 1.5″ or so driver for about the same price, let me know. It won't be very loud, nor capable of bass below 100Hz, but it will suffice for a throwaway proof-of-concept.

I intend to perform some blind testing—perhaps a comparison with stereo—of the imaging using several disinterested test victims, er, subjects. If anyone is interested in volunteering as a tester for a few hours, please let me know.

I used Easy-PC to lay out a 32-channel preamp PCB. I highly recommend this program for low-cost PCB layout; it is fast, simple, and effective. This board is about

3″ × 9″, and two of these plus a power supply make up all the electronics needed. For now, gain control occurs by swapping out resistor headers on the preamp PCBs.

So, I'll begin building the proof-of-concept system this month. Total cost should be about $1300, much of which is for the little PCB run and a 50′ cable. So far, no one has volunteered to help pay for any of this or to build an ESL. Well, what's money good for but speaker experiments, eh?

I've written an array simulation/polar dispersion plotter in the program Mathcad. The simulation indicates that the Linear Array should work about as predicted, and also demonstrates what sort of image aliasing we'll get for tough cases. It is interesting to set the number of channels to two or three (two plus center channel) to see just how badly stereo works.

Figure 4 represents a test case where a 5kHz sine sound source is placed 10′ from the microphones, or from a 6′ wide window in a wall. *Figure 5* is the polar plot produced by the simulation, for the case of the ideal "window in space"—actually, a 6′ long slot. It plots listening angle (from the center) versus relative amplitude in decibels and represents the relative sound level you would measure if you walked around the "window" in an arc, at a great distance, in an anechoic environment. The absolute amplitude plotted is meaningless.

It is important to notice that as the sound source moves out of "view" through the window, the level drops quickly. Within view, the amplitude is fairly constant. The strange wobulations and lobes in the plot are due to diffraction, or at least that's my guess. This simulation is not a finished or verified work, so swallow it with a grain of salt.

Figure 6 shows the results from two perfect omnidirectional speakers spaced 6′ apart. The mike technique is not important, as long as the two channels' signals are in phase. The resulting fan of lobes is produced by the alternate addition and cancellation of the two discrete sound sources as you move along the measuring arc. Room reflections, of course, will affect the measurements.

In *Figure 7* I added a center channel speaker, which is not much better in terms of reproducing a true sound wavefront. I used three omni mikes spaced just as the speakers. Coincident mikes would produce a similar result. The upshot is that any stereo imaging we hear exists only in our minds, and is just a convincing illusion, not a reproduced wavefront. In fairness, I should add that three channels will actually produce some wavefront imaging in a narrow frequency band around a few hundred Hz.

Figure 8 is the result for a 48-channel Linear Array, 6′ across, with 1.5″ channel spacing. The little drivers assumed here are ideal omnidirectional units, as in the other examples. It works. Through the listening area, one fused wavefront exists, "originating" at a point 10′ behind the speaker. At extreme angles, the side lobes are up about 5dB from the perfect case of the window. The image aliasing limit of this array is about 5kHz.

Figure 9 shows another test case, with the same sound source off 30° from the center axis of the mike. *Figure 10* is the result from the same 48-channel array. An alias is starting to form along the 270° angle, along the line of the speakers. It is about 15dB below the main image, but since it is being launched toward the side wall of the listening room, it is probably not a problem.

Figure 11 is a third case, with the sound source at 45° to the mike; *Fig. 12* shows the results. The alias at 270° is almost as strong as the real image, and probably has an audible effect. Note that this doesn't happen for lower frequencies, and that any aliasing is different for each frequency. The real image will dominate for properly recorded music.

Continued on page 64

Wayland's Wood World

BISCUIT JOINER CORNER JOINTS

By Bob Wayland

Of all of the joints you will make when constructing a speaker enclosure, the corner ones have the greatest exposure and the highest stress. A corner joint is normally visible to anyone looking at your enclosure. Besides looking good, it must be able to retain its integrity while withstanding the stresses produced by the vibrating drivers. It must remain airtight and structurally sound.

Last month, I showed you how to make a simple butt joint with a biscuit joiner. Although sturdy, the butt joint has the aesthetic appeal of a forklift pallette. The exposed edges and crude stuck-together appearance are normally acceptable only if you plan to cover them. This installment provides instructions for making aesthetically pleasing corner joints using a biscuit joiner. Although we mainly consider right-angle joints, I also discuss how to modify the techniques to produce joints at other angles. Next time I will cover how to make splined corner joints on a table saw.

The secret to making any joint satisfactorily is an accurately set saw blade and miter guide. Joints at other angles require an accurate standard against which to set your saw. The Veritas Poly-Gauge (about $25 from most woodworking mail-order houses) is one of the easiest to use. This gauge is a precision-made reference for layout of 4-, 5-, 6-, 8-, and 12-sided boxes. If you need other shapes, the Mite-R-Gage (about $20 from most woodworking mail-order houses) is an extraordinarily handy tool. This continuously adjustable gauge has two long arms attached to a 3½″ scale that can be set with care to about ±1°. While not as accurate as the Poly-Gauge, it is more accurate than most protractors.

For right angles, you simply set up your saw very accurately. First, you need an accurate try square (discussed in my column of *SB* 6/93, pp. 50–54). Make a test cut with your saw blade and miter gauge both set at 90°. Check the horizontal and vertical cuts for squareness and make any necessary corrections. With your saw blade at 45°, cut two test pieces and clamp them together to see if they form a 90° corner with your try square (*Photo 1*). Then readjust the angle on your saw blade and test again, continuing until you have a good right-angle corner.

Sometimes you need to cut a large piece. If you use your miter gauge as a guide, it may slip. I have glued sandpaper to the face of the miter gauge to help keep the board in place. This helps, but is not completely satisfactory. Tage Frid in his book on joinery (*Tage Frid Teaches Woodworking Book #1: Joinery,* Taunton Press, 1979) describes a simple technique that lets you overcome this problem. First, align your board, using the miter gauge, for the cut you wish to make. Then clamp a guide board so that it slides along the edge of the table, as shown in *Photo 2.* This enables you to cut large panels accurately.

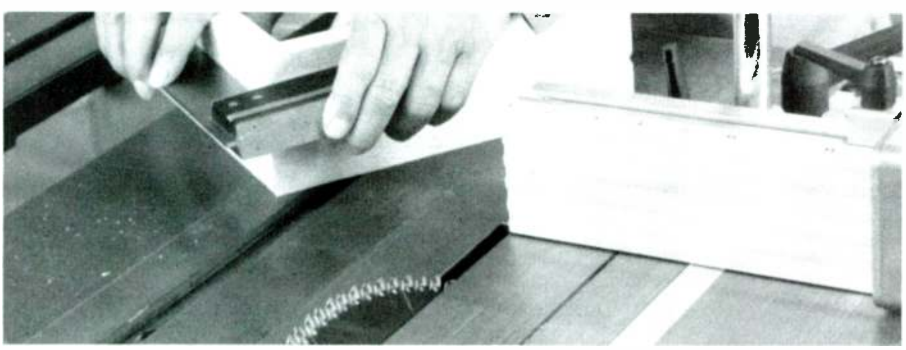

PHOTO 1: Testing that the two test cuts form a 90° corner with a try square.

BISCUIT JOINER CORNER JIG

This tool enables you to make aesthetic joints easily. The following is a step-by-step procedure of how to make and use this type of jig. First, cut a squared piece of 2″ stock (6″or wider and 12–20″ long) at the desired angle. For a right-angle joint this is 45° as shown in *Photo 3.* (If you are jointing two pieces to form an angle other than 90°, the angle of cutting should be half of that angle or its complement if the angle between the two pieces is greater than 90°.) Place one piece on top of the other to form a stack (*Photo 9* shows how the stack should be aligned), apply glue, clamp, and set aside to dry as shown in *Photo 4.*

PHOTO 2: Technique for sawing large panels.

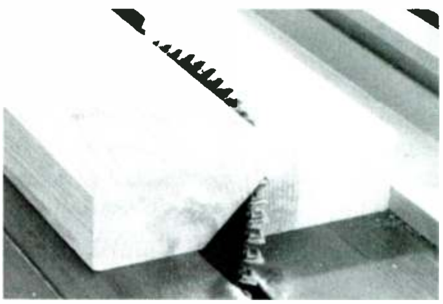

PHOTO 3: Cutting the support structure for the biscuit joiner jig.

While the stack is drying, you must make a few measurements. First, determine the distance from the base of the biscuit joiner to the top of the blade as shown in *Photo 5.* Record this distance as "a." For the Freud JS100 biscuit joiner this is 15/32″. We wish to make the joint as strong as possible, so choose the position of the cut slot very close to the inner edge of the joint.

