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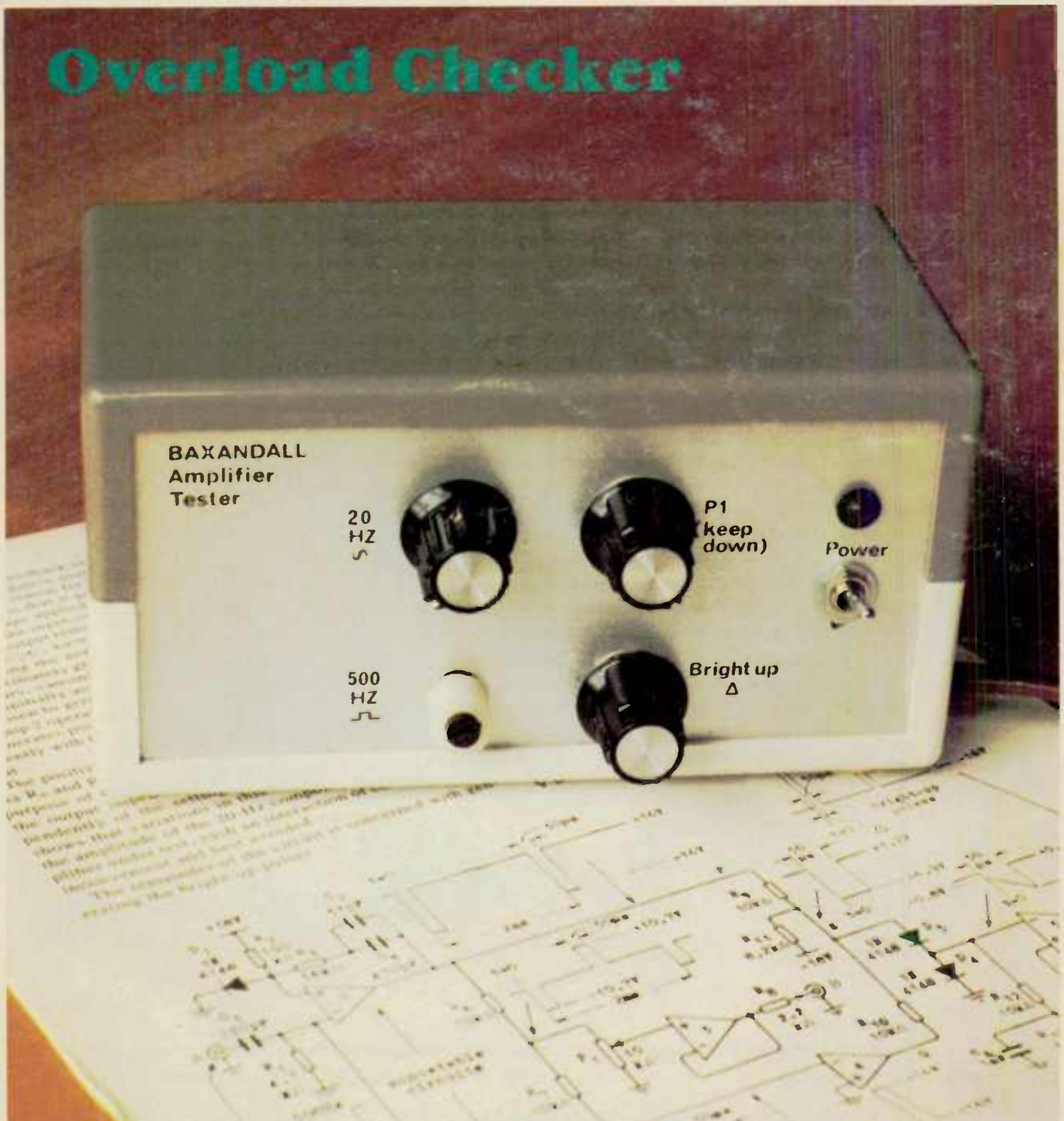
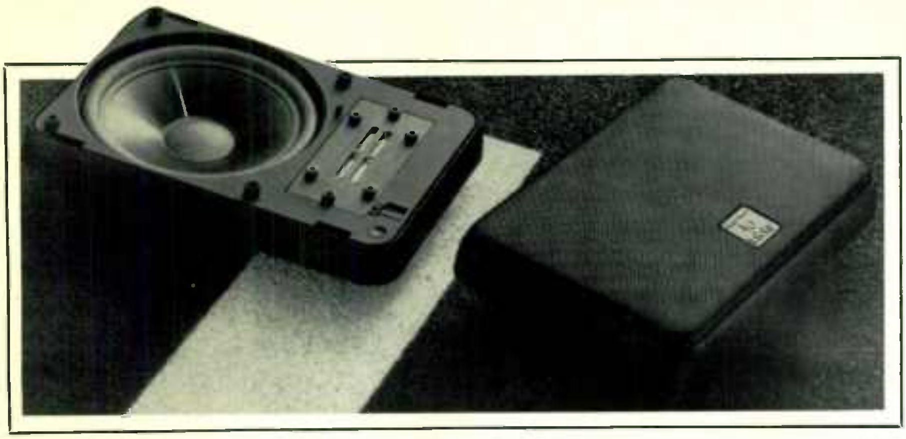
Speaker Builder

THE LOUDSPEAKER JOURNAL

Overload Checker



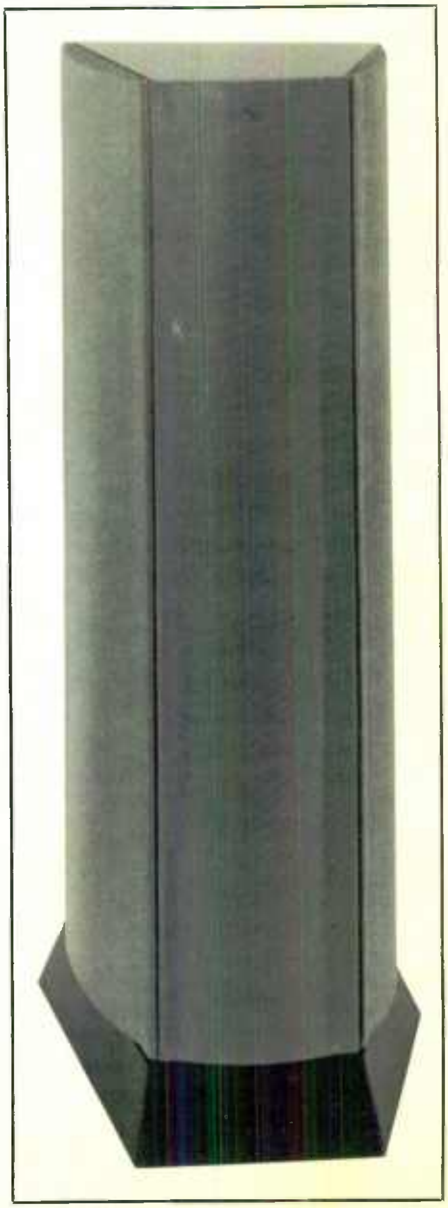
Good News



A newly designed line of loudspeakers incorporating **dbx**'s Soundfield Imaging technology is now available. The three new systems feature the styling of their flagship model SF 50, introduced in 1987, with a gray/black or beige/walnut color scheme and wrap-around, curved speaker grilles.

The dbx SF 150, a three-way, floor standing, vented loudspeaker, will retail for under $1,500/pair; the SF 1500, under $1,000/pair; and the two-way SF 2500, $500/pair. Contact: dbx, PO Box 100C, Newton, MA 02195, or call (212) 661-5300.

Fast Reply #JC85

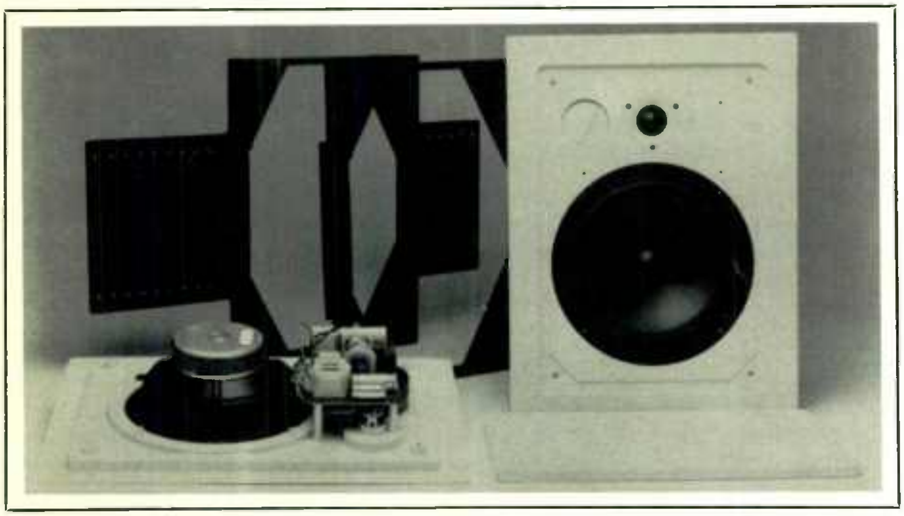


INFINITY's innovative 6.5-inch automotive plate speaker joins their reference standard Kappa automotive series. A two-way design, the new model features an Emit tweeter with neodymium magnet, an injection-molded graphite (IMG) woofer and high technology componentry, to reproduce vocal and instrumental music with the same accuracy as Infinity's home systems.

The plate system requires only a 4½-inch cutout and shallow mounting depth, and is suitable for a variety of foreign and domestic, compact and full-sized vehicles. It may be used in other mobile applications such as boats, and also wall-mounted in custom home designs.

Suggested retail price is $400/pair. Contact: Infinity Systems Inc., 9409 Owensworth Ave., Chatsworth, CA 91311, or call (818) 761-8838.

Fast Reply #JC354



SONANCE, one of the largest manufacturers of in-wall high fidelity speakers, has introduced the smallest system in its line, the M-30, which utilizes a 4-inch woofer and 1-inch polycarbonate tweeter.

Its size, 9.1 by 6.5 by 2.2 inches, makes it ideal for in-wall locations such as bathrooms, saunas, showers and small kitchens. The system is available with either retrofit or new construction brackets for quick installation; about 15 minutes. The brackets can be used to mount the speaker vertically or horizontally. The new Sonance line will all utilize the same mounting brackets.

The M-30 includes a cloth or metal grille, which can be decorated to match any interior design.

Suggested retail price range is $300–400/pair, brackets included. Contact: Sonance, 32992 Calle Perfecto, San Juan Capistrano, CA 92675, or call (212) 986-6668.

Fast Reply #JC107

SPEAKER BUILDER

Edward T. Dell, Jr. *Editor/Publisher*

Contributing Editors

Robert M. Bullock, Joseph D'Appolito, David Davenport, Vance Dickason, Bruce C. Edgar, Gary Galo, G. R. Koonce

Paul Battenfeld *Managing Editor*
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Christine Holt *Production*
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Techart Associates *Drawings*

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

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

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ELECTRO-VOICE announces the Satellite Sub-Scoop system, with two S-1202, 300W speakers and the 400W, SH-1810L subwoofer. The complete system comes with tripod speaker stands.

The benefits of the system are said to be a competitive price, and the flexibility of the nondirectional bass enclosure and elevated, compact main speakers to provide optimal room placement for complete audience coverage.

The subwoofer is a hybrid system combining vented and horn-loaded design, which yields high efficiency as well as extended bass response, for advantages that any full-range system can benefit from.

For more information, contact Electro-Voice, 600 Cecil St., Buchanan, MI 49107, (616) 695-6831.

Fast Reply #JC453

Acoustic suspension, acoustic blanket, Tetra-Helix tweeter plate, acoustic lens, and shielded drivers are featured in a compact box from **ACOUSTIC RESEARCH**.

The AR TSW 115Ps, with active circuitry, include the quality appearance of their counterparts in the TSW series, and are small and versatile enough for any home setting. Like the larger versions, AR Powered Partners, the TSW 115Ps have numerous uses, for example, plugged into a portable CD player, personal stereo, or MTS TV.

The single amplifier controls both channels and delivers 16W per channel. The 115P and the TSW 105, the passive version, feature a 4-inch woofer and titanium dome, liquid cooled tweeter.

In addition, a stage monitor type, acoustic suspension speaker, the Party Partner, features a 10-inch long throw woofer and 1¼-inch cone tweeter, in a wedge shaped, vinyl veneer cabinet. The Party Partner's suggested retail price is $225 each; TSW 105, $275/pair; TSW 115P, $400/pair.

For more information: Teledyne Acoustic Research, 330 Turnpike St., Canton, MA 02021, (617) 821-2300 or 1-800-225-9847.

Fast Reply #JC324

About This Issue

How are your amplifier and your speakers getting along? What happens when the signal drives the amp into overload protection mode? **Peter Baxandall** wondered about this and has come up with a device to analyze the problem (page 9). Contributing Editor **G. R. Koonce**, now on his third computer, shares his conviction that QB_3 vented boxes are best and presents supporting data for his view (page 22).

If you've never seen a truncated, convergent pentagonal box/tower before, now is your chance. **Bob Schoen** shares his adventure with his elegant system, starting on page 26.

Joe and Jim, the Swan twins, are back (page 34) with a masterpiece of construction description, including all the details on how to build version four of the Swan. The **D'Appolito/Bock** team will soon be offering a limited edition of the system to some lucky owners.

Contributing Editor **Bob Bullock** reviews the Marchand active crossover on page 46 and Editors **Koonce** and **Galo** present some new software they have written for box and driver analysis as well as passive and active crossover design (page 49). We begin a new feature on the human foibles which seem to accompany the audio mania, on page 48. Although **Dick Pierce** doesn't like our title, we think you'll like his special mordant viewpoint whether you like our title or not.

Cover photo by **Reg Williamson**.

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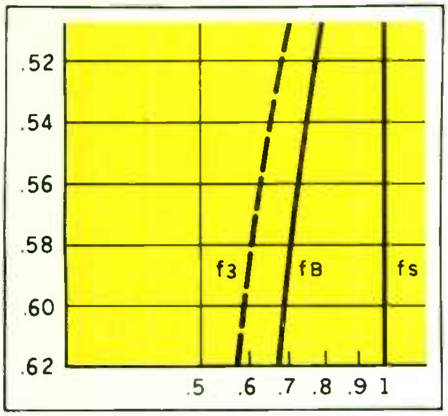

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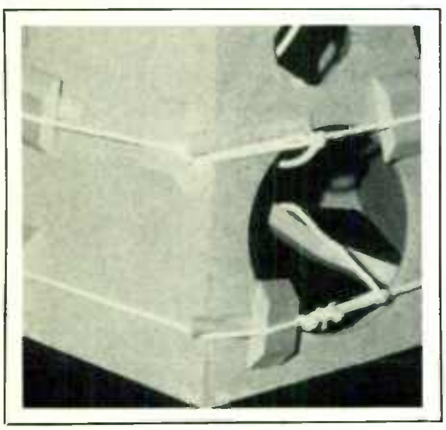
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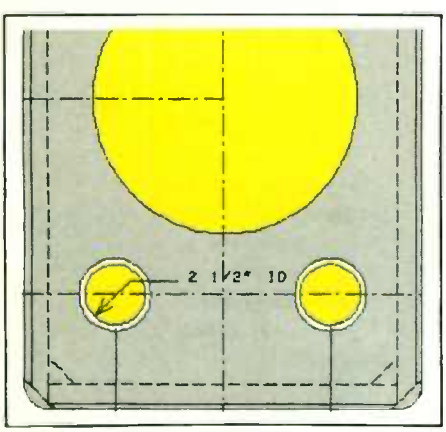


34

FEATURES

DEPARTMENTS

Editorial

Polarity Problems

Some of you may be more than mildly puzzled to notice that the lead article in this issue details a new device for testing amplifier overload characteristics. What is this doing in a loudspeaker magazine?

Peter Baxandall's article is here because it is also giving us clues about the importance of the amplifier/loudspeaker interface. It seemed important to place this article in these pages because we are too often tempted to forget the significance of what kind of interaction takes place between these two devices.

Author Baxandall, although famous for his electronic designs, especially his filters and justly admired tone controls, is also knowledgeable about loudspeakers. A check of his credits includes some remarkable work on a single driver system which included a "crossover." (See *Audio Amateur* 2, 3/1970 or *Audio Amateur Loudspeaker Anthology*, pp. 3-11.) His most recent work appears in the new *Loudspeaker and Headphone Handbook*, an anthology edited by John Borwick (Butterworths, 1988), where he writes comprehensively about electrostatic theory and practice.

The article is also appearing here because it is one more affirmation on this Editor's part that distortion measurements are vitally important to the search for good sound. I also happen to believe that instruments do not yet tell us all we need to know about the behavior of both amplifiers and loudspeakers.

Unfortunately, at least two polarized groups of people seem to believe I do not exist. Despite their opinions, however, I wish to affirm that I am among those whom the engineers accuse of being able to believe two impossible things before breakfast. I believe distortion measurements are important and we need more of them. I also hold equally devoutly to the belief that audible differences in sound may reliably be detected by humans and these clues are important to further refinement of the equipment.

I want to suggest that you read at least two articles and a book which may make some contribution to the seemingly endless discussions of the objective/subjective issue in audio. The most recent champion of the "objective" position is Douglas Self, with articles appearing in two UK journals, *Electronics & Wireless World* (July 1988), and *Hi-Fi News & Record Review* (August 1988). In the first Mr. Self sums up what he calls "the Subjectivist Manifesto" as follows:

> "Objective measurements of an amplifier's performance are unimportant compared with the subjective impression gained by listening tests. Where the two contradict the objective results may be dismissed.
>
> "It therefore follows that degradation effects exist in amplifiers that are unknown to orthodox engineering science, and that are not revealed by the usual objective tests.
>
> "Considerable latitude may therefore be employed in suggesting hypothetical mechanisms of audio impairment, such as mysterious capacitor shortcomings and subtle cable defects, without reference to the plausibility of the concept, or the gathering of objective evidence of any kind."

Mr. Self seems to have missed a few things. He apparently ignores the work of Walt Jung and John Curl on capacitor quality, published in *Audio Amateur* (4/1985, pp. 22–24). He also seems unaware that the British edition of the magazine *Elektor Electronics*, whose articles are written by staff engineers, published their concurrence with the work of Jung and Curl on the relative quality of capacitors (January, 1987). They repeated the Jung/Curl test and reported the results were reason enough to include higher quality capacitors in a preamp design.

Mr. Self also says "Presumably, the whole existence of music as a source of pleasure is an accidental artifact of our remarkable powers of speech perception..."

Music is not related to speech perception at all. The perception of music takes place in the right half of the brain. Speech and analytical reasoning take place in the left half of the brain. I recommend an interesting article "Towards a Biology of Music" by Andrew Stiller (*Opus*, August 1987)[1]. The article reports on experiments made at New York University in 1983, where positron-emission tomography (PET) scans of the brains of those listening to music showed the activity to be entirely in the right side unless the subject was trained in music.

Neurologists and those in other scientific disciplines have lately had quite a lot to say about music and its relationship to the human experience. See especially Oliver Sacks' *The Man Who Mistook His Wife for a Hat* (Summit Books, 1986).

I suspect that the left/right brain facts have quite a lot to do with the confusion and difficulties that attend double blind listening tests. I am not the first to make this suggestion, but certainly we need more work in this area before we can be sure that the tests are not asking something which our biological makeup is not presently trained to handle.

The Self articles have been brilliantly answered in the August issue of *Stereophile* by John Atkinson, who brings a far wider perspective to the issues.

All of us who love good sound and are concerned about the objectivist/subjectivist issue should also be talking to professionals in other fields. For nearly a century the scientific style of intellectual activity has grown steadily in popularity. The objectivist idea has become, for most intellectual disciplines, the only respectable standard. Fortunately, that is now changing in many fields, where those searching for truth about the world realize that the rigorously objective position is too narrow to include much that is obviously relevant to the human condition and to our understanding of the nature of the world. This is not to say that verifiable, repeatable experimental data is irrelevant, but that it may not be the only data that is important.

At least Mr. Baxandall, a dedicated and skilled objectivist, has devised this new test of how overload topologies function when attached to a variety of speaker loads. I hope many other "objectivists" will follow his example.—E.T.D.

1. You will, unfortunately have to go to the library to find the article since *Opus* is no longer published.

TESTING THE AMPLIFIER/ LOUDSPEAKER INTERFACE

BY PETER BAXANDALL

The amplifier under test is fed with a large low-frequency sine-wave input on which 50μsec pulses of alternate polarity are superimposed. These pulses drive the amplifier's protection circuits into current limiting, and the setup gives a CRT presentation having current-limit values vertically, and instantaneous output voltage horizontally. The displays obtained, which sometimes disclose unsuspected amplifier design faults, are compared with those derived in tests made on loudspeakers using program and other inputs.

ABOUT THE AUTHOR

Peter J. Baxandall was born in England in 1921, obtained a B.Sc. (engineering) degree from Cardiff Technical College in 1942. After two years as a lecturer/instructor in the Fleet Air Arm radio training scheme at the college, he moved to Malvern in 1944 to work in a government radar establishment known as TRE (now RSRE).

After some initial work designing microwave test gear, he gravitated toward the circuitry field, under Professor F.C. Williams, who had been closely associated with A.D. Blumlein. Work included the design of magnetic modulators, low-noise amplifiers, oscillators, wide-dynamic-range phase-sensitive detectors, and active filters, largely for use in research work being pursued in the physics department.

An interest in music and sound reproduction dating back to his school days resulted in the spare-time evolution of a negative-feedback tone-control circuit, now well known, in 1950.

In 1971 Mr. Baxandall left RSRE and became an independent electroacoustical consultant. Since that time his projects have included converter circuits for microphones, protective circuitry for loudspeakers, RF bridge circuits for capacitor microphones, amplifiers and power supplies for measuring microphones, a capacitive system for investigating loudspeaker diaphragm break-up modes, the design of oscillators for testing loudspeakers and microphones, and more recently, the design of some special, very high precision test gear for the National Physical Laboratory, to enable them to check the accuracy and long term stability of the setup for the reciprocity calibration of measuring microphones.

Transistor amplifiers usually incorporate protective circuit arrangements to limit the maximum peak instantaneous current that can be turned on, and the magnitude of this maximum current is often made to depend on the instantaneous output voltage.

Such protective circuitry frequently makes an amplifier unable to produce as large a signal-voltage swing across a complex, that is, partially reactive, load impedance, as it can across a resistive load. This sometimes leads to a disappointing performance when feeding typical loudspeakers.

The amplifier's protective circuit characteristics are not normally included in the specifications, and the purpose of my proposed test, described here, is to enable such characteristics to be examined readily.

I have found that the use of the present testing method sometimes discloses misbehavior of protective circuitry, in the form of overprotection, asymmetrical action, or RF parasitic oscillation, and therefore, that application of this technique during the design and development of amplifiers would be advantageous in enabling us to detect and eliminate such misbehavior.

THE BASIC TECHNIQUE. The essence of the scheme is shown in *Fig. 1*, and its functioning will now be described.

We apply a low-frequency sine-wave input, typically at about 20Hz, to the amplifier under test, its amplitude being sufficient to cause the amplifier output voltage to swing between voltage-clipping limits. Because the reactance of the 40μF capacitor on the output is about 200Ω at 20Hz, the amplifier does not give much low-frequency output current.

Superimposed on the low-frequency input is a waveform consisting of alter-

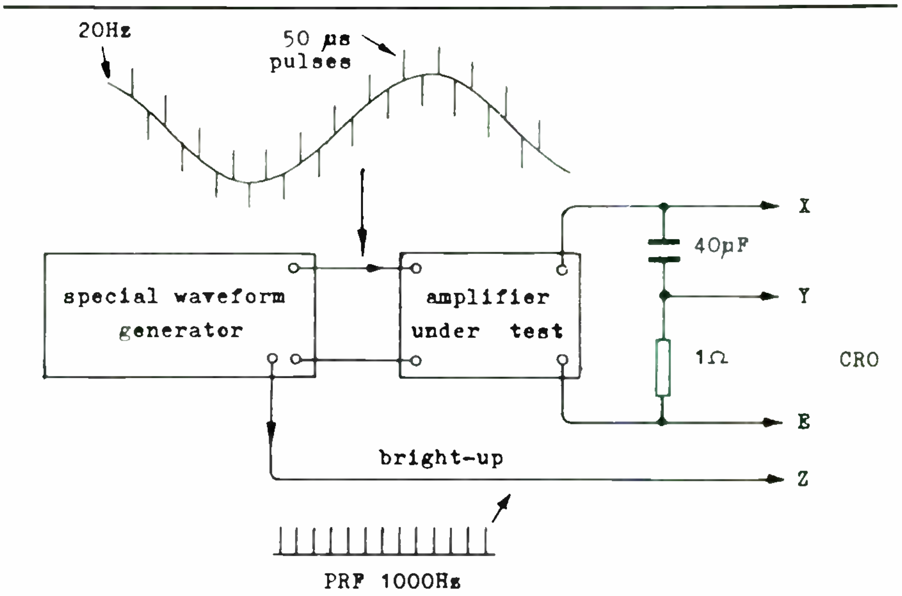


FIGURE 1: Essence of the scheme.

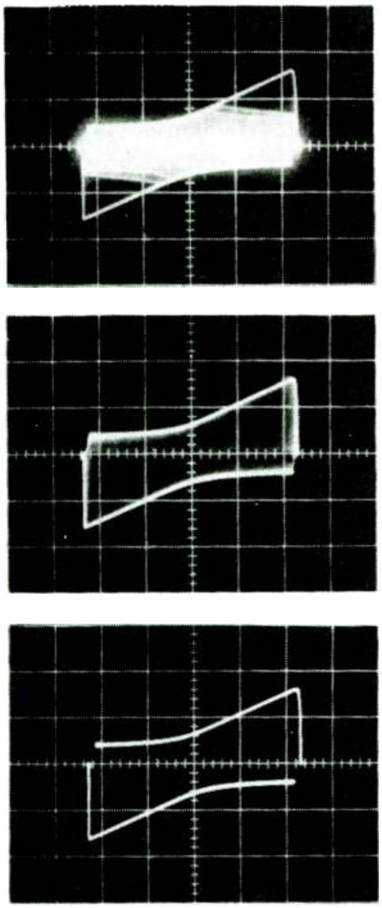

(a)
(b)
(c)

FIGURE 2: V-I displays for early version of the Quad 405 amp. (a) With no bright-up facility; (b) With 50µsec nondelayed-start bright-up pulses; (c) With start of bright-up pulses delayed by about 20µsec. Scale—vertical 5A/div; horizontal, 20V/div.

nate positive and negative 50µsec pulses with a fundamental frequency of normally about 500Hz. For the pulse component of the amplifier output, the 40µF capacitor has a very low effective impedance, so that large pulse output currents are produced. We make the amplitude of the input pulses sufficient to take the amplifier alternately into its positive and negative current-overload limits.

By feeding the amplifier output voltage to the X plates of an oscilloscope, and the voltage across the 1Ω resistor (representing output current) to the Y plates, the characteristic of the current-limiter circuit within the amplifier is displayed.

The waveform-generating circuit in *Fig. 1* also produces bright-up pulses for feeding to the Z-modulation input of the oscilloscope. These pulses are preferably somewhat shorter than that of the 50µsec pulses fed to the amplifier; the leading edge of adjustable timing being delayed by typically 20µsec after the leading edge of the amplifier-input pulse. This ensures that the amplifier output current has time to reach its limiting value before the trace is brightened up.

If you have no bright-up facility on your 'scope, a very messy and halated display is obtained. *Figure 2a* shows an example. With 50µsec nondelayed-start, bright-up pulses, the display of *Fig. 2b* is produced. Delaying the start improves the picture to that shown in *Fig. 2c*. A more striking improvement is sometimes obtained, depending on the nature of the amplifier being tested.

Though the figures of 20Hz, 500Hz and 50µsec adopted in this description will usually represent a suitable practical choice, you should not, of course, regard them as inviolable. Also, as I explain later, a minor advantage exists in using a lower value of current-monitoring resistor than 1Ω.

THE WAVEFORM-GENERATING CIRCUIT. The circuit for producing the required pulse waveforms is shown in *Fig. 3*. I incorporated it into the system as shown in *Fig. 4*.

The potentiometer in *Fig. 4*, in combination with the oscilloscope X-gain adjustment, if provided, is set to give an appropriate scaling, such as 1 division for 20V instantaneous output from the amplifier.

Some oscilloscopes give a spot movement to the left for a positive input to the X terminal, and you may then consider it worthwhile to insert a phase-inverting operational amplifier (op amp) stage after the potentiometer, thus ensuring that the display is on the normal conventional sign basis. I did this for the photographs reproduced here, using a Telequipment S54A oscilloscope.

Referring now to the detailed circuit of *Fig. 3*, input A is a 500Hz square wave, supplied, in my setup, by a Levell type TG200DM oscillator. The amplitude is uncritical and can be anywhere within the range 4–14V peak-to-peak. This square wave is differentiated by C1R1, whose time constant is approximately 1µsec.

Most of the time, the noninverting input of op amp 1 is at about +0.6V, owing to current flowing down R2 into diode D1, and this causes the op amp output voltage to be at its positive overload-limit value of about +16V. The occurrence of a positive-going pip at the inverting input then initiates a negative-going change at the op amp output, positive feedback via R3 ensuring that the output reaches its negative overload limit of approximately −16V. The reason for including resistor R3 in the feedback path is that it prevents the magnitude of the negative voltage applied to the non-inverting input from exceeding the input-voltage rating of the op amp.

The op amp output remains at this negative overload voltage until C2/C3 have had time to charge up sufficiently to bring the noninverting input terminal back up to approximately ground level. Regenerative action again occurs, causing the circuit to revert rapidly to the state initially assumed. The value of C2 plus C3 has been chosen to give a 50µsec negative pulse duration.

Op amp 2 operates similarly as a monostable circuit, but generates positive going 50µsec pulses, starting coincidently with the negative-going pips on the inverting input.