But you also wish the biscuit hidden, so cut the slot, leaving a 1/16″ lip on the inside edge of the joint. Now add 1/16″ to the distance "a" and mark this distance down from the inside edge of a test cut you have made on a scrap piece of your enclosure material (*Photo 6*). Now measure and record the distance from the outside edge of the joint to this mark as shown in *Photo 7.* You can calculate these distances if you wish (see sidebar).

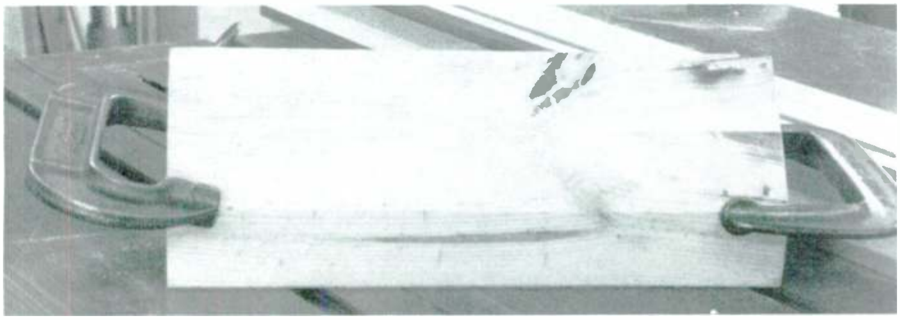

PHOTO 4: Clamping the angled stack for the biscuit joiner jig.

PHOTO 5: Measuring the distance from the base of the biscuit joiner to the top of the blade.

With the saw blade still at 45°, remove the front edge of the form so that just the length *l* will be removed from the leading edge of the form you have made previously *(Photo 8* shows cutting the form). If you are making a corner joint other than the normal 90°, set the saw blade at the correct angle and depth calculated using the information in the sidebar. Mount the form on a large piece of scrap wood a few inches wider than the form and long enough to offer good support of the piece you are biscuit-cutting (*Photo 9*).

USING THE JIG

Now that you have made the jig, using it is similar to operating the biscuit joiner as described in my *SB* 4/94 column. After you have cut the two pieces to the required angles, you must decide where you want to place the biscuits. The rule a woodworker often hears is that the center line of the biscuits nearest the edges of the joint should be about 2" in from each edge. The second bit of advice is that the biscuits can be up to 10" apart.

The first rule applies to our situation; however, I suggest that you consider a different biscuit spacing. Since we are concerned with strength and with an airtight joint, I suggest that you make your biscuit slots so that the biscuits are butted end to end. This gives you a very good seal and extra strength. Once you have established your biscuit spacing, use a piece of scrap wood to make adjoining marks as shown in *Photo 10*. If the joint is other than 90°, make a marking jig that has the correct angle.

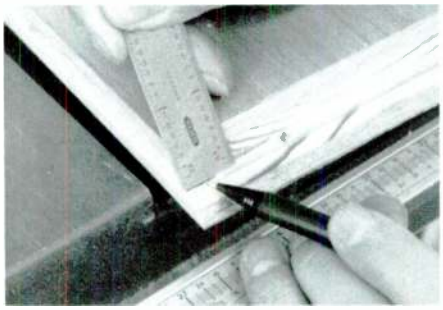

PHOTO 6: Marking the reference distance of 1/16" down from the inside edge of the joint. (The mark has been darkened to make it visible in the photo.)

PHOTO 7: Measuring the distance from the outside edge.

To use the jig, place it against a solid stop. With the piece firmly pushed into the overhang and with the biscuit joiner resting securely on the top of the jig, you are ready to make the biscuit slots (*Photo 11*). The alignment of the biscuit joiner is easy to determine by using the index on the joiner and the marks you have made on the pieces to be jointed (*Photo 12*). It is best to remove the fence.

If you are working with an especially wide piece, just slide the jig along the edge, making the biscuits as needed. After you have finished the slots cuts, insert the biscuits (*Photo 13*) and assemble the joint (*Photo 14*) to test for good fit and alignment. Remember that you should glue the joint as soon as possible after cutting.

With a little practice you should be able to quickly make strong and good-looking corner joints. Once you have made this jig, you'll be able to make them in just a few minutes. You can make other corner joints with the table saw. I will describe one of the best in the next issue.

CORRECTIONS

In my discussion of finishes ("Simply Great Finishes," *SB* 5/93 p. 72) the statement con-

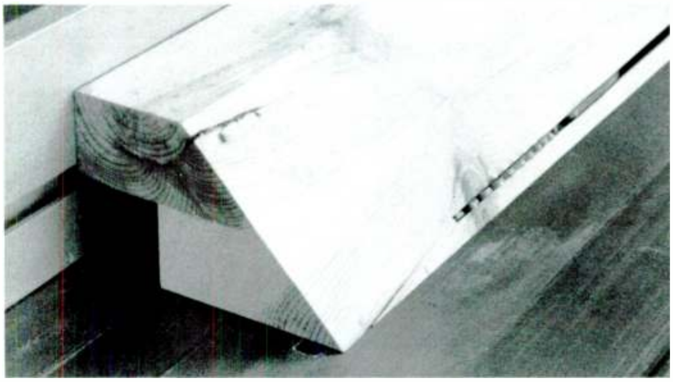

PHOTO 8: Sawing off the overhang *l* from the leading edge of the jig ramp. Note that the offset of the top piece relative to the bottom piece was caused by the way I cut the board. As long as the structure is strong you can have any overlap that you happen to cut!

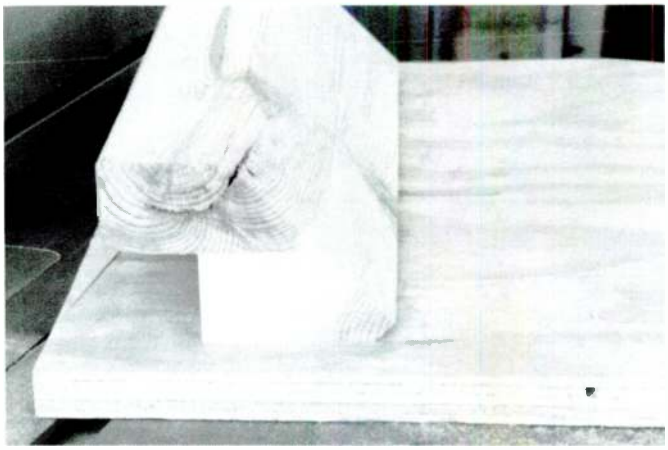

PHOTO 9: The completed jig.

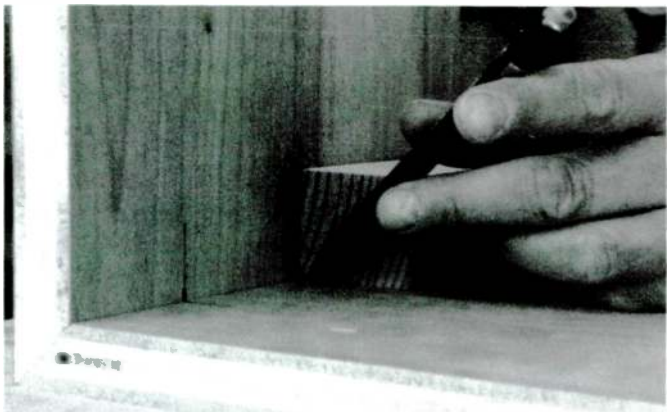

PHOTO 10: Marking the joint for the biscuit cuts.

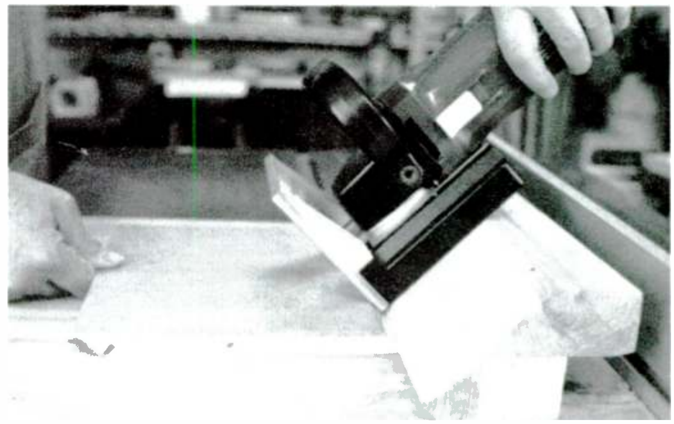

PHOTO 11: Setting up the jig for making biscuit slots with a biscuit joiner.