The positive and negative 50µsec pulses are combined via R6 and R7 and controlled in amplitude by P1. The purpose of the follower, op amp 3, is to ensure that the output impedance at socket B is constant, independent of the setting of P1. As *Fig. 4* shows, variations in this impedance would affect the amplitude of the 20Hz component fed to the amplifier under test; such an interaction of controls is inconvenient and best avoided.

The remainder of the circuit is concerned with generating the bright-up pulses.

Op amp 4 inverts the 500Hz positive-going 50µsec pulses from op amp 2, so that negative-going pulses occur on points P and Q, interleaved in timing. These pulse waveforms are added by means of R9 and R10. If no other connection was made to point S where these resistors join, the negative-going pulses there, at a repetition frequency of 1kHz, would have an excursion from approximately +16V down to ground.

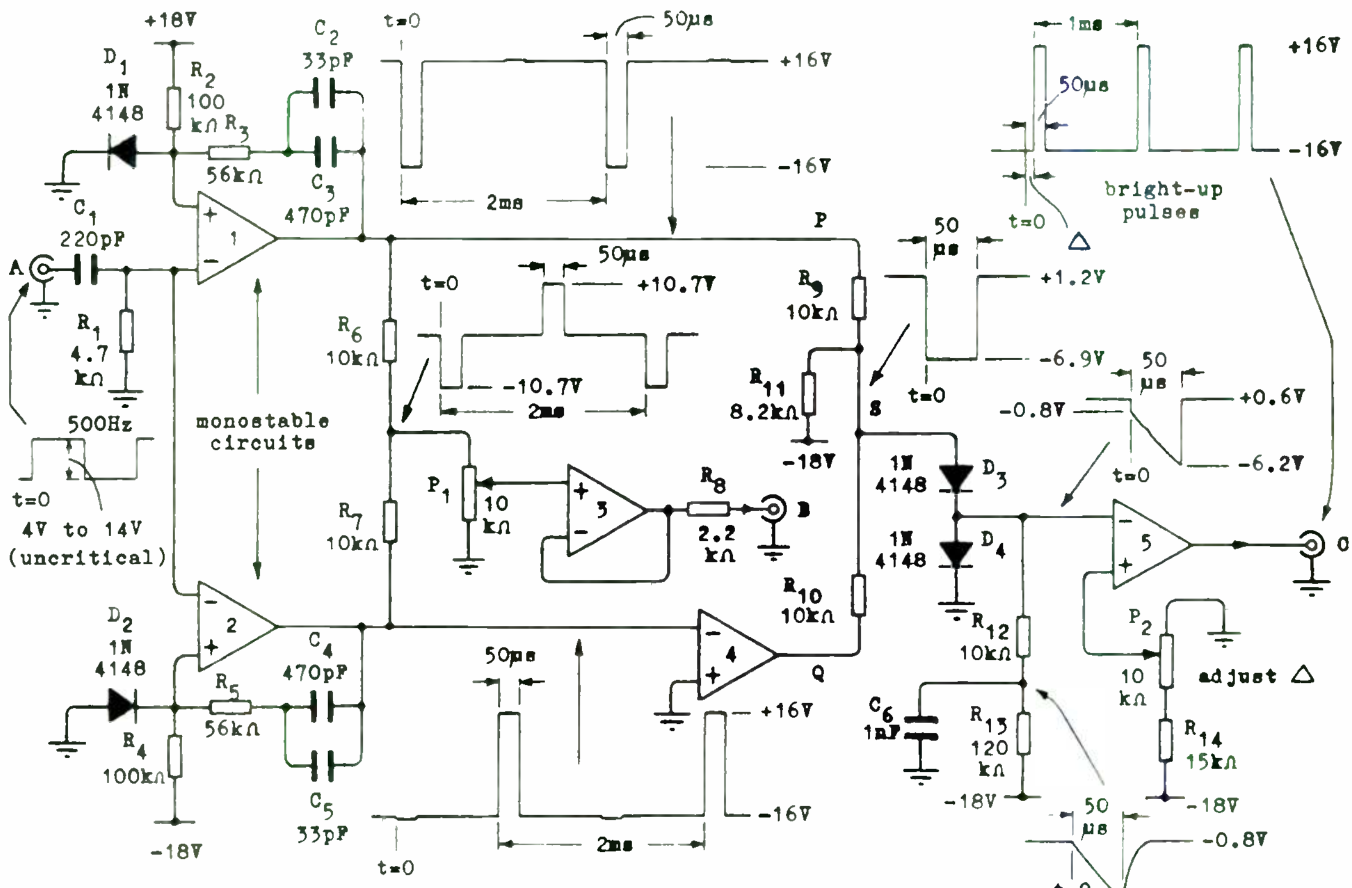
R11, however, shifts the mean level downward and gives some attenuation in amplitude. With D3 open-circuited, the pulse waveform at S would have an excursion from +3.1 to −6.8V, the effective source impedance being 3.1kΩ.

When D3 is present, the upper level of the pulse waveform at S is held at approximately +1.2V by conduction of D3 and D4, the inverting input of op amp 5 being about +0.6V. The output of this op amp is −16V, no matter what the setting of P2 may be.

Under these conditions, the top terminal of C6 is at about −0.8V, as determined by R12 and R13.

When a negative pulse occurs at S, then D3 and D4 cease to conduct, and the voltage at the inverting input of the op amp

Presented at the 82nd Convention of the Audio Engineering Society; London, March 1987. All figures courtesy of *JAES*.



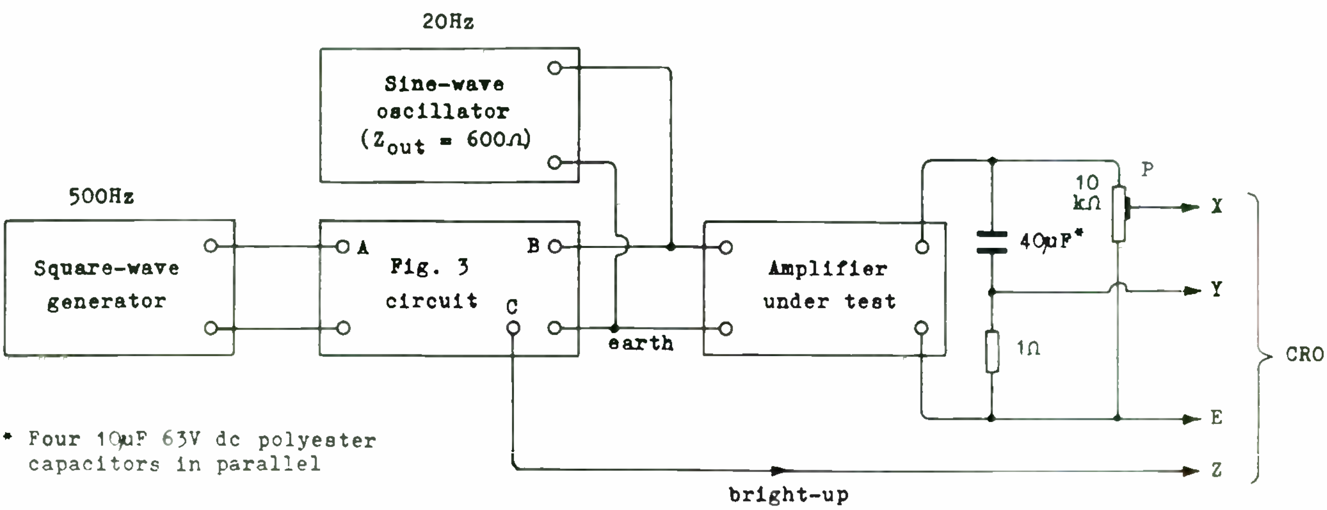
Op. Amps. 1-4 Type 084
Op. Amp. 5 Type 081

FIGURE 3: Waveform generating circuit.

becomes equal to that of C6, that is, approximately −0.8V. If the slider of P2 is set at the top of the track, this −0.8V on the inverting input is sufficient to produce a pulse excursion from about +16V down to −16V at the op amp output, virtually coincident in timing, therefore, with the beginning of the 50μsec pulse being fed to the amplifier under test.

If, now, the slider of P2 is assumed to be set well down the track, the initial excursion to −0.8V at the inverting input is not enough to cause the op amp output to swing over. However, C6 is meanwhile being charged negatively via R13, and the voltage at the inverting input of the op amp soon descends to the voltage level at the slider of P2 whereupon the op amp produces a positive-going output transition, delayed in timing by an amount D dependent on the setting of P2.

The supply voltages to the circuit have been made as high as the op amp ratings permit, to ensure comfortably adequate output amplitudes, particularly that for Z modulation. Lower supply voltages may be used if found suitable, though a reduc-



* Four 10μF 63V dc polyester capacitors in parallel

FIGURE 4: Complete setup, incorporating circuit of Fig. 3.

tion to ±9V gives about a 10% shortening of the pulse duration because the op amp slew rates are then less significant.

The two supplies should be kept well balanced for a predictable performance, because a 10% inequality in these voltages causes considerably more than 10% inequality in the durations of the positive and negative pulses.

Figure 5 shows some actual waveforms produced by the circuit of *Fig. 3*. The input square-wave frequency was temporarily increased from 500Hz to 2kHz to enable the pulses to be more clearly displayed. P2 was set to delay the start of the Z-modulation pulses by about 20μsec.

THE DISPLAY MECHANISM. The adoption of a lower value than 1Ω for the current-monitoring resistor has the advantage that certain small parts of the protection-circuit characteristic are then not omitted from the display. The explanation of this point is as follows:

Suppose that the low-frequency input to the amplifier under test has swung its output voltage out to point A in *Fig. 6*. We will ignore the small low-frequency current component taken by the 40μF capacitor, for present purposes. A 50μsec pulse then occurs, say of such polarity as to produce a positive-going change in the amplifier output voltage, if the amplifier is assumed, for the moment, to have a very rapid response, the working point jumps almost instantaneously to B, at which the current limiter operates. The slope of line AB corresponds to the 1Ω value of the current-monitoring resistor. After B has been reached, the output current continues to flow, charging the 40μF capacitor and causing the working point to move to C by the time the 50μsec pulse ends.

In practice, the rate of rise of amplifier output current is finite, so that the approach from A to the current-limiting condition is along a curve, the 40μF capacitor charging considerably during this approach. Thus current limiting commences at a point appreciably to the right of B. However if P2 (*Fig. 3*) has been set to delay the start of the bright-up pulse by an appropriate amount, this relatively slow transition from A to the current-limiting state will not be displayed and bright-up occurs at a point between B and C.

At C the 50μsec pulse ends, but the state of charge of the 40μF capacitor remains markedly different from what it would have been if the 50μsec pulse had not occurred. The result is that substantial reverse current flows back into the amplifier output for a short time. However, since blackout occurs at C, this reverse current flow is not displayed.

We are more concerned about what happens later, when the next 50μsec pulse arrives, this time producing a negative-going charge in the amplifier output voltage. The 20Hz input to the amplifier has meanwhile caused a slight change in output voltage, say to point D. Again assuming a very rapid amplifier response, the occurrence of the pulse shifts the working point to E, DE having a slope of 1Ω. Alteration in the state of charge of the 40μF capacitor shifts the working point to F by the end of the pulse.

With a sufficiently large positive output voltage due to the 20Hz input, the point G at the corner of the complete V-1 characteristic will be reached before the end of the 50μsec pulse has occurred. Output current then collapses, since to maintain it would require the capacitor voltage, and hence the amplifier output voltage, to continue rising at a rate given by dV/dt = I/C. Voltage clipping in the amplifier prevents this further rise. The fall of current along line GH is displayed, for blackout has not yet occurred.

Whereas the corner G of the V-I characteristic can be included in the display, the lower corner J cannot. The nearest approach is obtained by starting from H, which would enable point K to be displayed if the amplifier had a sufficiently rapid response. Since the slope of HK is 1Ω, clearly, under these conditions, reducing the value of the current-monitoring resistor to a much smaller figure would enable point J to be much more nearly included in the display. In practice, however, the finite slew rate of the amplifier would prevent this improvement from being fully realized.

There is a possible risk that some amplifiers may not retain adequate feedback stability when supplying a load consisting of a large capacitor in series with for example, only 0.1Ω. This favors adopting a higher value, such as 1Ω, for general use. A better compromise might be 0.5Ω.

CURRENT-MONITORING RESISTOR PURITY. In the earlier stages of this work, before I added the facility for delaying the start of the bright-up pulses, I found the importance of keeping the amount of stray inductance in series with the current-monitoring resistor adequately small, especially if I used resistor values of less than 1Ω. Such series inductance can result in the display overshooting the proper boundary of the V-I characteristic when a fast responding amplifier is being tested, though usually for less than 1μsec. Quite a small amount of bright-up delay prevents this effect from being seen, rendering the use of ordinary wire-wound resistors perfectly satisfactory, however.

CURRENT MONITORING RESISTOR RATING. The amount of power dissipated in a 1Ω current-monitoring resistor due to the low-frequency output component from the amplifier is quite small, provided the frequency is kept low enough. At 20Hz with an amplifier

Continued on page 14

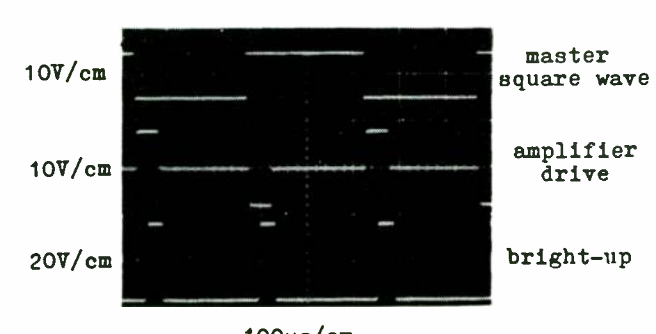


FIGURE 5: Waveforms produced by circuit of Fig. 3. Square-wave frequency was increased to 2kHz for greater clarity.

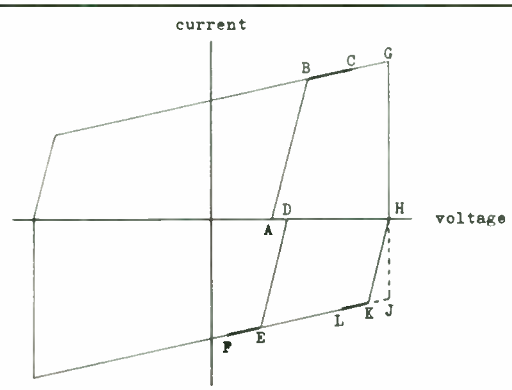


FIGURE 6: Display details.