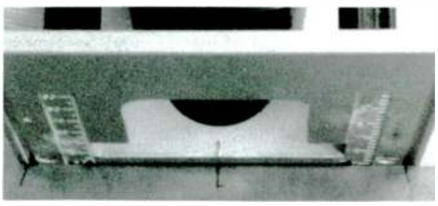

PHOTO 12: Aligning the biscuit joiner.

cerning prepared oil finishes, e.g., Watco Danish Oil, "I then apply a very thick coat of Watco and allow it to set overnight" is not what I meant to say. A woodworking friend, who is kind enough to help in the preparation of these articles, correctly pointed out that this is a good way to produce a mess. I meant to say that after you have applied and rubbed off the excess oil, don't apply another coat for at least 12 hours to allow the oil finish to set up. If you made the mistake of following this bad advice, please accept my apologies. You can correct this mistake by rubbing the hardened finish with fresh oil and immediately buffing it a bit.

A note on safety in the shop: Don't wear loose-fitting, long-sleeved garments. The sleeves can easily become caught in your machinery and cause great harm. In the pictures for the previous, current, and next articles, we dressed up by wearing a lab coat. Don't!

PHOTO 13: Inserting the biscuits.

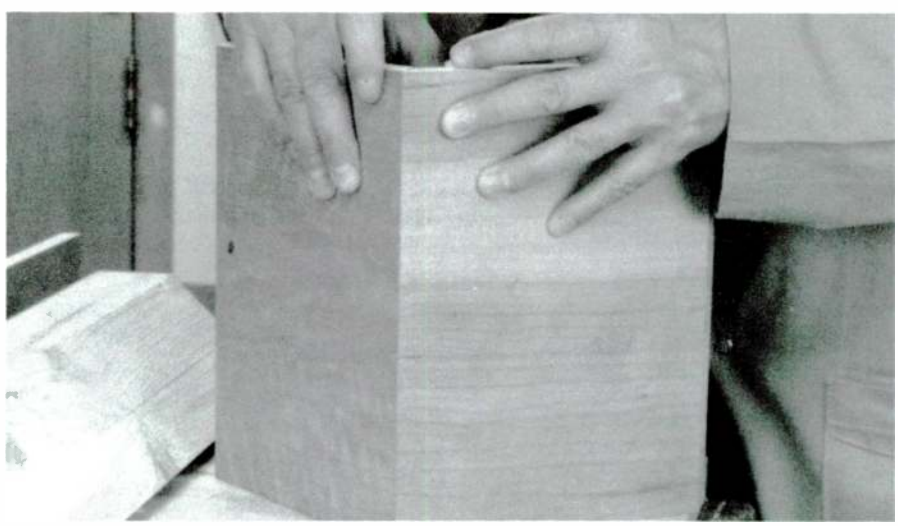

PHOTO 14: Dry assembly of the corner joint.

CALCULATION OF OFFSET FOR JIG

Consider the arrangement shown in *Fig. 1*.

We wish to determine the length, *l*, of the support to cut off. A little trigonometry gives us the formula

$$l = [w - (a + d) \mathrm{Tan}\phi] / \mathrm{Cos}\phi$$

For example, if $w = 3/4''$, $a = 15/32''$, $d = 1/16''$, and $\phi = 45°$, then $l \cong 17/32''$.

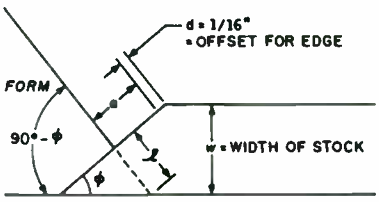

FIGURE 1: Sample arrangement for determining length of support, *l*.

Book Report

Loudspeaker Recipes, Book One

Reviewed by Richard Campbell
Contributing Editor

Loudspeaker Recipes, Book One by Vance Dickason. Available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458-0243, (603) 924-6371, 144 pp., $24.95.

This is the kind of book which will serve well many budding but fearful loudspeaker system designers. Perhaps fearful is not the right word. Intimidated? Daunted? The thought process might go like this: Spend a few hundred dollars and couple of weeks of part-time labor to construct a loudspeaker system, bring it upstairs, and wire it up for its grand introduction. Your wife says, "That sounds terrible"; the kids say, "We want the old ones back"; the dog howls. Not a nice day.

Loudspeaker Recipes, Book One assures a nice day. The book describes four very different loudspeaker systems which collectively fulfill the normal requirements in a household listening environment. They are, in four chapters, vented two-way, sealed two-way, two-way satellite with subwoofer, and dual woofer two-way. Each design features "vertical" appearing left and right enclosures on stands which are intended to be placed away from walls and corners.

Of course, the satellite system has a subwoofer cabinet. You should have a fairly large listening room before you consider building one of these systems. Perhaps the author will include systems for tiny apartments (bookshelf, and the like) in *...Book Two.*

Each chapter has a consistent style beginning with an overview, which leads directly to a discussion of driver descriptions and the reasons for selecting them. Then there is a discussion about the T/S alignments and the driver interaction with the cabinet. Many graphs and diagrams (how about 130 figures in Chapter One alone, which is 33 pages) illustrate measurements made with various computer-based instruments to support statements and conclusions in the text.

There is probably no way around it, but this book requires a lot of chasing around to find figures which relate to text several pages removed. Likewise, finding the text associated with a figure can be quite a chore. How about putting a page number reference next to the figure caption?

I like the author's practice of discussing several alternate crossover networks for each system design. In fact, this portion of each chapter contains the greatest instructional value. The mental agility required to sort out different crossover responses from a systems viewpoint is amply demonstrated, and a lot of it rubs off. Admiration rubs off, too. Mr. Dickason takes you by the hand and down the path of righteousness, and when he's done, no other solutions are worth their salt. Very impressive.

Besides the normal ones, two interesting sets of measurements are made on each system: off-axis responses and cabinet panel acceleration. Especially interesting are the off-axis responses in the vertical plane—over the top of the speaker. I consider these important because you hear their response shape as a first reflection from the ceiling of your listening room.

The uniformity of the frequency responses has much to do with the subjective judgment of the loudspeaker system in a particular listening room. It's OK if they diminish uniformly with frequency, but if they exhibit big dips and peaks, then they will undoubtedly alter what is heard subjectively in the context of complex musical sounds. If the non-uniformity of off-axis radiation is severe enough, it becomes far more important to the listener than any re-radiation from the cabinet walls, if the room has reflecting surfaces.

CHAPTER ONE

The design in this chapter shows a 10dB variation at 90° off-axis and a nasty 20dB hole at the vertical response at 30° up. These few data points, although alarming, are insufficient to draw any conclusions, but the trend bears further scrutiny. A higher data density is needed. If that hole lasts from 25° to 60°, for example, it means trouble. If it lasts from 25° to 29°, it may not be worth worrying about.

The accelerometer data from the cabinet walls is particularly interesting because one of my students just completed a similar project in investigating ultra-lightweight cabinet materials. My student took MLSSA acceleration data, wall displacement data using a Kaman precision displacement meter, and sound pressure data at 1cm above the surface with a calibrated B&K 4134 microphone. The idea was to determine a method to calculate the resonance frequencies of an edge-clamped plate (the back wall of a loudspeaker cabinet) to be sure that these frequencies did not coincide with one of the box modal frequencies.

The author shows several graphs of cumulative spectral decay (CSD) done with MLSSA and with an uncalibrated accelerometer which means that the data is relative only. The ringing of the plate modes is clearly visible, and you could easily estimate the frequencies involved. The author then states that displacement can be computed by dividing the acceleration curves twice by the frequency. This is not strictly true, since they must be divided twice by *omega*, which is $4*\pi^2 f^2$. It's OK here since the curves are only relative, but it would not be correct if he had used a calibrated accelerometer like my student did and computed actual displacement.

MLSSA has a feature which allows double integration of the time-domain acceleration shot, and you can then take an FFT of that and get displacement as a function of frequency. Except for the continuous upward slope of the double integral caused by using less than the whole sequence (which can be easily zeroed out), the time-domain displacement plots from MLSSA and from the Kaman instrument are nearly identical.

I predict that in the next few years cabinet wall displacement analysis will play an increasingly important role in speaker system evolutionary development. The fight to reduce it to minimal levels causes an impressive increase in cabinet weight. I prefer to control it—let it happen if the results do not degrade the system performance significantly, but misery to anyone who lets the plate modes land on the box modes.

Since I am in the mood to complain, I will do so loudly about another section. In a paragraph near the beginning of the chapter, the author states "...fast Fourier transform (FFT) analyzers tend to be only semiaccurate, depending on the resolution and bandwidth of the measurement. LEAP's sophisticated and modified Hilbert transform routine generally provides more accurate data than the measured phase..." What is that all about? What does semiaccurate mean? Half-accurate? The Hilbert transform, modified or not, forces the assumption of minimum phase in the device being measured. Now, you could assume the T/S model is minimum phase and use the Hilbert transform to compute phase from the magnitude response, but is that better than actually measuring it? Nonsense!