EUROPEAN LOUDSPEAKERS OF AMERICA
COAXIAL DRIVERS

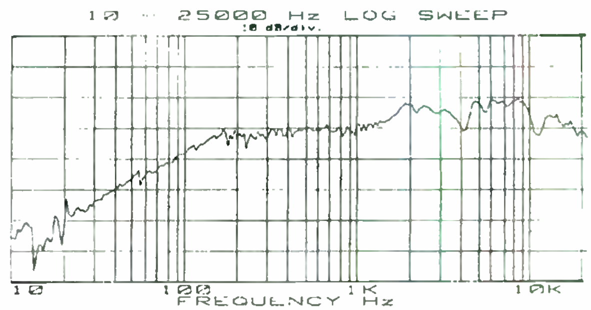
Europa

Specifications

4502/AUDAX $22.00

Fs 102.0 Hz
Vas 3.8 ltr
Rscc 3.7 ohms
VcL .18 Mh
Qms 7.77
Qes .46
Qts .43
Eff 90.5 db 1W/1M
Power 40.0 watts
Depth 2 1/16"
Cut out 4"

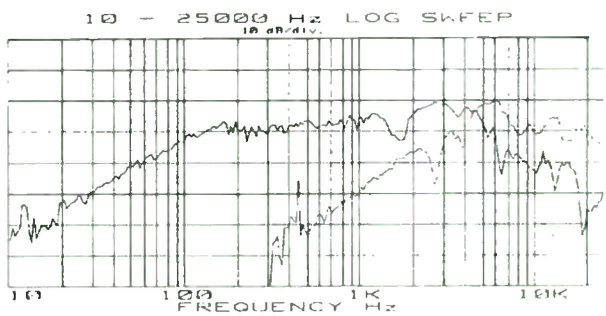
4502 /AUDAX Madisound Price $22.00



5402/AUDAX $23.00

Fs 91.0 Hz
Vas 5.42 ltr
Rscc 3.68 ohms
VcL .22 Mh
Qms 7.84
Qes .606
Qts .56
Bl 4.8 W/M
Eff 91.8 db 1W/1M
Power 45.0 Watts
Depth 1 15/16
Cut out 4 7/8"

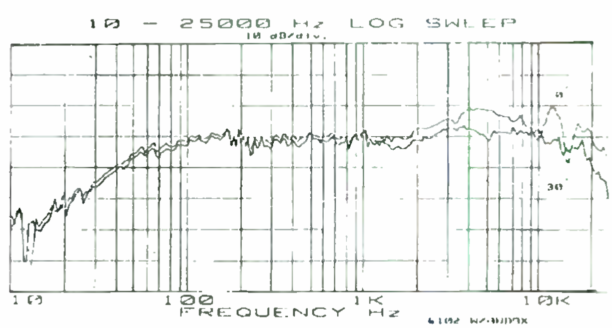
5402/AUDAX Madisound Price $23.00



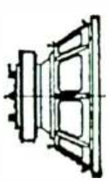
6102/AUDAX $24.00

Fs 59.0 Hz
Mmd 13.1 G
Cms .6x10-6
Vas 13.1 ltr
Rscc 3.6 ohms
Z Min 4.1 ohms
Z Max 40.2 ohms
VcL .25 Mh
Qms 5.6
Qes 5.5
Qts .50
Eff 92.5 db 1W/1M
Res. 50-20K
Power 50.0 Watts
Depth 2 13/16"
Cut out 5 7/8"

6102/AUDAX Madisound Price $24.00



Polyester 6db filters included. Ultimate steel snap on grills with mounting rings are available for these drivers at $4/$5/$7 per grill.



MADISOUND SPEAKER COMPONENTS
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MADISON, WISCONSIN 53711
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Fast Reply #JC112

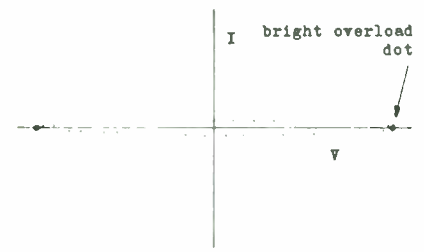


FIGURE 7: Display obtained during set-up adjustments.

operating on ±50V supplies, the dissipation will be less than 0.1W, even though the 20Hz input voltage is made large enough to give voltage clipping and hence a nonsinusoidal 20Hz output waveform. However, take care during set-up adjustments not to let the sine-wave oscillator operate inadvertently at a high frequency, since this could easily burn out the 1Ω resistor.

Assuming, by way of example, that the amplifier under test has a constant-current limiter characteristic limiting at 10A, the power dissipated in a 1Ω current-monitoring resistor during a 50μsec pulse is 100W. If the square-wave frequency is 500Hz, there are 1,000 pulses per second, that is, one 50μsec pulse every millisecond. The mean power dissipated in the 1Ω resistor is then 5W. A resistor, or series-parallel combination of resistors, for example, rated at 10W, should therefore be comfortably adequate in most circumstances, permitting peak currents up to at least 14A.

STANDARDIZED DISPLAY SCALINGS. To facilitate easy comparison of the V-I limiter displays for various amplifiers, I found it convenient to adopt standardized scalings for current and voltage, and suggest that an appropriate choice will normally be 5A and 20V, respectively, per division.

This also aids the comparison of amplifier and loudspeaker V-I displays, as I describe later.

OPERATING PROCEDURE AND PHOTOGRAPHY. To set up for the standardized X scaling of 20V/div, apply only the sine-wave input to the amplifier under test, at 20Hz or other fairly low frequency. (Beware of making the frequency more than about 200Hz, as this may burn out the current-monitoring resistor or cause excessive dissipation in the amplifier output transistors.) Put a voltmeter across the amplifier output and adjust the input for a reading of 28.3V RMS. Then adjust P (*Fig. 4*) and/or the CRO X gain for ±2 div deflection.

Set the CRO Y sensitivity to 5V/div to obtain the standardized Y scaling of 5A div. Assume a 1Ω current-monitoring resistor.

With zero signal input, carefully set the spot to the center of the graticule. Now apply 20Hz or other low-frequency, sine-wave input at comfortably sufficient level to make the amplifier overload, that is, give voltage clipping. The potentiometer P1 in *Fig. 3* must be at zero, so no pulse input goes to the amplifier. Set the CRO brilliance control for a not-too-bright display, as shown in *Fig. 7*.

In general, the brightened-up small dots will be seen to be moving around the ellipse, unless the 20Hz and 500Hz are in precise integral relationship. For good photographs, set the frequencies so the dots drift around quite slowly and then give a fairly long exposure, such as 5sec, for the subsequent V-I display photographs.

Now turn P1 (*Fig. 3*) up to maximum, or at least up to a sufficiently high setting to produce hard current limiting as indicated by the display. You should probably keep P1 turned up for no more than about 10sec, though most amplifiers would actually withstand continuous stressing in this manner.

Adjust the bright-up delay control P2 for the best picture and make the photographic exposure.

8Ω AND 4Ω LINES. Lines representing 8Ω and 4Ω resistive loads may be introduced into the photographed displays if desired. This may be done by proceeding as follows, to make a separate exposure:

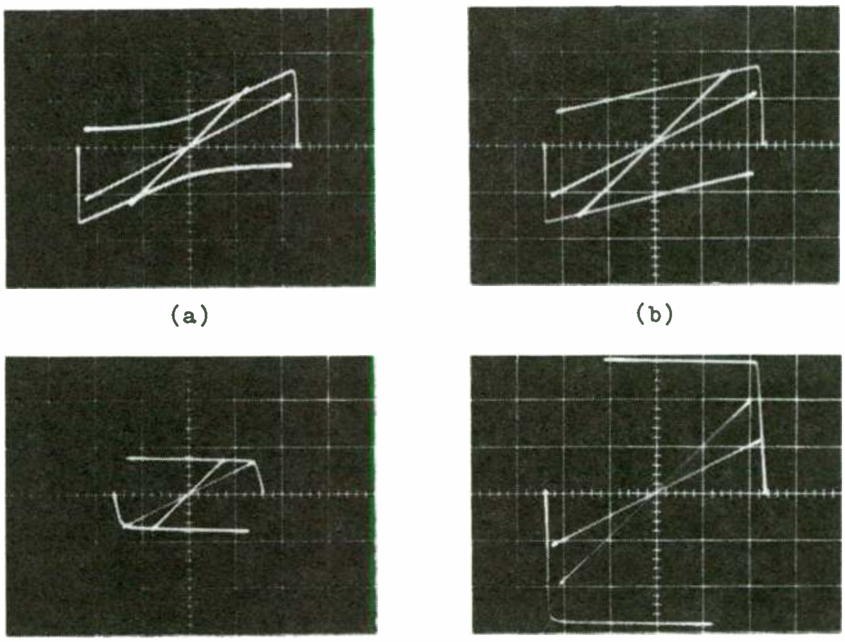
Set P1 to zero (no pulse input) and disconnect the CRO Z modulation.

Assuming the current-monitoring resistor is 1Ω, replace the 40μF capacitor with an accurate 7Ω resistor of adequate power rating, for the 8Ω line, or a 3Ω resistor for the 4Ω line, and turn up the sine-wave input for just long enough to take a photograph with, say a 1sec exposure. (The 1Ω resistor will probably be dissipating much more than its normal rating during this burst of output).

With the standardized X and Y sensitivities, the 8Ω line should be at 1 div up for 2 div across slope, the 4Ω line being at 45° slope.

CURRENT-MONITORING TRANSFORMER. You may see advantages, in certain respects, by inserting a step-up transformer between the circuit whose current is to be monitored and the monitoring resistor itself.

A 1Ω monitoring resistor used straightforwardly gives a sensitivity of 1V/A, inserts 1Ω in the monitored circuit, and dissipates 100W for a current of 10A. Using, for example, a 1:1000 transformer with a 1kΩ resistor across the secondary, this same sensitivity of 1V/A is obtained, but the effective resistance inserted in the monitored circuit is now, ideally, only 1mΩ, and a current of 10A now dissipates only 10mW in the resistor.



(a) (b) (c) (d)

FIGURE 8: V-I displays, including 8Ω and 4Ω load lines, for four amplifiers. (a) Early Quad 405; (b) Later Quad 405; (c) Quad 303; (d) Preproduction Quad 520. Scale—vertical 5A/div; horizontal, 20V/div.

The use of a transformer would therefore avoid the dangers, of accidentally burning out the current-monitoring resistor.

The point concerning amplifier stability, mentioned previously, should be borne in mind when a transformer is employed, and this may necessitate the addition of a small amount of resistance in series with the primary when obtaining the V-I limiter displays of some amplifiers. The added resistor's value need not be accurately known, since it does not affect the display except in the minor respect explained previously.

The added resistor would not be required when displaying 8Ω or 4Ω lines, and using a transformer avoids the need for employing load resistor values of less than 8Ω and 4Ω.

The negligible effective series resistance introduced when a transformer is employed is also advantageous when determining the V-I displays for loudspeakers.

Some advantages of a transformer may be obtained merely by employing a very low value of current-monitoring resistor with appropriately increased CRO Y gain. It becomes increasingly difficult, however, as the resistance value is reduced, to avoid significant effects due to series inductance, and the connection of several resistors in parallel with very short leads is desirable. The measurement of the resistance value with good accuracy is also more difficult at very low values, and both these difficulties are avoided when a transformer is employed.

I used a transformer, in fact, for most of the limiter-characteristic, load-line, and loudspeaker displays shown in this article; I included a 1Ω resistor in series with the primary for the limiter displays only, in the interests of good stability.

The transformer has a 1000-turn secondary of 0.12mm diameter enameled wire (40 SWG, 36 AWG) wound in a single section on a core of 0.38mm (0.015") Mumetal (Permalloy C) laminations of maximum dimension 25.4mm (1") and stack 12.7mm (0.5"); center-limb width 6.3mm (0.25"), no. 187 laminations. The primary is a single turn around the outside of the secondary.

A 100Ω ±1% metal film 0.125W resistor is connected across the secondary, giving a sensitivity of 100mV/A. I also added a 2.2nF shunt capacitor to suppress a slight tendency to ring at about 1MHz, involving distributed winding capacitance and leakage inductance. The 100mV/A sensitivity is less than that given in the previous example, but the use of a 100Ω rather than 1kΩ secondary resistor gives a much better very low-frequency response, which is less than 3dB down at 1Hz, the phase error being less than 2° at 20Hz. This phase error is sufficient, however, to give a noticeable opening out of the 8Ω and/or 4Ω lines if the oscillator frequency is made as low as 20Hz; about 70Hz was therefore used.

My transformer had actually been made for other measurement purposes, and was therefore conveniently available. Had this not been the case, I must confess that straightforward current-monitoring resistors probably would have been used for all the measurements.

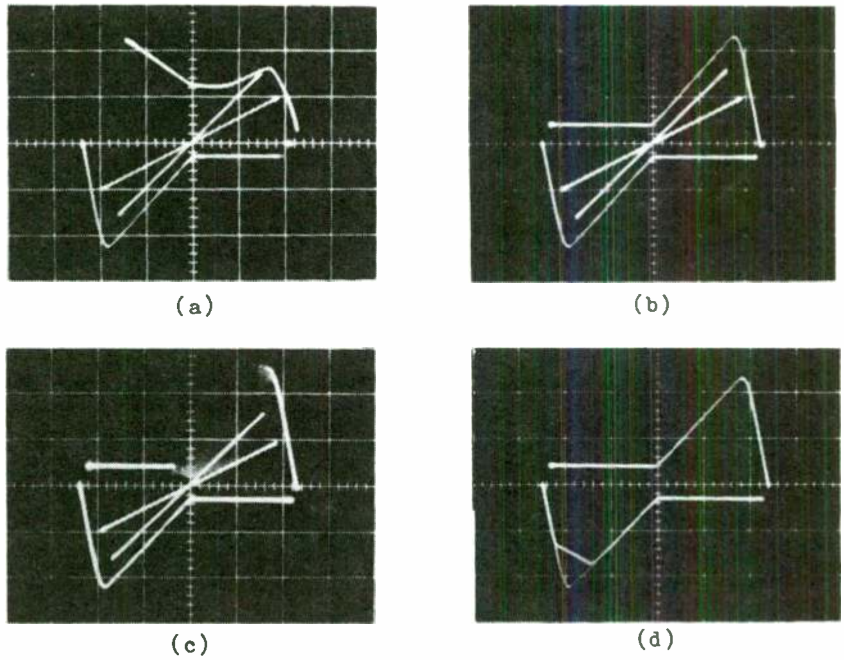


FIGURE 9: V-I displays, including 8Ω and 4Ω load lines, exhibiting faulty behavior features. Scale—vertical 5A/div; horizontal, 20V/div.

DISPLAY EXAMPLES. The display shown in *Fig. 8a*, for an early version of the Quad 405 amplifier, is the same as that in *Fig. 2c*, but with the addition of 8Ω and 4Ω load lines. Note that the excursion with the 4Ω load goes outside the limiter characteristic. This means that the unregulated positive and negative DC supply voltages fall considerably when the amplifier is feeding a 4Ω load at a high signal level. The nature of the voltage-dependent current-limiter circuit is such that a fall in supply voltage increases the current value, for a given instantaneous amplifier output voltage, at which current limitation occurs. A decrease in line voltage also increases the peak current that can be turned on into a 4Ω load.

With an 8Ω load, on the other hand, overload occurs because of voltage clipping rather than current limitations. A fall in DC supply voltage then causes a reduction in the achievable peak output level.

Later versions of the Quad 405, with serial numbers above 29 000, had resistors R27 and R29 in the limiter circuit increased from 8.2 to 15kΩ and *Fig. 8b* shows the display obtained with this modification incorporated.

Figure 8c is for a Quad 303 amplifier, which has a stabilized DC supply and a simple form of nonvoltage-dependent current limiter. This amplifier has a smaller output rating than the 405, which is instantly conveyed by the smaller size of the display, emphasizing the advantage of adopting a standardized scaling for the photographs.

In contrast, *Fig. 8d* relates to a preproduction version of the Quad 520 rack-mounted professional stereo amplifier, which has paralleled transistors of high rating in the output stage.

Figure 9 illustrates some defective performance features, obtained with a rack-mounted amplifier made by another British firm. *Figure 9a* was photographed with the amplifier in its original state, as supplied. Investigating the circuit I found that the current-limiter clamp transistor, on the positive side of the circuit, was called upon to clamp an indefinitely large current turned on by the previous transistor, this current being restricted only by the current-gain factor of the transistor. It was thus a matter of which transistor would win in a rather brutal contest. Attempts had evidently been made to rescue the situation, including fitting a sizable heatsink to the amplifying transistor and a 470Ω resistor in

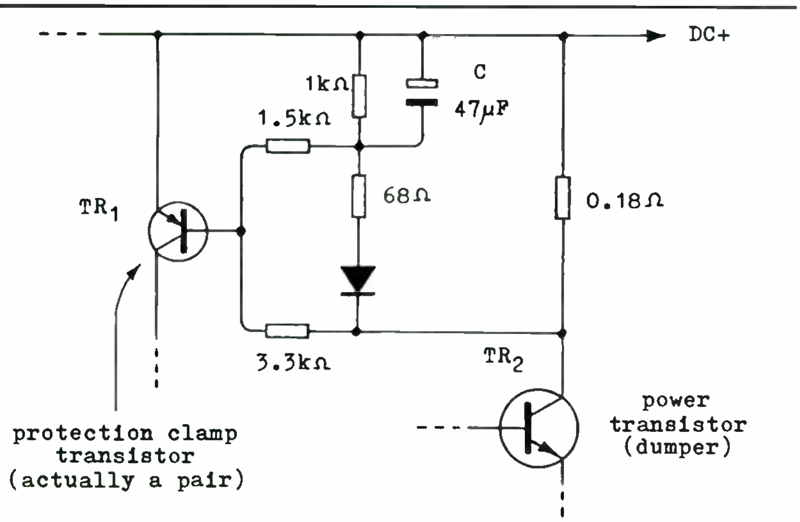


FIGURE 10: Simplified circuit illustrating principle of Quad 405/2 time-dependent protection scheme.

series with its output—the latter, however, being singularly ineffective under the conditions applying in the top left quadrant.

By inserting a 10Ω resistor in the emitter lead of the p-n-p stage whose output is clamped, and a series pair of small silicon diodes to limit the base voltage applied to this transistor, the clamping operation was made predictable and relatively gentle, giving the display shown in *Fig. 9b*. Unenlightened design features such as these are only too prevalent in much equipment whose appearance and advertising imply the highest standards.

The display in *Fig. 9b* is, no doubt, as intended by the designer, and it shows that the amplifier comfortably meets its specification of 70W into 8Ω and 100W into 4Ω, these being mean-power figures for continuous sine-wave operation with a resistive load. However, the voltage dependency of the current limiter has been made so drastic that the full output voltage level cannot be obtained without serious distortion, even under sine-wave conditions, when there is a substantial amount of reactance in series with the load resistance, particularly when the latter is well under 8Ω.

Analysis shows that if a straight line is drawn from the right of the 4Ω load line, for example, down to intersect the X axis at an equal negative output voltage, and if any part of this line goes outside the limiter characteristic, then a sine-wave output voltage equal to that represented by the load line will not be obtainable without distortion when certain values of series reactance are present. Conversely if this line lies everywhere below the limiter characteristic, the full output level will be obtainable with negligible distortion for any value of series reactance. Similar considerations apply, of course, to the lower half of the display[1] (see also *Fig. 17*).

Figure 9c shows that RF oscillation occurs when the limiter operates, if a 47nF capacitor in the positive side of the limiter circuit is removed, thus relieving my curiosity as to why this capacitor had been included. In another amplifier, of European origin, however, oscillation of a similar kind occurred with the amplifier in an unmodified state.

For *Fig. 9d* the value of the bootstrap capacitor, joining the amplifier output to the junction of the two resistors that constitute the collector load of the predriver p-n-p common-emitter amplifying stage, was reduced from 47μF to approximately 8μF. The low-frequency input was at 20Hz, as for the other oscillograms in *Fig. 9*. Near the lower left of the display, the slanting line just above the tip itself is traversed with the spot moving left-to-right, the previous right-to-left spot movement being via the actual tip. What happens is that the voltage across the bootstrap capacitor, and hence the voltage across the upper of the two collector load resistors mentioned above, falls off while the amplifier output voltage is near its maximum negative excursion, ultimately becoming insufficient to provide the required base current to the lower driver transistor. The output transistor is then unable to turn on the full output current desired.

Increasing the low-frequency input from 20 to 50Hz or above eliminates this effect from the display. With some amplifiers the effect may be observed, at least mildly, at 20Hz, even without reducing the value of the bootstrap capacitors fitted.

Notice that the right and left boundaries of the displays in *Fig. 9* are much less steep than most of the previous ones. This is because the amplifier in question has 0.5Ω emitter resistors in the complementary output stage, while the Quad amplifiers (*Figs. 2* and *8*), except for the 303, have much lower resistor values.

PROTECTION CIRCUITS WITH TIME-DEPENDENT BEHAVIOR. Some protection circuits, though not those already considered, incorporate arrangements whereby the current limitation becomes more severe than normal if the operating point remains for more than a short time in "hazardous parts" of the V-I display area, or if the excursion to these places, though of short duration are very frequent.

The Quad 405/2 amplifier is fitted with protection circuit modules having the features just mentioned.[2]

The essence of the protection circuit on the positive side of the amplifier is conveyed by the highly simplified diagram of *Fig. 10*. The device shown as a simple diode is actually a two-transistor circuit, rendered functional only when the voltage across TR2 exceeds about 45V. Thus for the conditions existing in the top left display quadrant, the circuit shown is the relevant one.

For large short-duration current pulses occurring in TR2 only occasionally, C remains virtually uncharged, and the protective clamp transistor TR1 is not brought on until about a 1.5V drop occurs across the 0.18Ω resistor, corresponding to about 8A. The time constant of 68Ω and 47μF is 3.2msec, so that for isolated current pulses approaching this duration or longer, the capacitor is charged to a significant voltage. When this has occurred, a smaller voltage drop across the 0.18Ω resistor is sufficient to bring on the clamp transistor. Therefore for an isolated current pulse of long duration, the maximum current the amplifier can turn on is more severely limited during the later parts of the pulse than at the beginning.

Though, as already mentioned, an isolated short pulse does not significantly charge C, a rapid succession of short pulses will do so, again causing the maximum current that can be turned on to be reduced.

1. Baxandall, P.J., "High-Fidelity Amplifiers," *Radio, TV and Audio Technical Reference Book*, S.W. Amos, Ed., Newnes-Butterworth, London, 1977, Chapter 14.

2. In some Quad diagrams, the left transistor, of the four in each module, is inadvertently shown as p-n-p, whereas it is actually n-p-n.

This more subtle type of protection circuit considers the transistor manufacturer's V and I rating limitations as expressed in the form of SOAR curves and as extended beyond these to allow for frequency operation with current pulses of large magnitude[3,4] The circuit does not, however, introduce the type of instantaneous negative output resistance inherent in more normal types of protection circuits, the latter sometimes producing rather unpleasant sounds, with some types of load, if overdriven. Only the minimum necessary protection is provided when short-duration musical transients occur.

Figure 11a was obtained with a Quad 405/2, with the low-frequency sine-wave input at 70Hz rather than 20Hz to avoid a small amount of the effect described previously in relation to *Fig. 9d.* The pulse repetition frequency of the pulses was 100Hz for the dimmer display and 2kHz for the brighter one—the exposure times are approximately equal.

Figure 11b was obtained with the 47μF capacitor in the negative side protection circuit removed, the pulse repetition frequency being 500Hz. This demonstrates that the effective diode of *Fig. 10*, but in the negative-side protection circuit, becomes functional only when the amplifier output voltage is more positive than about −8V.

The pulse duration for the *Fig. 11* tests was 50μsec, as with the other displays. You may use much longer pulses, but to avoid excessive mean power dissipation, the pulse repetition frequency should be greatly reduced. Making the pulse repetition frequency nearly equal to that of the low-frequency sine wave is a satisfactory technique, so that the pulses are phased with respect to the X deflection in a slowly varying manner, gradually tracing out the relevant limiter characteristic.

You may observe that the limiting values of output current, as displayed, gradually change over a period of minutes, normally reducing in magnitude, after switching on the amplifier or the test signal. This effect, usually fairly small, involves warming of the protection circuit transistors by heat conducted from hot components in the amplifier, often mainly along circuit board conductors.

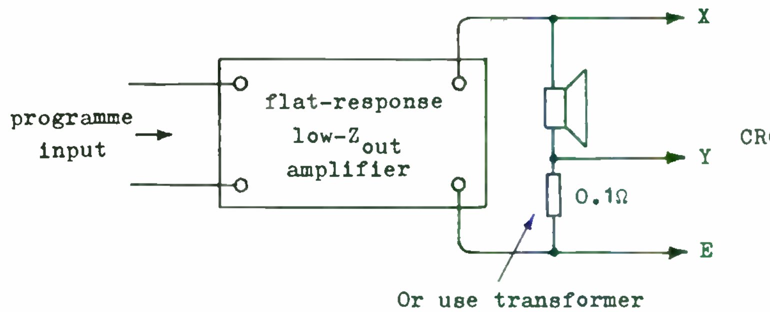


FIGURE 12: Setup for loudspeaker tests.

AMPLIFIER AND LOUDSPEAKER V-I DISPLAYS. For your greatest benefit, the V-I display for an amplifier should somehow be related to the characteristics of the loudspeaker to be driven, and it should here be borne in mind that loudspeaker data obtained on a sine wave basis are liable to be only approximately relevant to the practical problem of feeding the loudspeaker with music waveforms.

On light load, an amplifier normally clips at a fairly closely defined peak instantaneous output voltage, and usually you will want to know whether the amplifier can feed a particular loudspeaker with a wide variety of program voltage waveforms peaking up to about this same instantaneous clipping voltage without running out of current-turning-on capability. Or, if the amplifier cannot produce this full instantaneous voltage level, then you would need to know up to what level it may be allowed to go before current limitation sets in.

The answers to such questions may be obtained by carrying out tests on the loudspeaker using the setup of *Fig. 12.*

To make the results easy to compare with the amplifier displays described earlier, you can conveniently set up the system so that a 45° line is produced on the CRO if the loudspeaker is replaced by a 4Ω resistor.

On the assumption that the loudspeaker may be regarded as a linear device, testing at a high volume level is not necessary—the X and Y gains may be increased appropriately so that displays of similar size to the amplifier displays may nevertheless still be obtained.

A variety of loud music passages should be used for this test, and they should be chosen to contain both high peak voltages and high peak rates of voltage change. In other words, there should be plenty of high-frequency as well as low-frequency content. An indication of the degree to which a given music excerpt is suitable may be obtained using my special, full-wave peak program meter design, featuring:

- Charging time constant about one hundredth of that for a normal peak program meter;
- Every time a program transient occurs that is of higher peak instantaneous magnitude than that already being indicated on the meter, the needle jumps up to indicate this new level and remains stationary at the new reading for 1sec before starting to fall back, thus facilitating easy and accurate reading of the program peak. If an even higher program peak happens to occur during this 1sec interval, the needle jumps up further to indicate this new peak level, and a new 1sec "dwell time" is initiated;

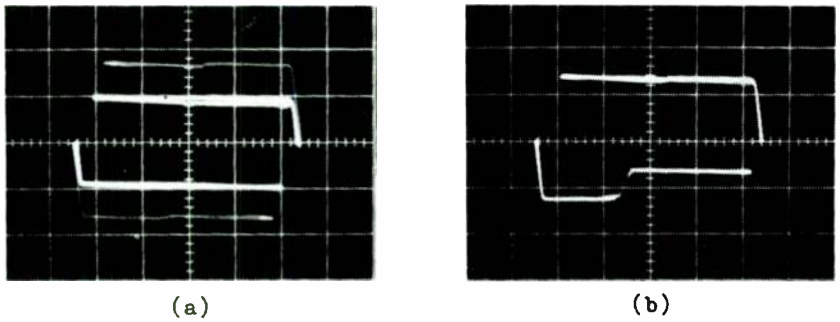


FIGURE 11: V-I displays for the Quad 405/2 amp. (a) 50μsec pulses at 100Hz (dim) and 2kHz (bright); (b) 50μsec pulses at 500Hz, with 47μF in negative-side protection circuit removed.

3. Ballard, M.F. and T.W. Gates, "Safe Operating Area for Power Transistors," *Mullard Tech. Commun.*, April 1974, Vol.13, pp. 42-65.
4. Noble, P.G., "The Safe Operation of Power Transistors," *Mullard Tech. Commun.*, July 1978, Vol. 14, pp. 346-375.

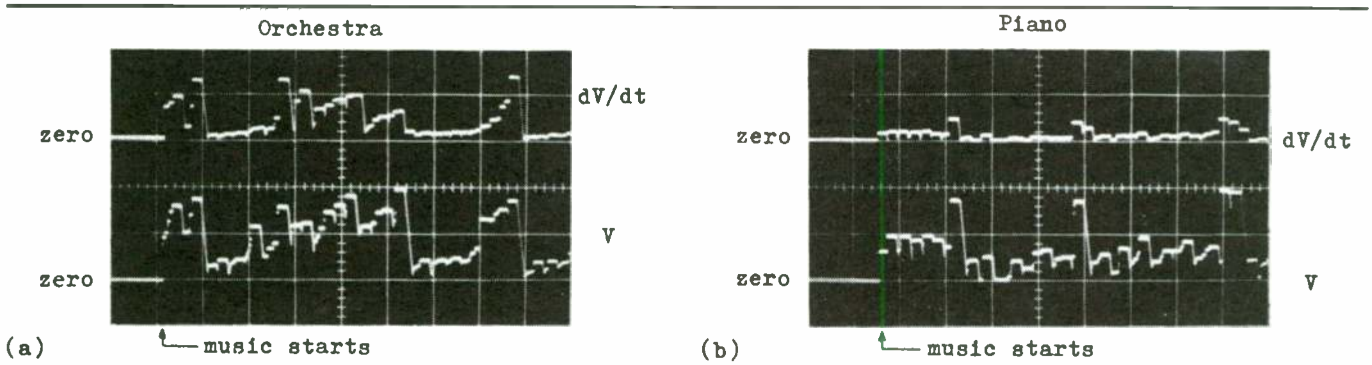


FIGURE 13: Results using fast-response linear peak program meter with 1sec dwell time. (a) Orchestra; (b) Piano. Scale—vertical 2V/div; horizontal, 5sec/div.