FFT analyzers like MLSSA (the user thinks it's an FFT analyzer because FFTs are used to present the data—in fact most of its processing algorithms operate in the time domain—measure actual phase and further have the capability to compute the difference between the Hilbert-transformed phase (phony phase) and the actual phase (real phase). Which one would you like to see—pretty pictures created by an equation which makes an assumption about a system which may not be true, or the ugly, real thing?

The author implies that truncated impulse responses are less accurate than swept-sine measurements. Among the professionals in the audio and acoustic community who do lots of measurements, this implication is well known, and is referred to, with a wink and a smile, as "TEFie talk."

I can't blame them in a way. Here you are manufacturing an instrument which uses a swept-sine technique originally proposed by one of the greatest audio engineers I ever knew—Dick Heyser. Let's say you want to make an impedance measurement with 1Hz resolution in 2kHz bandwidth. The sine sweep rate must not be greater than SQRT (resolution), which is 1Hz per second. For a linear sweep, my calculator shows 2,000 seconds divided by 60 equals 33.3 minutes.

Now comes MLSSA onto the market with a radical new technique based upon the work of another god in the acoustics field, Manfred Schroeder, and makes the same measurement, with the same accuracy in two seconds. In one additional second, it does a complex least-square curve fit to the T/S model (using real phase, not phony phase). This new instrument costs six times more, but is 1,000 times more productive. With a wink and a smile, I have identified the genesis of "TEFie-talk."

FFT analyzers have received an unfair share of condemnation in this business because users either do not understand or are unwilling to learn recovered amplitude scaling and band integration methods. I will agree that an uneducated user can easily generate useless data. The brave souls who perform hearing aid measurements have approved FFT analyzers and written them right into an ANSI standard because the coherence function as a measure of distortion is a really good way to judge hearing aid quality, and the coherence function is easily computed using either a statistical noise—or an MLS-based FFT analyzer (admittedly—scaling not required). Sorry for this digression—but I dislike misinformation and misleading statements.

The author comments about the consistency between drivers of the same model from the same manufacturer, and he shows minor differences among four sample woofers from Vifa, an established manufacturer with an excellent reputation. My OEM experience (the end-user buys directly from the manufacturer) is similar. However, many driver resellers offer generic models in their catalogs under their own name, and I have found much greater differences between samples from such sources. I suspect it's related to lack of QC as the drivers pass through their hands on the way to you—or they have bought a bunch of out-of-spec units at a fire sale. If you decide to try a different driver that has the right advertised parameters specified by the author, be careful—don't cut wood until the drivers are exercised and carefully characterized (including an inspection to see if the voice coil is centered in the gap!).

CHAPTER TWO

The sealed two-way design in this chapter looks like a real winner. It has a Q_{TC} of 0.655, nearly perfect for optimum transient response. It is a good notion that much time is spent on time delay analysis between the woofer and the tweeter so as not to compromise this optimum LF performance exhibited by the 5ms group delay at 50Hz. If I had to take one of the four systems to a desert island for the rest of my life, I think this would be the one.

There is one equivocal statement in the discussion of Q_{TC} and f_3: "Apparently, changes in the total box volume have only relatively minor influence on this woofer's f_3 and Q_{TC} for box sizes larger than 0.75 ft^3." We never hear the reason for this "apparentness," which deserves some explanation. Consider the equation $Q_T = \omega * M_a/R_a$. Total Q of a closed-box system is affected by the box acoustic mass, M_{ab}, as a term in the numerator being two orders of magnitude less than total acoustic mass, M_a, and by the box resistance, R_{ab}, as a term in the denominator being three orders of magnitude less than the b^2l^2 damping (with no stuffing).

Therefore, I would not expect any significant change in Q_{TC} as a result of changing box volume (over reasonable values). The system resonance frequency, ω_0, is roughly inversely proportional to the square root of the box volume, which means Q_{TC} is changing. Box volume goes down, ω goes up, Q_{TC} must go up if M_{ab} and R_{ab} don't change. The universal resonance curve starts to grow a bump at ω_0. For Qs greater than unity, indeed f_3 does not change very much. I don't know what's going on with the author's design, but a little more "why" would be nice.

I agree with the concept of X_{MAX} + 15% as being a criterion for 3% distortion-based excursion limits, but I have not seen the justification which may be from another reference. The last of the voice coil wires pass through the gap fringing field on their final plunge to X_{MAX}, so we have an interesting and complicated nonlinear function which may be driver dependent. Also, most music transients are monopolar; that is, the initial huge peak is never followed by one as large whether positive or negative, at least in an orchestra hall, and that would be heard as even-order distortion (if we could hear the distortion at all) making it more aurally palatable.

CHAPTER THREE

The system featured here is a superb example of the author's extraordinary ability in this field, and why we all keep buying his books. Here, he uses the low-frequency rolloff of a closed-box design, which is a B2 response, and combines it at the same corner frequency and Q with a B2 electronic crossover to make up the fourth-order L-W high-pass filter. This all happens around 115Hz. A detailed discussion of the subwoofer design and an extensive discussion of the tweeter section and of the additional crossover requirements follow. The use of asymmetric crossover slopes is very well explained. This chapter is a big read—over and over before it's 100% absorbed.

CHAPTER FOUR

The final chapter describes a tall, skinny woofer-tweeter-woofer combination à la Joe D'Appolito. The cabinet has two stagger-tuned cavities each with a Variovent and is very substantially constructed (75 lbs.!) as evidenced by the lack of activity on the MLSSA acceleration shots. Mr. Dickason suggests that a subwoofer be added to the ensemble for applications requiring high SPLs.

NIT-PICKS

Earlier I made some remarks which will probably end up in the dustbin of academic nit-picking. In addition, I must include a couple of technical shortcomings which are not serious but annoying, especially for the first-time builder:

1) There are no "construction" drawings of the type woodshop people are used to seeing. I recommend that the author prepare a set for each design and offer them as an extra item for

Continued on page 64

Product Review

THE WOOFER TESTER

By Vance Dickason
Contributing Editor

The Woofer Tester (model DSP-126) is an outboard hardware box that contains a digitally controlled oscillator and voltmeter interfaced to the computer through an RS-232 port. The device is professionally packaged in a small 3" x 5" x 6" metal enclosure with silkscreened panels. The rear panel has the usual 9-pin com port receptacle and a 9V power pin connector, with the front panel and two probe jacks labeled for impedance, two RCA jacks (labeled "Aux In" and "D/A Out"), and a power LED indicator. A line transformer with a 9V output is supplied for powering the Woofer Tester, along with probe cabling, an RS-232 cable, and a calibration resistor. Software, The Woofer Tester v. 1.0, is supplied in both 5¼" and 3½" disks.

Setup of the Woofer Tester is quick and straightforward. Simply plug the computer cable into com1 or com2 of the computer, insert the probe cables into the impedance jacks, and plug in the transformer line to the 9V pin. The software is copied directly to an appropriately named subdirectory (no installation routine is used), with no setup required.

HOW IT MEASURES UP

The system can be initiated to look for the device operating out of either com1 or com2 by typing W1 for com1 or W2 for com2. The software pops up with a simple seven-item menu: Plot Impedance, F_S and Q_{TS}; Measure V_{AS}; Inspect Results; Measure Vented Box; Calibrate System; Arbitrary Impedance Plot; and Quit.

To test the accuracy of the DSP-126 parameter tester, I measured three 6½" drivers (one Bravox and two SEAS). My benchmark for parameter testing is the LEAP 4.51 Speaker Parameter Measurement routine, which has a high correlation to static measured BL. I used LMS to perform impedance plots, and, after subtracting the measurement-cable resistance, imported the data into LEAP to calculate parameters. I measured DCR with a Fluke 8060 DVM, using the DVM's "relative" switch to remove the test cable resistance (the Woofer Tester calculates DCR at a near-DC-voltage level automatically). The LEAP box simulations I produced using the LEAP/LMS method, which is slow but accurate, are always within a few hertz and decibels from the measured prototype.

All three drivers were exercised with a low-frequency sine wave for 24 hours prior to testing. I performed the test sequentially, first testing the product on the Woofer Tester, then performing box Q and V_{AS} tests, then running both free-air and delta-compliance impedance curves using LMS. The current software version of the Woofer Tester supports only a delta mass type of V_{AS} test.

Time required for the Woofer Tester to perform its tests was very close to that described by the product's author, Brian Smith. The Q test procedure took 1¾–2 minutes, and the V_{AS} test required about 30 seconds. The Woofer Tester's vented box measurement routine, which produces F_{SB}, F_L, F_M, F_H, α, and ha, took about 1¾ minutes to complete.

Test results on the Bravox 6½" (a new Brazilian woofer using an injected cone and dual shorting-ring T-pole piece motor) and the SEAS 6½" T17RE-P-EA and T17RE (both of these have a new crystal-clear poly cone material), using both the Woofer Tester and the LEAP/LMS method, are shown in *Table 1*.