- The circuit may be switched to read either the peak values of the program voltage V or the peak values of dV/dt. I made the relative sensitivities so that equal readings are given in the two switch positions for a sine-wave input at 5kHz;
- An output voltage proportional to meter readings is made available for operating a chart recorder or an oscilloscope.

The instrument as it now exists is linear rather than logarithmic, though it can be modified for logarithmic operation.

Figure 13 shows the results obtained using this meter on two music excerpts, each lasting about 40sec. *Figure 13a* displays the beginning of Johann Strauss's *Radetsky March* (Deutsche Grammophon CD no. 410 027-2), and involves loud and vigorous playing by the Berlin Philharmonic, with cymbal clashes, which give the high values of dV/dt shown.

Figure 13b describes the beginning of track 6 (Denon CD no. 38C37-7043), *Debussy Preludes from Book 3*, played by Jacques Rouvier on a Steinway. This is loud and dynamic piano playing, but you can see that the ratio dV/dt to V is fairly small compared to the orchestral display.

Other digitally recorded material giving a high ratio of dV/dt to V includes applause, a brass group, a bell, and fast drum rim shots. The ratio in each case may be expressed in terms of f_o, the frequency of a sine wave that has the same ratio of peak values of these quantities.[5] For items such as the last two, f_o values as high as 10kHz can occasionally be obtained, f_o for *Fig. 13a* being about 3.5kHz if derived on the basis of the ratio of the highest dV/dt value during the excerpt to the highest V value.

Music with cymbal clashes, applause, and so on, evidently is likely to lead to problems with loudspeakers whose impedance drops to low values at high frequencies.

The displays of *Fig. 14* were obtained with three different loudspeakers, at a fairly low signal level, using the *Fig. 12* setup. A suitable 3sec orchestral excerpt from *Fig. 13a* was chosen. The gain was set to obtain a maximum V value on the peak program meter corresponding to 2-div horizontal spot deflection.

Because the signal, even during loud passages, actually spends most of its time at relatively small instantaneous voltage values, the photographs show a halation problem, and to obtain a clear, permanent record of the peak spot excursions is difficult. You can see more detail directly on-screen, especially in a darkened room with a high CRO brilliance setting.

I attempted to improve the result by adding a circuit to brighten the trace during large amplitude or high-velocity spot movements, but this was only partially successful.

Ideally, we want a circuit arrangement that leaves a permanent record on the CRT of the most extreme spot displacements as they are reached during a given musical passage—a bright and clear boundary line that keeps being pushed outward every time a larger spot excursion occurs.

A circuit for doing approximately this could be made, but would be fairly elaborate and expensive. For example, a number of gate circuits might be utilized, all fed in parallel from a signal voltage representing the loudspeaker current, the output of each gate feeding a peak rectifier with a very long decay time. Each gate would open only when the instantaneous loudspeaker voltage is within a certain small voltage interval. The magnitudes of the outputs of the various peak rectifiers would be sampled and displayed as Y deflections at horizontal positions representing the relevant amplifier output voltages. Negative-responding as well as positive-responding peak rectifiers would be required. A display

5. Baxandall, P.J., "Audio Power Amplifier Design—I," *Wireless World*, January 1978, Vol. 84, pp. 53-57.

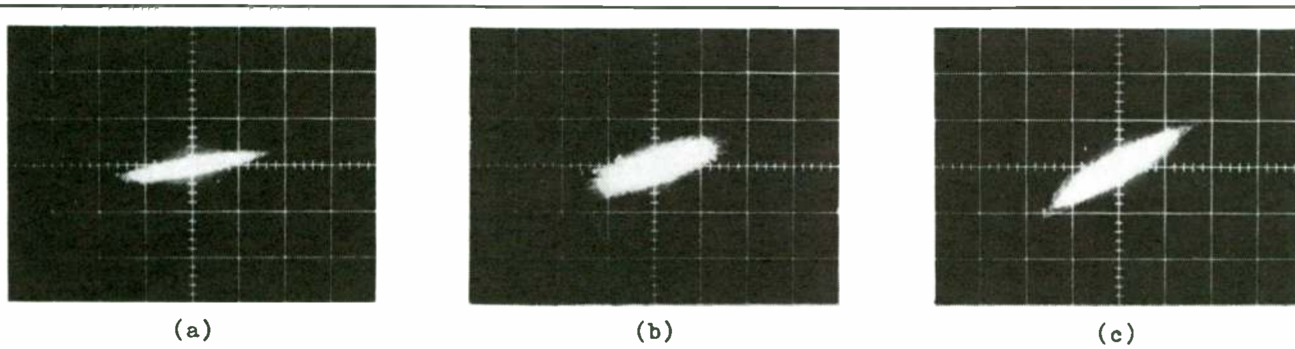


FIGURE 14: Loudspeaker V-I displays on loud orchestral music with cymbals. (a) Rogers/BBC LS3/6, 15Ω nominal; (b) KEF Corelli, 8Ω nominal; (c) Quad ESL63, 8Ω nominal. Scale—vertical 5A/div; horizontal, 20V/div.

would thus be built up on a stepped basis, the steps being small and close together if a sufficiently large number of peak rectifiers, and so on, were used.

For a much easier technique for obtaining V-I displays for loudspeakers, use the setup of *Fig. 12*, but with a swept sine-wave input.

A low brilliance setting should be used, the camera shutter being open throughout the sweep, which should be logarithmic, occupying at least several seconds. A logarithmic sweep avoids an excessive sweep rate at low frequencies, allowing virtually steady-state current values to be established, and it gives equal photographic exposure for each octave, which is appropriate.

The oscillograms of *Fig. 15* were obtained in this manner for the same loudspeakers used for *Fig. 14*. Comparison of *Figs. 14* and *15* does not support the notion that the peak currents demanded by loudspeakers fed with program input are liable to exceed those for sine-wave voltage input of the same peak magnitude.[6,7] On the contrary, the sine-wave sweeps seem to give larger peak current values.

A weakness of the sweep technique, in one sense, is the height of the display obtained is enhanced just as much by low loudspeaker impedance at 20kHz as it is by a low impedance at frequencies which are more significant from a program point of view. This point shows up particularly in *Fig. 15c* for the Quad ESL63, whose impedance modulus at 15kHz is only about 3.5Ω. I obtained *Fig. 15d* by limiting the sweep to 40Hz-5kHz, instead of 20Hz-20kHz; clearly more relevant to likely current demands under program conditions.

On the other hand, certain artificial test-signal waveforms may be produced that will cause a loudspeaker to draw larger peak currents than with a sine-wave input of the same peak value. This effect is not confined to devices such as loudspeakers, where a motional electromotive force (emf) is involved, but is a property of many passive networks containing reactive elements. The simplest example is that of *Fig. 16*, where you can see that increasing the magnitude of the load impedance by adding a series capacitor doubles the peak current.

Another test on the KEF Corelli loudspeaker showed that whereas with constant-voltage sine-wave drive it took maximum current at about 8kHz, the maximum current with square-wave drive occurred at a frequency of about 4kHz, and peak instantaneous value was approximately 35% greater, the drive voltages being the same peak value in both cases.

However, no music waveform observed has approximated, even roughly, a square wave of full amplitude; waveforms of very large peak amplitude tend to be of a spiky nature.

A combination of a hefty low-frequency component with a short-duration impulsive spike superimposed on it can, with suitable phasing of the two, give an unusually large peak current for a given total peak voltage. But the probability is that, at other times in the music, such components will be differently phased, in such a manner to give a larger peak voltage and hence voltage clipping. The

6. Martikainen, I., A. Varla, and M. Otala, "Input Current Requirements of High-Quality Loudspeaker Systems," presented at the 73rd Convention of the Audio Engineering Society, *JAES* (Abstracts), Vol. 31, May 1983, Preprint No. 1987, p. 364.

7. Otala, M., and P. Huttunen, "Peak Current Requirement of Commercial Loudspeaker Systems," *JAES*, Vol. 35, June 1987, pp.455-462.

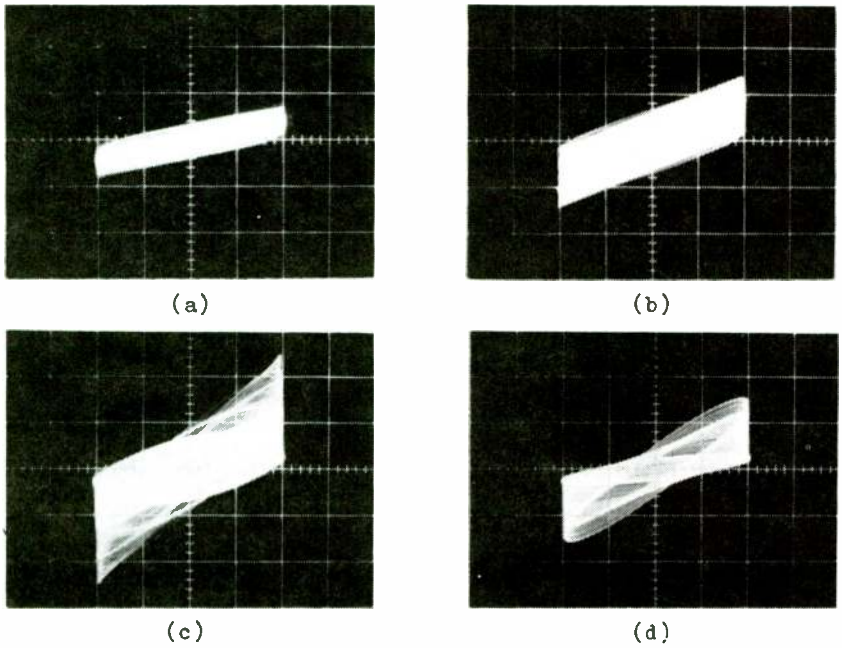


FIGURE 15: Frequency sweeps on three loudspeakers. (a) Rogers/BBC LS3/6, 15Ω nominal, 20Hz–20kHz; (b) KEF Corelli, 8Ω nominal, 20Hz– 20kHz; (c) Quad ESL63, 8Ω nominal, 20Hz–20kHz; (d) Quad ESL63, 8Ω nominal, 40Hz–5kHz. Scale—vertical 5A/div; horizontal, 20V/div.

program level then needs to be turned down to avoid overloading, and the previously excessive current demands now no longer arise.

Thus although the topic could obviously be further investigated at length, I am inclined to think that if an amplifier can provide peak currents in accordance with the requirements indicated by sweep tests of this (*Fig. 15*) type, it will also cope adequately with any normal program material having the same peak instantaneous voltage.

An alternative approach, which is attractive because it is so easy to apply, is based on a notion mentioned earlier, illustrated in *Fig. 17.* No matter how complex the equivalent circuit representing the electrical impedance of a loudspeaker may be, this impedance is always purely resistive at, or very near to, the frequencies where it dips down to minimum values. At frequencies in the same broad region, either side of the minimum, the impedance remains fairly low but now has significant reactance in series with the same resistance value. In such regions at least, the impedance of a loudspeaker thus has the same nature as that of the simple R and X combination shown in *Fig. 17a*, in which R remains constant and X varies with frequency. The V-I display for such a combination, with constant sine-wave output voltage of varying frequency, is therefore a series of ellipses, and it may be shown that these all fall within, and tangential to, the broken-line parallelogram of *Fig. 17b.*

Hence if the minimum impedance of a loudspeaker is known, as it usually is from the published impedance-modulus curve, a line representing this (resistive) impedance may be drawn on the V-I plot, a standardized scaling being adopted again, such that a 45° slope would represent 4Ω. The broken-line figure is then completed, starting at a point conveniently representing the peak voltage swing that the amplifier to be used is expected to be able to produce.

The actual V-I display for a loudspeaker determined as for *Fig. 15*, will not normally occupy quite the full space inside the parallelogram obtained as in the previous paragraph, for full occupancy would require the series reactance to vary with frequency from zero up to an infinitely large value.

Thus the parallelogram represents a worst case, so to speak, and if it fits within the protection-circuit display for a specific amplifier, when drawn to the same width, then that amplifier should be able to comfortably deliver its full output voltage to the loudspeaker without significant distortion. If the parallelogram will not fit within the amplifier display on this basis, then it should be scaled down until it just will. Because of its simple shape, this is easily done. The reduced output level likely to be obtainable without significant distortion can then be seen.

Using the minimum (Z) values for the

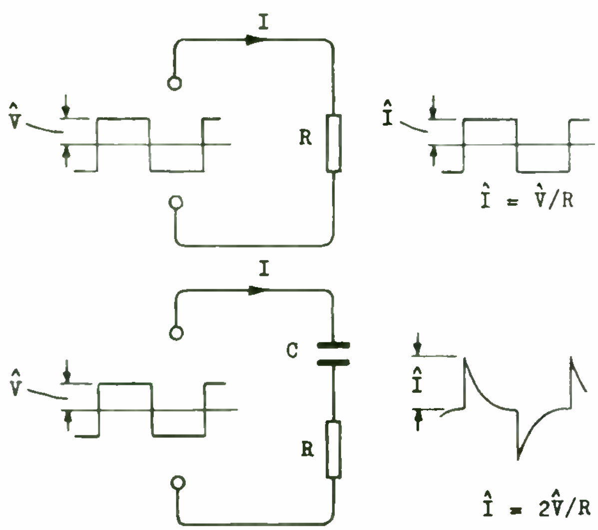


FIGURE 16: Adding C increases the impedance but doubles the peak current.

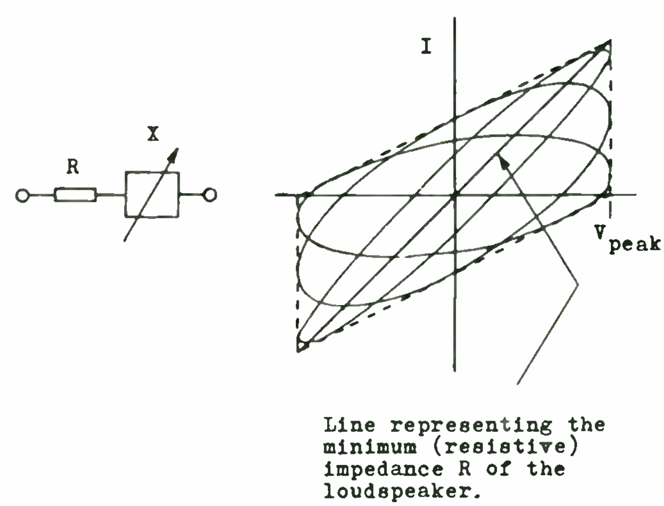


FIGURE 17: Load ellipses and tangential parallelogram.