FIVE CENTS' WORTH

F_S, Q, and R_E results were reasonably close between the two methods. *Figure 1* shows the result screen for the Bravox 6½"; an example of the graph screen that appears on the monitor during testing is depicted in *Fig. 2*. (The Woofer Tester does not have a print routine, and the pictures were created using the HiJaak® screen-capture software.)

My comparison between LEAP and the Woofer Tester is almost as close as that between the LEAP/LMS system, the Audio

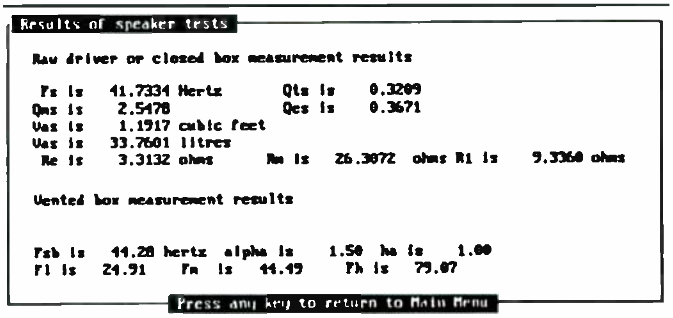

FIGURE 1: Text screen for the Woofer Tester.

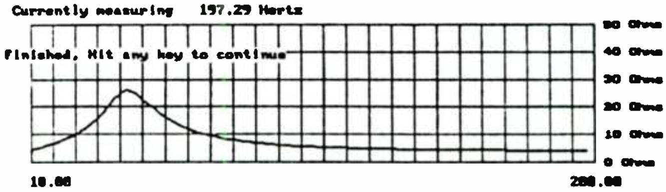

FIGURE 2: Test mode screen for the Woofer Tester.

Precision System 1, and the MLSSA-automated parameter measurement routines (September 1991 *Voice Coil* newsletter). Woofer Tester V_{AS} calculations for the two SEAS samples appear to warrant some attention, however (*Table 1*). The results for the Bravox driver were not bad even for the V_{AS}. *Figure 3* shows a comparison of a box simulation using both the LEAP data and the Woofer Tester data.

Although two different V_{AS} calculation methods—delta mass with the Woofer Tester and delta compliance with LEAP—are being compared, the results should be closer than were obtained for all three drivers. As part of the delta mass procedure, the Woofer Tester V_{AS} calculation routine calls for the placement of nickel coins on the cone surface, inputting the number of nickels it takes for the program to obtain the mass weight.

Not only is this system not particularly accurate (compared to using an Ohaus Triple Beam scale, for example), but there is a potential problem securing the nickels to the cone surface. I discussed this problem with C&S Audio Labs, and the company intends to change the routine so that either delta mass or delta compliance can be used and the delta mass routine will ask for the weight in grams, not nickels.

Considering the cost, fairly quick testing, and reasonable accuracy of the F_S, Q, and R_E data, the Woofer Tester looks to be a useful device for sorting and testing woofer parameters for medium- and small-production quantities. Its accuracy should be sufficient for this purpose, and it is certainly less expensive than using a dedicated MLSSA or AP System 1. The Woofer Tester costs $399 and is available directly from the manufacturer, C&S Audio Labs, PO Box 1012, Savage, MD 20763-1012, (301) 498-8737.

TABLE 1

TEST RESULTS FOR WOOFER TESTER AND LEAP/MLS

Parameter	Bravox 6.5 WT	LEAP	SEAS T17 RE-P WT	LEAP	SEAS T17 RE WT	LEAP
F_S	41.7Hz	41.2Hz	40.8Hz	40.1Hz	42.0Hz	40.4Hz
Q_{MS}	2.55	2.37	1.50	1.42	1.53	1.36
Q_{ES}	0.37	0.34	0.72	0.69	0.540	.49
Q_{TS}	0.32	0.30	0.49	0.46	0.40	0.36
V_{AS}	33.8 ltr	29.0 ltr	35.3 ltr	21.0 ltr	43.0 ltr	24.0 ltr
R_E	3.31Ω	3.17Ω	5.65Ω	5.51Ω	6.60Ω	6.46Ω

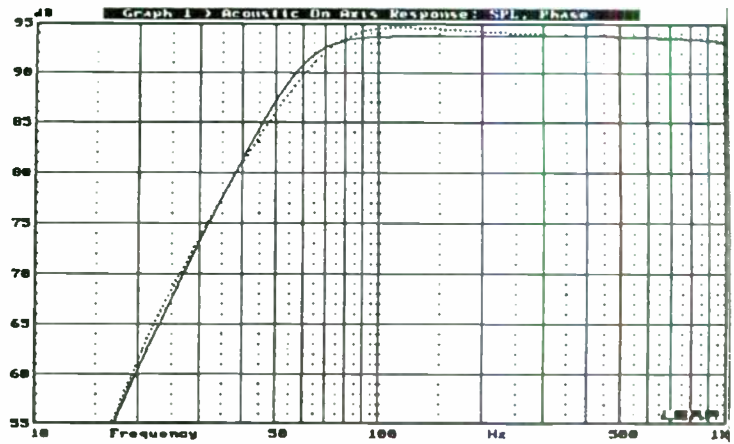

FIGURE 3: Box simulation comparison using LEAP data (solid) and Woofer Tester data (dotted).

SB *Mailbox*

IN THE DARK

I read with interest the article entitled "A Full-Range Open-Baffle System" by Warren Hunt and Joseph Janni (*SB* 1/94, p. 30). I wish to build the speakers, but was left only partially instructed after reading the article because it omitted dimensions for the baffle itself, i.e., thickness, top width, driver locations, and so on.

Can you provide more specific information concerning the construction and installation of the crossovers and wiring? It might help you to understand my benightedness knowing that my construction experience is limited to building kits by following the numbered instructions. The anticipation of the sound from the finished project is intriguing, and I certainly will appreciate any help you can give me.

Mark Watanabe
San Francisco, CA 94122

As a strong proponent of the sound qualities of dipole speakers (mostly electrostatics), I was immediately intrigued by your design ("A Full-Range Open-Baffle System," *SB* 1/94, p. 30). My goal is a speaker that maintains the best qualities of a good electrostatic, but with greater dynamic range and better low-end extension and power. Since the literature is so sparse in this area, I wish to pose several questions to help me implement your design with any possible improvements.

1. Have you modified the design to include a 15″ subwoofer, and were the results as you expected, i.e., extended low-frequency cutoff and perhaps greater output with flatter response? If so, is it a biamplified design? I have plenty of clean power (500W) and would prefer a single amplifier, but can accommodate biamping if necessary. What 15″ driver did you choose and can you provide the crossover design?

2. Do you have any measurements or estimates of the system's dynamic sound-level range capabilities, especially with the 15″ subwoofer?

3. Given the large cone excursions required by the open baffle, have you measured the distortion levels?

4. To further minimize front panel vibrations, have you experimented with a double-thickness front panel (1½″) and did you find a noticeable difference?

5. Would a nonsymmetrical horizontal positioning of the drivers on the front panel further diversify the front to rear path lengths and further smooth the response? If so, do you have a recommendation on the horizontal offset?

6. The rear tweeter is shown mounted behind the baffle. What are the benefits of this mounting versus surface mounting?

7. Would you suggest any changes that further flatten room response without the need for a parametric equalizer? Or, is this so room-dependent that an external parametric equalizer is the only practical solution?

8. Are the drivers listed still the best available for this design?

9. Is a dimensions chart available for both 12″ and 15″ subwoofer designs?

10. Would a 10kΩ input impedance for the power amplifier require different values for the passive low-pass circuit? The preamp output impedance is well below 1kΩ.

11. What type of capacitors do you recom-

mend for the tweeter and subwoofer circuits ($4.7\mu F$ and $330\mu F$)?

12. Did you experiment with foam or felt padding on the front panel to further minimize diffraction? If not, do you expect this would make a noticeable improvement?

13. On p. 40, the low-pass network reactance, R1, is 0.74Ω, but is inserted in the $Q_{SUBWOOFER}$ formula as 2.0. Am I reading this correctly?

I would greatly appreciate any additional ideas you have on improving the design.

Charles G. Catledge
Marietta, GA 30068

Joseph Janni and Warren Hunt respond:

The replacement of the original tweeters and subwoofer is described in our reply to a letter from Roger Cox ("Knock 'em Dead," SB Mailbox, SB 3/94, p. 65). The dimensions of the 12" and 15" subwoofer versions are identical except that the bottom of the 12" subwoofer is 4" from the floor. Please refer to Photos 1 and 2 in our article for the following discussion. The dimension tolerances are not critical, and are about 1/8". These dimensions could be changed by as much as an inch—as long as it is done consistently—without substantially altering the sound, but you must preserve the symmetry and shape of the "Tombstone" design.

A) The two side panels are identical and are tapered, with a 4" minimum depth at the top and a maximum depth of 9" at the bottom. The height of the side panels is 52" measured perpendicularly from the floor along the rear vertical edge. The front edge of the side panels is not quite vertical, and slopes back with a 6° rearward tilt forcing the front panel to do the same. This tilt minimizes the formation of horizontal standing waves in the listening room and is very important to the sound quality.

B) The top to bottom height of the front panel is 54" measured from the floor, and as illustrated in Photo 2 *is 2" longer than the side panel height. The bottom width of the front panel is 24" and the top width is 14".*

C) The bottom of the 15" subwoofer is 2½" from the floor; the bottoms of the 10" drivers are 18" and 34½" from the floor. By the way, we are extremely pleased with the performance of the Madisound 1052DVC 10" drivers. The tweeter is centered equidistant between the 10" drivers. All drivers are centered horizontally in the front baffle.

D) The rear-firing tweeter is immediately behind the front-firing tweeter, and is mounted as close to it as practical. All drivers are front-surface mounted in the final design, including the rear-firing tweeter. Remember to wire the rear tweeter out-of-phase with the front tweeter. The $4.7\mu F$ series capacitor should be Mylar® or polypropylene, not electrolytic.

E) The top left and right of the front baffle is curved (radiused) for appearance, not for sound quality.

The subwoofer resonant crossover is as simple as we could make it. It consists of only two components. In the 15" subwoofer version it is simply an 18mH series inductor between the amplifier and the subwoofer, with a large $660\mu F$ nonpolarized electrolytic capacitor (35V rated) directly across (paralleled with) the subwoofer terminals. Two $330\mu F$ capacitors would do as well, and are more readily available (we got ours at a large computer parts supply house).

The crossover looks complicated in Fig. 2 *of our article because everything is shown in detail, including the amplifier internal resistance R1 in series with the inductor and the electrical subwoofer model on the right half of the* Fig. 2 *schematic. We used 16-gauge multistrand connecting wire throughout, including in the crossover.*

When we tried a biamplified design, the sound quality was virtually identical, probably because we were using a high-quality solid-state amplifier even though it was only 60W/channel. Only at very loud levels could we perceive a subtle improvement by biamping. Unlike the 12" subwoofer version, the 15" version requires an amplifier capable of driving 4Ω loads.

We have not measured the low-frequency distortion levels because we do not have the equipment to do so. Calculating the mid- and high-frequency regimes indicates that the distortion should be quite low because the cone excursions are small for those frequencies. We have not measured the maximum sound levels, but in our listening rooms they can sound very loud.

We did not experiment with a double thickness front panel—equivalent acoustically to a single thickness panel with twice the mass—but the physics of the design indicates that it might reduce the front panel cabinet talk a bit. No front panel vibration effects exist in the present design, but a heavier and stiffer front panel could only help and certainly couldn't hurt.

A nonsymmetrical horizontal positioning of the drivers to further diversify the pathlength distribution from front to rear is unwarranted; the tombstone shape is very effective at doing this already. Moving the drivers off-center will reduce the minimum front-to-rear pathlength Pm, which is a critical design parameter that must be as large as possible.

Moving the open baffles further into the listening room would noticeably smooth the in-room frequency response and further improve the subwoofer performance. This approach is preferable to using a parametric equalizer, which can be used—but more sparingly—if still needed. Be extremely cautious when using equalization to flatten the high-frequency response of any speaker because this process usually produces sound that seems too bright.

Although a bit unusual, an amplifier with a 10kΩ input impedance would definitely alter the performance of the passive low-pass circuit. Most solid-state amplifiers have an input impedance that is usually at least 50kΩ. Nevertheless, using this amplifier requires you to increase the value of C1 in Fig. 4 of our article from .22μF to 2.2μF to correctly attenuate extremely low frequencies below audibility. This circuit modification is very important and mandatory.

Of lesser importance, you should increase the value of R2 from 3.9kΩ to 4.7kΩ, although this change is fairly minor. Unfortunately, using 4.7kΩ increases the insertion loss by about 2dB. More importantly, it preserves the shape of the equalization curve within 1dB of our original design. The capacitors in the passive low-pass circuit should be either Mylar or polypropylene. Do not use electrolytics for this application.

We experimented with circular 2" felt rings around the tweeters on the front baffle. They produced an audible, but subtle, effect. We could not decide whether we liked the sound better with or without the felt rings, so we left them off.

The equation for the $Q_{SUBWOOFER}$ on page 40 is an approximation; it is not exact. The Q of the subwoofer and the Q of the low-pass resonant network can only be manipulated in this fashion if the driver resonance and the circuit resonance are identical, which is not rigorously true in our case. The low-pass network reactance is 0.74Ω only at the circuit resonance of 35Hz, and we believe you are interpreting it correctly. If the circuit resonance and subwoofer resonance had both been 35Hz, the 0.74Ω value would be the one to use. However, when you take other factors into account, 2Ω is a better approximation for the series effect of R_l at the subwoofer resonance of 25Hz as used in this approximate equation.

The section on Transient Responses on p. 40 has incorrect units (watts) for the amplifier internal resistance R_g (ohms). It was printed as 0.05W, which should have been 0.05Ω. Also be advised that the units of gain in the far right column of Fig. 7 *are decibels.*

IMAGE PROBLEM

As an avid reader of *SB* since its inaugural issue in 1980, I am happy to see this forum for our marketplace grow and prosper. The sharing of ideas and ensuing discussions are healthy and insightful for all of us.

That is the purpose of this letter, since I want to clarify some erroneous information about Image Communications (aka Waldom) that was presented as factual in "Foam Surround Repair" (TT & T, *SB* 5/93, p. 71) and in a subsequent follow-up from one of our authorized recone centers regarding foam surrounds ("Foam," *SB* Mailbox, *SB* 2/94, p. 65) and rebuttal from Len Moskowitz. I agree with many points Mr. Moskowitz states, but disagree with the comments regarding the nature of Image Communications, its operations, and competitiveness.

Since 1946, Image Communications has provided speaker repair kits to the audio repair marketplace. Our 800+ international authorized repair centers provide a blanket of high-quality, state-of-the-art speaker repair service to the audio marketplace. In Mr. Moskowitz's rebuttal to David Young in *SB* 2/94, he states, "While we hope they (Image) approximate the original manufacturer's component specs, the consumer can't be sure of this."

This is untrue. Our own Waldom brand kits utilize the identical cones, spiders, surrounds, and so forth, and are procured from the same component manufacturers that the OEMs use. Waldom kit components are specifically acquired from the same tooling that have run OEM parts, and are not factory overruns, seconds, or questionable quality.

Image does not manufacture cones, as Mr. Moskowitz also states, but is involved in value-added assembly such as adding surrounds to cone bodies, adding eyelets, and adjusting voice coil openings in cones. We do, however, manufacture voice coils to OEM specs.

In most cases, OEM speaker builders only assemble the components (cone, spider, surround, voice coil, and so on), and perhaps wind their own voice coils (as we do). The only time we offer a non-original part is when the original speaker is obsolete, in which case we offer the closest possible alternative, or if the original parts are proprietary or have outstanding patents (which are very few).

Many speaker parts are no longer manufactured, and we have built a successful complementary business on locating and providing these obsolete parts or equivalents. Contrary to Mr. Moskowitz's opinion, on average, a typical Waldom kit would cost 35–50% *less* than an OEM kit.

As a side note, we do work closely with some OEMs, and offer OEM kits as well. We provide original Electro-Voice kits, and are the exclusive supplier for Eminence kits worldwide. In the future, we will be working more closely with other OEMs as well.

In Mr. Moskowitz's "Foam Surround Repair" article, he states that "reconers replace the entire cone assembly." This isn't always the case; our recone centers frequently replace only the surrounds. The concern with this type of repair, however, is the possibility of other damage to the speaker parts due to the surround decay. In some cases, the voice coil should be inspected for damages since the surround provides a support for proper cone/voice coil excursion.

In summary, I agree with Mr. Moskowitz that diversity is good for all of us. It's one of the reasons Image has become the world's largest supplier of aftermarket speaker repair kits. Our authorized repair centers have also flourished because they provide parts and service at more competitive prices. They also advertise in *SB*, but I will not use this opportunity to "plug" my favorite vendors or customers. Any individuals with questions about Image, its authorized repair centers, or products should contact us toll free at (800) 552-1639.

Jay Peabody
Image Communications

SALVAGE OPERATION

I would appppreciate your help and suggestions to salvage a pair of old Altec Lansing speakers from the late '60s—my father's "The Avalons." I wish to make a two-way system, 16 or 8Ω, using the horns and perhaps "new woofers" in a smaller enclosure. Can you help?