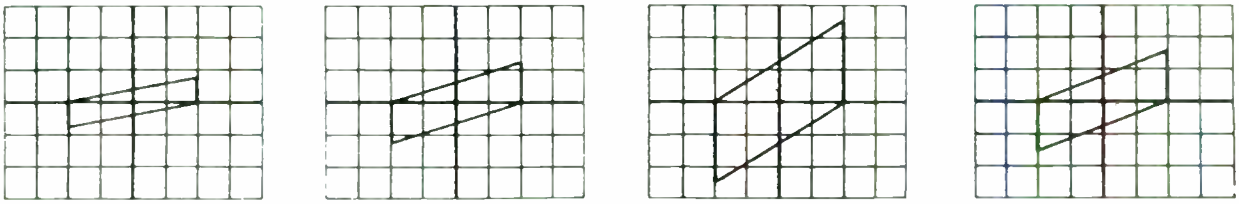


FIGURE 18: V-I parallelograms deduced from minimum-impedance values. (a) Rogers/BBC LS3/6, Z_{min} = 10.7Ω, 20Hz–20kHz; (b) KEF Corelli, Z_{min} = 6.4Ω, 20Hz–20kHz; (c) Quad ESL63, Z_{min} = 3.2Ω, 20Hz–20kHz; (d) Quad ESL63, Z_{min} = 5.2Ω, 40Hz–5kHz. Scale—vertical 5A/div; horizontal, 20V/div. 40V peak signal assumed.

three loudspeakers tested, and completing the parallelograms as described, gives the results shown in *Fig. 18*. These are not greatly different from the sweep results of *Fig. 15*, though the latter represent slightly easier amplifier-loading conditions.

The above parallelogram technique can occasionally give an unduly pessimistic prediction of the output level capability of a particular amplifier—loudspeaker combination. A good example of this would be a KEF 104/2 loudspeaker used with the amplifier to which *Fig. 9b* relates. This loudspeaker is unique in that, by the use of appropriate conjugate networks in the internal circuits, it has an almost purely resistive impedance of 4Ω throughout the audio spectrum. The amplifier can therefore produce about 32V peak across this loudspeaker, no matter what waveform may be involved, corresponding to a peak power of 256W. Since, with this loudspeaker, the current is zero when the voltage is zero, the rather severely restricted current capability of the amplifier at zero instantaneous output voltage is of no consequence. If, however, the above parallelogram technique was thoughtlessly applied in this case, a maximum peak output power of only about 64W, without significant distortion, would be predicted.

The rather awkward problems discussed concerning the relationship between the peak instantaneous current demands of loudspeakers under sine-wave and program conditions completely disappear, of course, ideally, when the conjugate-network technique is used to give a purely resistive impedance.

ADDENDUM. As a result of this work, a proposal for the addition of a new group of characteristics and measuring methods to IEC Publication 268-3 has been made. This addition is expected eventually to provide measuring methods for the characteristics of all

Continued on page 64

THE QB_3 VENTED BOX IS BEST

BY G.R. KOONCE
Contributing Editor

Various types of enclosures for housing dynamic bass loudspeakers produce subjectively different bass qualities. I recognize the dynamic capabilities of transmission line and horn loaded enclosures, but rather than to compare or discuss these types, I intend to focus on the various vented-box (VB) alignments, including the special case of the passive radiator, and closed-box (CB) systems.

I have built numerous speaker systems using CB and VB enclosures and to my ears the quasi third-order alignment (QB_3) VB produces the best bass quality. In this article, I propose a reason for this quality advantage with the QB_3 alignment.

Starting first with the VB, a study of the alignment charts[1,2] indicates the following:

- For QB_3 alignments the box tuned frequency (f_B) and the driver resonant frequency (f_s) are below the system −3dB cutoff frequency (f_3).

1. Bullock, R, "Thiele/Small and Vented Loudspeaker Designs," *SB* 4/80, 2, 3/81 and 1/82; vented box alignment charts.
2. Dickason, V., *The Loudspeaker Design Cookbook*, pp. 22, 23.

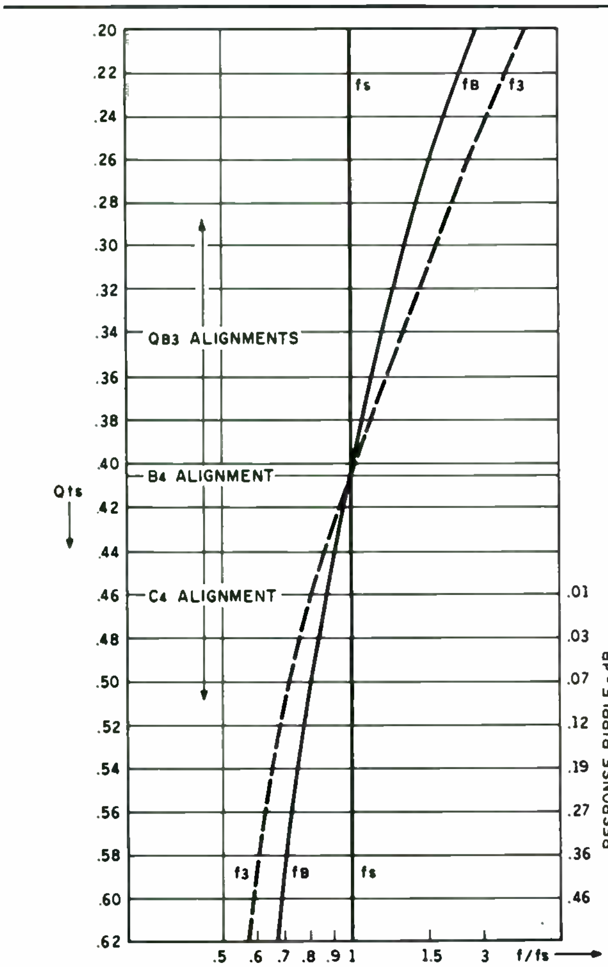


FIGURE 1: f_B/f_s and f_3/f_s versus Q_{ts}. Optimum flat vented box for Q_B = 7.

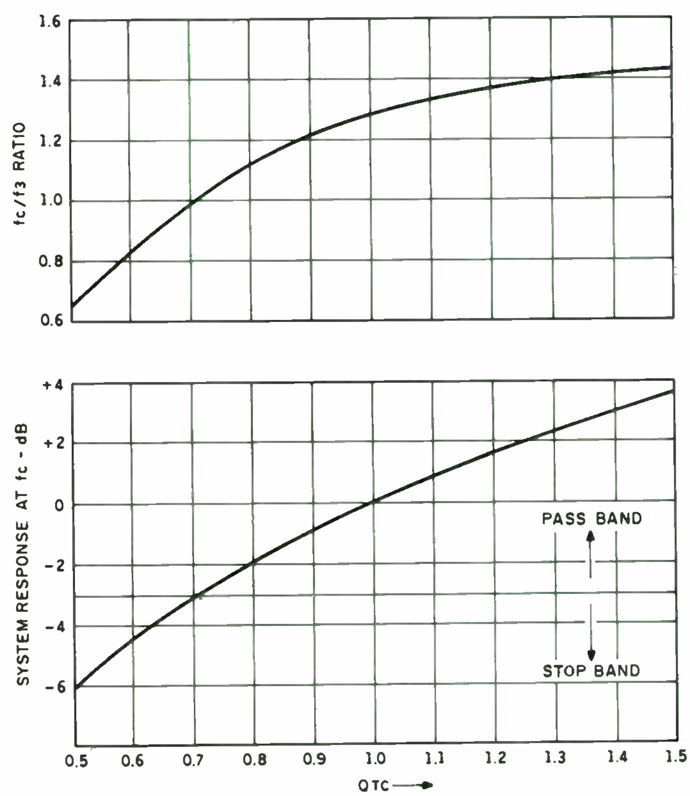


FIGURE 2: Closed-box alignment effects. (a) f_c/f_3 ratio; (b) system response at F_c in decibels.

• For the single point case of the fourth-order Butterworth (B_4) alignment, all three of the frequencies are approximately the same.

• For Chebychev (C_4) alignments, f_B and f_s are both above f_3.

Figure 1 indicates these relationships for flat VB alignments with a total box Q (Q_B or Q_L) = 7. The horizontal axis is frequency normalized to f_s. As driver Q (Q_{ts}) is reduced, the values of f_B and f_3 move higher in frequency, with f_3 always higher than f_B. Conversely, high driver Q_{ts} causes f_B and f_3 to move lower in frequency with f_B always above f_3. Frequencies f_B and f_s represent the two resonances that must be considered for the VB. Driver resonance f_s may shift slightly (to f_{sb}) in the VB; we can ignore this minor effect for this discussion.

The passive radiator (PR) system is somewhat more complex than the VB, but alignment charts[3,4] indicate the relationships of f_s, f_B and f_3 are basically the same as presented for the VB. At low driver Q_{ts}, f_s is less than f_B which is less than f_3.

An additional resonance is introduced, f_P (the PR unit resonance), which you can adjust by adding weight to tune the system. The value of f_P is related to the ratio of PR unit compliance (V_{ap}) to the box volume. Frequency f_P is always below f_B, being well below f_B for systems with low Q_{ts} drivers and PR units with a compliance equal to or greater than that of the driver ($V_{ap} \geq V_{as}$). Since typical PR systems with low Q_{ts} drivers will have the same characteristics as QB_3 VB systems, I will not go into any further details.

The CB has a single resonance to consider, system resonance (f_C), resulting from the upward shift of f_s caused by the combination of box and driver compliances. Using the equations presented in my article on the CB (see *SB* 2/84) the relationship of f_C and f_3 to the total system Q (Q_{tc}) can be developed. *Figure 2a* indicates the ratio of f_C to f_3 as a function of the selected Q_{tc} alignment. Note that for Q_{tc} greater than 0.707, the most commonly used alignments, f_C is above f_3.

Taking the generally used arbitrary definition that response above f_3 represents the system pass band, and frequencies below f_3 are in the stop band, an interesting point is observed. For the VB, only the QB_3 alignment has no resonances in the pass band. I believe this is a major factor in the superior bass quality I perceive with the QB_3 VB alignment. For the CB you must build with Q_{tc} less than 0.707 to move f_C into the stop band.

SYSTEM RESPONSE. A comparison of just how far the major resonances can be suppressed merits attention. *Figure 2b* indicates the CB must be built with Q_{tc} well below 0.707 to have the system response at f_C well below −3dB. By Q_{tc} equal to 0.5, the system response is down 6dB at f_C. *Figure 3* displays how far down the response of a QB_3 VB is at f_B and f_s. If you build the system with driver Q_{ts} below 0.325 then the response at f_B is down at least 5dB and the response at f_s is down at least 10dB.

Many think the locations of the two impedance peaks (f_L and f_H) in the VB input impedance are important, and I agree. Richard Small presents the equations to compute the location of f_L and f_H,[5] useful information to see if your tuning is going as expected.

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3. Knittel, M., "Passive Radiators a la Small," *SB* 3/81, p. 29, 30; passive radiator alignment charts.
4. Dickason, V., *The Loudspeaker Design Cookbook*, p. 31.
5. Small, R. H., "Vented-Box Loudspeaker Systems," *JAES*, June-Oct. 1973, Vol. 21, Nos. 5-8.

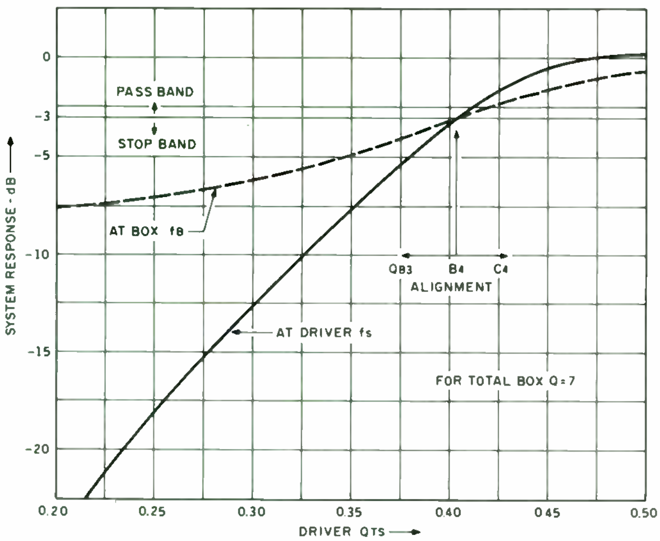

FIGURE 3: System response at F_S and F_B versus driver Q_{ts} for vented box; total Box Q = 7.

	TABLE I		
	−3dB CUTOFF FREQUENCY vs. ALIGNMENT		
	(Q_B=7)		
Driver f_s	**Driver Q_{ts}**	**System Align.**	**System f_3**
30Hz	0.25	QB_3	59.0Hz
30Hz	0.375	QB_3	42.4Hz
30Hz	0.405	B_4	30.0Hz
30Hz	0.50	C_4	21.5Hz

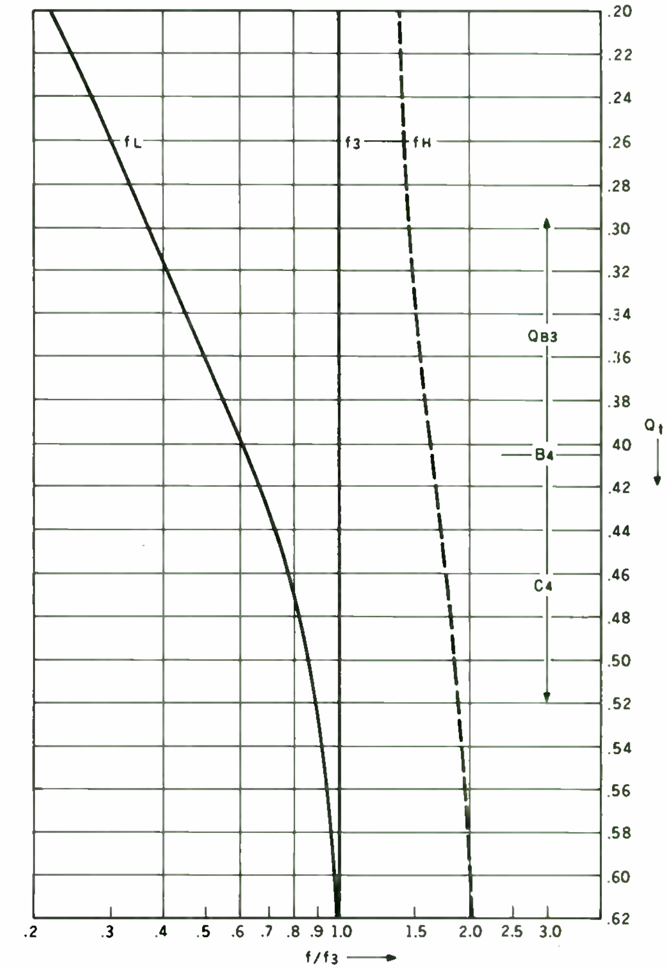

FIGURE 4: f_L/f_3 and f_H/f_3 versus Q_{ts}. Optimum flat vented box for Q_B = 7.

For optimum flat QB_3 through C_4 alignments (Q_B=7), f_L is always below f_s, and f_H above f_s, with their spread wider for low Q_{ts} alignments (i.e. alignments with high compliance ratios, Alpha). *Figure 4* shows how f_L and f_H are located with respect to the −3dB frequency f_3. Frequency f_L is below f_3 until driver Q_{ts} exceeds 0.62.