The original literature describing these 16Ω speakers states, "The Avalon consists of a single 12" heavy-duty woofer, a compression-type high frequency driver coupled to a cast aluminum sectoral horn, and a full dual-section crossover network which provides attenuation at 12dB per octave above and below the 800-cycle crossover point. The bass is a reflex-type cabinet with a volume of 6.4 ft^3."

The horn unit 804A and the woofer 414z were used in this speaker. I obtained the following information about the woofer from Altec Lansing Co.:

1. free air resonance 30Hz
2. X_{MAX} 0.15
3. V_D 11.8
4. Q_{TS} 0.21
5. Q_{MS} 7.4
6. Q_{ES} 0.21
7. V_{AS} 8.13
8. ref. efficiency 2.9%

Unfortunately, I did not obtain any technical information about the horns, but they did tell me that the diaphragm of the horns is available and the woofers could be reconed.

Does this project have any hope?

Alex Hoffmaister
Potomac, MD 20854

Contributing Editor Bruce Edgar responds:

Although I'm unfamiliar with the particular Altec model, the components are standards of the Altec line. You can try one of two approaches: 1.) Use the components in an all-horn system. Unfortunately, the Altec woofer will roll off at 300Hz in a bass horn and the mid horn turns on above 800Hz, leaving a big hole between 300 and 800Hz. So an all-horn system with the existing Altec components is impracticable. 2.) Restore the Altec Avalon to its original state. Examine the woofer for the condition of the cone and surround and contact an Altec dealer for any needed repairs. Power up the system with music to determine how the high frequency horn responds. If you hear any anomalies, check with an Altec dealer for service. I hope these suggestions help.

Impedance

Continued from page 39

their phase at the crossover frequency. Ignoring this phase can result in a nonflat summed response.

CONCLUSION

More exotic dividing circuit design techniques exist for getting around a driver's impedance peak at resonance and its inductance. They also require additional resistances and reactances, otherwise a nonlinear optimization program is necessary. The techniques provided here should be food for thought for those of you who are dissatisfied with your present system, or if you are contemplating a new design and would like to head off problems at the outset.

Clearly, only inductance compensation may be required for a woofer voice coil and the full treatment for a midrange driver or a tweeter. Some networks are not very sensitive to the fundamental resonance, and this could go uncompensated. If you have patience for detail, you can gamble that the subjective listening test will favor the more fully neutralized design.

I can vouch for the improved definition of my dome midrange using the extended zobel network to compensate the voice coil inductance. The upper frequencies are cleaner than with the single-capacitor zobel network. My dividing network was conceived to be tolerant of the midrange's impedance peak at its fundamental resonance, so I have not applied complete neutralization of its voice coil impedance. Anyone using a network that capacitively feeds the midrange driver or tweeter should try the complete compensation.

I hope you will find the measurement and prediction techniques presented here to be useful, even if you don't have access to a computer and an optimization program. A programmable calculator can be very powerful when applied with the right theory.

IMPcycling

Continued from page 42

When just "hacking" a crossover, patch the network together during tuning using heavy-duty alligator clipleads, available at Radio Shack or other parts houses, usually in sets of ten. Don't trust the factory connections between the wire and the alligator clips; they are usually just pressed together and inadequate for loudspeaker currents. Pull the sleeve back on each alligator clip and solder the wire to the clip.

With your kit of crossover goodies and IMP showing an updated response plot every few seconds, you can quickly customize your system. This method represents quite a change from the days of crossing your fingers, installing a "stock" crossover and endlessly guessing at why the system sounded funny.

[A customized kit of IMPcycling parts is offered by Old Colony in its advertising elsewhere in this issue.–Eds.]

Linear Array

Continued from page 49

Figure 13 shows the result for the *Fig. 4* test case, but at 2kHz. The sound is getting less directional at the lower frequency, with diffraction limited by the width of your window or speaker system.

Figure 14 is the last case, with the sound source moved up to 3′ from the mike, in the center. *Figure 15* gives the result at 2kHz. Notice the wide angle over which the output is strong, now over 60°.

It's difficult to visualize what these plots mean in terms of the apparent image position for a listener, but in general they suggest things appear in the right spot. In any event, this simulation doesn't indicate what the system will sound like. I'll keep you posted. My FAX number is 619-727-5243, and you can email me at: pwitham@crash.cts.com.

Loudspeaker Recipes

Continued from page 55

sale to those potential builders who would like to lay out and cut wood from such drawings.

2) Besides the odd photo, there should be exact layouts of crossovers showing size of the base board and location of the rivets. There is no picture to accompany the text about the advantages of certain coil orientations.

This book is so comprehensive it is bedtime reading for a year. It is full of useful ideas such as putting crossovers inside the stands, Well-nut© driver mounting, Sonotube© sand-filled cavities, damping treatments inside the cabinets, proper dowel bracing, spike and pad isolation, just to name a few. Any one of the systems described would grace the Queen's drawing room and make wonderful music.

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41	42	43	44	45	46	47	48	49	50

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CLASSIFIED INFORMATION

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

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

WANTED: Help readers find equipment or services.

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

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

PRICING

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

AD COPY

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

SUBSCRIBERS

Receive ***free*** *For Sale, Wanted,* and *Club* advertising; 50 words maximum; each additional word just $.20. Please only one ad per category, per issue.

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

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

FOR SALE

Klipschorn Parts: Never used. K-77-M Tweeters, K-55-M midrange drivers with horn lenses, crossover parts, and schematic too. Update your corner horns or top off your own folded horn. I paid $710, for $500 I will ship all to you. Taylor or Anita, (708) 940-0409.

Aria 3 kit with hard-wired crossovers, no cabs, used 100 hours ($1,200) $650 or make an offer; Focal Audiom 7K midrange ($175) ea. $170/pr.; Focal 8K415's woofers - 90' version, $95 ea., $95/pr. Norris Wilson, 305 Farallone Ave., Fircrest, WA 98466, (206) 564-7627.

Dynaco PAT-4, in working condition, $25 plus shipping. Dynaco FM-3, in working condition, $40 plus shipping. Bob Grieb, 322 Tavistock Dr., Medford, NJ 08055, (609) 953-5519.

Macintosh "SE/30" computer. Eight megabytes RAM, 80 megabyte disk. Color card for external monitor; Sony high-resolution monitor. Both monitors no burn-in. Extended keyboard, cables, new mouse. Excellent condition, all original packaging and software. $1600, I pay shipping (without monitor $1250). Dave Bessey, 559 Valencia St., San Francisco, CA 94110-1114.

Two Revox A77 Dolby B, low hours, excellent, $350 each; two Revox A700 with built-in mixer, excellent, $550 each; all Revoxes quarter track; Sony P52000 servo turntable system with arm and cover , mint, $350. Andy, (516) 269-9796 (NY) after 7 PM EST.

MicroSeiki turntable: mod. DQ-3, massive direct-drive platter, 33/45 quartz-controlled, lightweight tubular arm adjustable in all planes, will track below 1 gm., free Shure M-95 cartridge. $250/offer + shipping. Lou Garner, 1440 S. Valley View, #2004, Las Vegas, NV 89102.

Matching pair of vintage KEF speakers, model K-2 Celeste MKIIs, made in England, sold by Radio People, Ltd. Kowloon, Hong Kong, with matching serial numbers, size 6½" x 18" x 11", collector quality, in top shape, solid wood construction, rare units in perfect condition, price $200, I pay shipping, (305) 931-7131.

Mitey Mic, assembled with TLO62 IC, $120. Two Ballard crossovers from *SB* 3/82, partially assembled PWAs, $50. IMP parts, all resistors and caps only, 4 sets, $10 ea. Crown PSA-2 variable active crossover, $135. Contact Mike at (414) 965-4550.

Audio Cyclopedia, Second Edition; turntables: Dual 1257 model and Connoisseur BD2/A model; Dyna Stereo 70 parts, power transformer, two 7199 tubes. All best offer. Phone (216) 534-4225 after 5:00 PM weekdays, EST.

Vieta L-1004 11" woofer, $130/pr.; Focal T90T102, $50/pr.; Zalytron DBL6 cabinets, bio, Tara QII 201PR speaker cable, $100; AudioControl Richter Scale 3, $165, Ed, (817) 244-5763 (TX).

Soundcraftsman SE450 equalizer plus test LP, $65; Electrovoice Sentry IVB x-overs, $40/pr.; four Precision T-305F swan bass module woofers, $120; four Focal 8V416 (Brother John) bass drivers, $110/pr.; University N2A or N2B x-overs, $25/pr. Shipping extra. Jim, (708) 425-6719.

Professional designer cleans out his bone pile. Cabinets, drivers, parts, cash and carry only, Vista, CA, (619) 598-3763.

HP 180A oscilloscope and cart., $200 + shipping. HP 711A power supply, $100. J. Curtis, PO Box 758, Stevens Point, WI 54481, (715) 341-5880, FAX (715) 341-3708.

Tektronix 2213A 60 MHz dual trace oscilloscope with accessory pouch and two probes. Tucker price, $895. $700, Ashby, (305) 475-5225.

Selling off my personal collection of Dyna tube and transistor equipment. SASE for list. B. Brown, RR 1, Box 50, Valley Springs, SD 57068.

Dynaco PAT-4* preamp and Stereo 120* amp; Fisher FM-100-B* tuner; Heath signal generator IG-18*, Heath frequency counter IM-2410; Harman Kardon 230 receiver; Garrard Zero 2000-B turntable; Sansui electronic crossover CD-5; Pioneer eight track recorder H-R99; (* has manual), Kent Johnson, (303) 989-8272.

Information on IMF RSPM Mark IV loudspeaker. Trying to revive a pair, replacement drivers no longer available. We need: copy of service or owner's manual, information on source or TS parameters for racetrack-shaped woofers, or other drivers. Any help appreciated. Bob Grieb, 322 Tavistock Dr., Medford, NJ 08055, (609) 953-5519.

Pair of Carver Silver Seven T mono amps, mint condition, boxes and manual, $1,000. Len, (410) 247-9581.

Awesome pro-type speaker system: Altec 817 (half-sized VOT) subwoofers loaded with Gauss Alnico 800W, 30Hz 15" woofers, with pole-attached Altec 9849 satellite speakers, each with 300W JBL 12" and 2425 compression driver. Outrageous fullrange sound. $1,500/pr., split shipping. David, (914) 688-5024.

B+K Pro 5 preamp, mint condition, $250US/350CDN; Audio Logic crossover 3-way stereo or 5-way mono, 18dB slope, very clean, $250US/350CDN; Bi-amp crossover 2-way stereo, $100US/140CDN; Focal mini-monitor T-120K/SK-013L - SCR CAPS/Solen, external crossover, gold posts, 1¼ MDF, finished in black, pro built, cost $500 to build, $325US/450CDN. Call Andrew McCrae, (514) 281-7954.

WANTED

KEF B139 woofer SP1044. Randy, (206) 953-4734, leave message.

Ohm A loudspeakers, Jerry, (908) 522-6093, weekdays.

Recordings: Fairport Convention, "Bonny Bunch of Roses"; Steeleye Span, "Hark! The Village Wait"; various Fairport and Steeleye Span spin-offs; Caravan, call; Nick Cave and the Bad Seeds, call; French, Frith, Kaiser, Thompson, "Love, Life, Larf, Loaf" (CD only); Van der Graff Generator, "The Long Hello". David, (814) 862-3505.

Bottom view of C5 printed circuit board from Dynaco PAS3X preamp and schematic M15T University crossover. Network R-E Fabreguettes, Lot Pré Vescal, 05000 Rambaud, France.

50-80W transistor stereo power amp, one 12" D.V.C. woofer, pair 10" woofers. Jim, (708) 425-6719.

Schematic for Phase Linear 400 Series II power amp. Will pay for copies and postage. Matthew Honnert, 815 Burning Trail, Carol Stream, IL 60188.

Quad ELS. California Institute of Technology Music Lab needs gifts of several pairs of original Quad electrostats (not ESL-63). James Boyk, (818) 395-4590, FAX (310) 470-9140, Email: boyk@puck.caltech,edu.

Preamp schematic: Convergent Audio SL-1, Signature CJ, Premier 7; tubes, NOS: WE, Mullard, CBS, Amperex, Sylvania, KT88, KT99, 6550A, 6LF6, 6L6G, 5AR4, 6CA7, 7308, 7025A, 6681, 12AX7A, 6201, 6679, 6922, 12BH7A. Please offer the prices. Lawrence Yip, 13/F Jardine House, 1 Connaught Place Central, Hong Kong, (852) 844-9313.

PR Altec 301 speakers in good condition. I work swings. Please write. William Irvin, 4300 Cullen Dr., Cleveland, OH 44105.

Pair Bozak B-407A Monitor C loudspeakers, SWTP headphone amplifier. Tom Vetromile, (206) 546-0857, evenings (Seattle).

Hafler XL-600, (817) 244-3913.

Technics EPA-500 or EPA-100 tonearm. Stax Lambda Signature headphones - headset only, no step-up device needed. Call Brian, (206) 568-6757 or 206-335-2389 (day/message).

CLUBS

WANTED: SPEAKER AND AUDIO AMATEURS IN THE BRADENTON/SARASOTA/ST. PETERSBURG/TAMPA, FL AREAS. Would like to form a club and develop a lab for testing speakers/amps/preamps and passive and active crossovers or just to discuss speaker projects and ideas. Angel Rivera, Bradenton, FL 34206, (813) 792-3870.

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

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

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.

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.

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

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

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 SASE to: Barry Waldron, 1847 Country Club Dr., Placerville, CA 95667.

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

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.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues include 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 Frank J. Alles, (908) 424-0463, 209 Second St., Middlesex, NJ 08846; or Bob Young, (908) 381-6269; or Bob Clark, (908) 647-0194.

THE COLORADO AUDIO SOCIETY (CAS) 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 joumal, five newsletters, plus participation in meetings and lectures. For more information, send SASE to CAS, 1941 S. Grant St., Denver, CO 80210, (303) 733-1613.

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

ESL BUILDERS GROUP is a new address for people who have built or want to build electrostatic loudspeakers and associated (tube) drivers, or are just interested? An answer is ensured, if you include some kind of compensation for postage and handling. Write to: Gunter Roehricht, Bühlerstr. 21, 7030 Böblingen, Germany.

MONTREAL SPEAKER BUILDER CLUB. Meets when it can, BYOB, discussions range from speaker design and testing to equipment modification. All welcome. Contact Andrew McCrae, 4701 Jeanne Mance, Montreal, PQ, H2V 4J5 Canada, (514) 281-7954.

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.

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.

WASATCH AUDIO, located in Salt Lake City. Our club is interested in construction, modifications, design, and listening to music. We are looking for members and ideas for our new club. Contact Edward Aho, (801) 364-4204.

HI-FI COLLECTOR/HOBBYIST seeks audio penpals to correspond via reel-to-reel tape. If you are into restorations, discarding old recorders, or parting out derelict gear, or have arcane technical secrets you'd like to discuss, or just want an excuse to obsess over mylar, make it so. I'll return all tapes via parcel post. James Addison, 171 Hartford Rd., A-7, New Britain, CT 06053.

LONDON LIVE DIY HI-FI CIRCLE meets quarterly in London, England. Our agenda is a broad one, including any aspect of audio design and construction. Subscription newsletter. We welcome all, from novice to expert, in free association. Contact Brian Stenning, UK, 081-748-7489.

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

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

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

Reader Service #21

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

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

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

VINTAGE AUDIO LISTENERS AND VALVE ENTHUSIASTS (VALVE) meets the first Sunday of every month to swap vintage audio gear, audition rare and collectible equipment, and evaluate modifications and scratchbuilt projects. Dues provide a monthly newsletter with current reviews of vintage components and modification information; vintage service data; and access to an active network of serious collectors. For information, call (206) 697-1936 or write to 1127 NW Brite Star Ln., Poulsbo, WA 98370.

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, write to WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, design analyses, digital audio, AB listening tests, equipment clinics, and audio fun. Club publication, *LC, The SMWTMS Network*, journals the club's activities and members' thoughts on audio. For information, send name and address: e-mail aa259@detroit.freenet.org; phone (810) 544-8453 (machine); letter SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

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

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 3,000 "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.

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 Alexandria, VA. Prospective members are welcome but must register in advance in order to be admitted to the meetings. No exceptions please. Call Horace Vignale, (703) 578-4929.

Ad Index

AUDAX KIT PLANS

Speaker Kit Plans Are Now Available From Parts Express!

"Kit Plans, Build Your Own Loudspeakers" is a 25 page booklet containing plans for 7 speaker kits. The Signature Series has 4 different kits utilizing Prestige and Reference Series components and was designed exclusively by Vance Dickason, author of "The Loudspeaker Design Cookbook". The enclosures, crossovers, and components were carefully selected by Mr. Dickason and computer optimized to ensure the highest level of performance. The Classic Series was designed in house by Polydax and includes 3 easy-to-build kits that offer great sound quality at an affordable price. All of the components and accessories you need to build any of these projects are available and in stock at Parts Express.

To receive the booklet "Kit Plans, Build Your Own Loudspeakers", and a copy of the 188 page Parts Express 1994 Catalog, just send $1.00 to cover postage and handling ($3.00 to Canada).

SIGNATURE
Vance Dickason
SERIES

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