For QB_3 alignments, f_L is well suppressed into the stop band, system response at f_L being even lower than at f_s. For all alignments, f_H is in the pass band. For QB_3 alignments it is closer to f_3, but it is always a point that should receive consideration. *Figure 4* will allow you to estimate its location during system design; note that for high driver Q_{ts} with C_4 alignments, f_H is an octave above system cutoff.

CB systems with the woofer Q_{tc} in the 0.5 range should produce good bass quality. I have never built such a system, or even heard one, but they are used commercially. The penalty you pay is an extremely large enclosure and low power handling capability for the driver used. Alignments with Q_{tc} in the 0.5 range are practical with open back mid-range drivers and I have built these with good results.

Construction of VB systems using the QB_3 alignment with low Q_{ts} drivers results in small enclosures with relatively high power handling capability. If the QB_3 VB alignment has these advantages why is it so rarely used?

The answer: very few drivers exist that can be used to build "good" QB_3 systems. By "good" I mean systems with acceptably low f_3 values. A driver is required with low Q_{ts} and very low f_s. *Table 1* indicates the problem, comparing the f_3 value obtained for flat system alignments using drivers with f_s equal to 30Hz, but with various values of Q_{ts}. The low Q_{ts}/QB_3 alignments do not yield impressive f_3 values for a driver with f_s = 30Hz.

Of the many woofers with which I have tested and built systems, only three types produced good QB_3 system designs. Unfortunately, all were surplus speakers that cannot presently be purchased, but their desired common features are:

1. An excessively large magnet for the driver size and voice coil size, which yields the low Q_{ts} value needed. Music speakers fit this requirement, but have unacceptably high f_s values.

2. Rubber surround, which helps to produce the low f_s needed. The suspension must be very compliant. Generally these speakers test with a rather high mechanical Q (Q_{ms}).

3. Light cone, which helps to maintain the low Q_{ts}, but makes obtaining a low f_s very difficult.

There is one area where the QB_3 VB system will be inferior to other VB alignments and all CB systems. With the extremely compliant suspension required, the drivers are subject to damage from very low-frequency input. Be sure your system has the necessary infrasonic (sometimes called subsonic) filter to prevent this condition. I believe you should include this function in your systems, no matter what type speakers you use.

If you can obtain 10- or 12-inch woofers with Q_{ts} below 0.3 and an f_s value less than 20Hz, give them a try in a QB_3 VB alignment. If you use a passive crossover be sure to keep the woofer section resistance very low (0.2Ω max for

Continued on page 65

World Radio History

MAN OVERBOARD

BY BOB SCHOEN

I don't know whether you'd call my loudspeaker construction a labor of love (now I may know what that means), or I just needed to have a long-term project to noodle over, or I needed to put all my woodworking tools to use to justify their purchase, but I can honestly say I began this endeavor in April 1987 and finished just in time to enjoy the Christmas vacation.

The title came since I often wondered if I was getting a little carried away (overboard) with my project, which competed with boating and water-skiing during my summer hours.

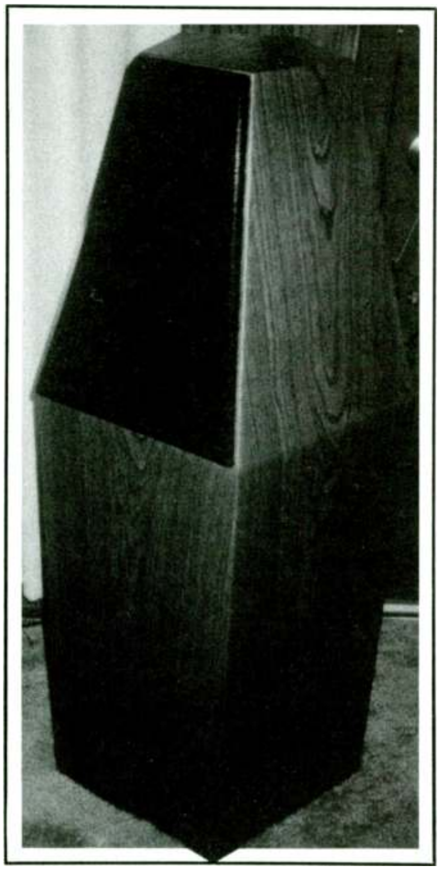

PHOTO 1: A truncated, convergent pentagonal box/tower, with red oak veneer and foam-lined grille.

Before getting into the details, I'll present the features:

- Audax 8" TPX bass/midrange;
- Audax 1" soft dome tweeter;
- Tuned port/Isobarik dual woofer enclosure;
- Truncated convergent pentagonal upper box over a pentagonal tower for nonparallel sides, small driver mounting face and uneven spacing between the upper box edges;
- Double wall construction in the upper box with long hair wool damping;
- "Front" woofer magnet support structure;
- A sand and glue putty used extensively in the upper chamber to add stiffness, weight, air tightness and damping;
- Lateral ribs in the lower chamber;
- Generous application of hot melt glue to the woofer "spider" to dampen driver structural resonance;
- Fiberglass damping on three sides and the bottom of the lower chamber;
- Layer of automotive windshield mounting urethane on the lower box walls;
- ¼" closed-cell foam weatherstripping covering the entire front box surface and the inboard facing surface of the grille mounting molding to eliminate reflections;
- Fourth-order crossover using Solen polypropylene capacitors, air core inductors, IAR WonderCaps (bypass caps), WonderSolder, and Monster Cable.

BACKGROUND. Having toyed with a bass guitar in junior high, I've always appreciated a good low end. The dilemma arose in reading all the back issues of *Speaker Builder* and various reviews in other magazines that stated a larger woofer will go lower in frequency but sacrifice impact and speed. My first pair of homebuilts, back in college, had a decent low end. An EPI clinic (remember those?) technician told me my Jensen acoustic suspension monsters, for a rookie, were the best 15-inchers he'd tested and could shame some pros. I wasn't prepared to do without that luxury.

After reading about the Isobarik design in *SB*, I thought my prayers were answered; a second set of woofers could cut the box size in half. This sounded like the best way to get my low bass without moving some major furniture out (my wife was relieved). I used computer simulations to get a −3dB frequency of 25Hz with a fairly large box behind the Audax, so I figured I could get the low bass without sacrificing woofer speed.

Concerns from *SB* readers' letters noted the muddy midrange response caused by (supposedly) the rear woofer blowing behind the front woofer. Not wanting to add more crossover complexity, I decided not to "choke" off the rear woofer, but to orient it at 90° and absorb whatever mid and high frequencies I could with damping material (*Fig. 1*). Later, I found an ad for a Dynaudio model, the Consequence, using the same principles. I decided, if it's good enough for Dynaudio, it's good enough for me.

DRIVER DILEMMA. Having decided to experiment with the Isobarik, for cost reasons I decided to go with a two-way system in a $400—$500 range. If this double woofer concept was cost-effective, I would consider drivers of Dynaudio or KEF quality for my next system.

Two articles in *Stereophile*[1,2] and promos in mail order catalogs persuaded me to try Dick Olsher's Dahlia system with obvious modifications. It seemed the

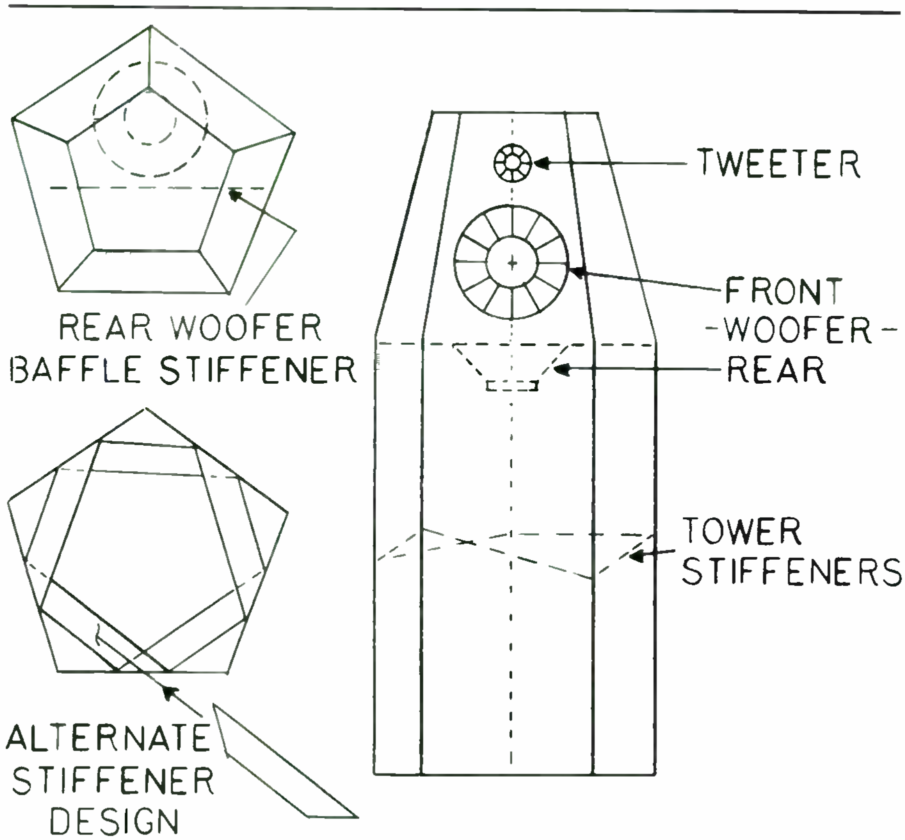

FIGURE 1: Cabinet construction.

advertised midrange clarity of the TPX driver might be coupled to the Isobarik to account for the admitted bass deficiency noted with Mr. Olsher's box.

Before getting too far along, I must report I ran into trouble with the Audax drivers I received. I had borrowed enough equipment from a friend to check the actual parameters, according to Mr. Bullock's procedures in *SB*. Rather than the advertised natural frequency of 33Hz, my drivers tested in the 60Hz range, even after lengthy break-in. The vendor I purchased the drivers from offered no useful assistance. They wanted to keep sending me additional samples until I found a pair I liked, since they had no testing capability.

I ended up talking directly to Polydax. It turns out they had a production problem with an early batch and were very willing to take mine back, test replacements to guarantee specs, and ship them directly to me. They did acknowledge the specs had been revised to a new natural frequency of 49Hz. At this point, I decided to accept this compromise. I measured the new set in the mid-40Hz range, at least. I will say Polydax was very helpful, and I have since received much better response from two different mail-order, parts vendors.

CROSSOVER. From all my readings, I know how complex and important the crossover is. Since I don't fully understand all the theory, I decided to stick with Mr. Olsher's calculations, adjusting for the parallel woofer impedance (*Fig. 2*). I'll be the first to admit I'm more of a mechanical than electrical type.

CABINET PLANNING. I originally planned to build a pentagonal tower with the top surface angled at 15° to eliminate any parallel internal surfaces. Having access to a computer-aided design system at work, it was fairly easy to find the compound angles required at the top, and the odd shaped top itself, by directing the computer to actually do the drawing and measure the angles. I used a computer program provided by Polydax to obtain a box volume that would result in minimal response hump, which indicated a required height of over three feet. I abandoned this design when I saw the tilt necessary to align the drivers (15-20°, as recommended by Mr. Olsher) would cause the tall tower to become unstable.

At this point, I designed a "flower pot" upper box over the pentagon tower. With this, I could incorporate the recommended tilt angle in the driver mounting face. This presented a greater challenge in determining the cut angles. At first, I even considered a seven- or nine-sided box. This inspiration led me to develop a computer program for my Commodore 64 to predict all the necessary cut angles and interior volume, given the vertical height, number of box sides, speaker diameter, side spacing to the speaker, bottom spacing to the speaker, and the desired tilt angle of the upper box faces.

Figure 3 shows a sample output and *Fig. 4* is my program listing. For the volume I required, the bottom box ended up 28″ tall with a 16″ tall upper. The 44″ overall height allows the drivers to project above our sofa sides, another design objective. This became especially nice since my finished system borders this sofa, angled in for imaging purposes. It also puts them right at ear level when

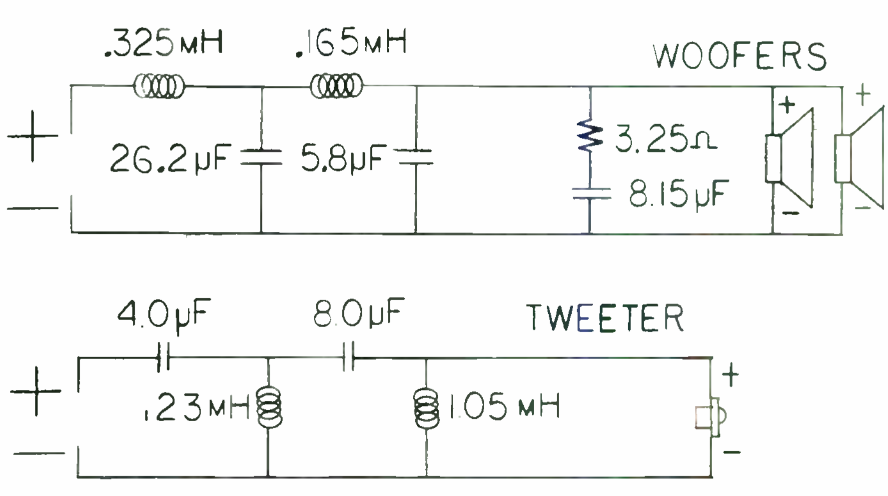

FIGURE 2: Olsher's fourth-order crossover, modified for the double woofer.

FOOTNOTES

1. Olsher, D. "The Dahlia Speaker System: A DIY Project," *Stereophile*, Feb. 1986.
2. Olsher, D. "The Dahlia Debra," *Stereophile*, April 1987.

```
HEIGHT= 16
NUMBER OF SIDES= 5
TILT ANGLE= 15
SPEAKER DIAMETER= 8
SIDE SPACING TO WOOFER= 1
BOTTOM SPACING TO WOOFER= 1
TOP INSIDE WIDTH= 5.65078795
BOTTOM INSIDE WIDTH= 11.8804283
CUT HEIGHT= 16.7653807
ANGLES TO CUT PANELS...
     PANEL SIDE BEVEL= 34.5935196
     PANEL VERT ANGLE= 10.6496978
VOLUME= 2204.13828

HEIGHT= 25
NUMBER OF SIDES= 5
TILT ANGLE= 0
SPEAKER DIAMETER= 11.375
SIDE SPACING TO WOOFER= 0
BOTTOM SPACING TO WOOFER= 0
TOP INSIDE WIDTH= 11.375
BOTTOM INSIDE WIDTH= 11.375
CUT HEIGHT= 25
ANGLES TO CUT PANELS...
     PANEL SIDE BEVEL= 35.9995482
     PANEL VERT ANGLE= 0
VOLUME= 5565.34738
```

FIGURE 3: POLYBOX output.

```
10 PRINT"POLYBOX SPEAKER ENCLOSURE PROGRAM"
100 INPUT"BOX HEIGHT?";H
200 INPUT"NUMBER OF SIDES?";N
300 INPUT"TILT ANGLE?";LDEG
400 INPUT"WOOFER DIAMETER?";S
500 INPUT"SIDE SPACING TO WOOFER?";P
600 INPUT"BOTTOM SPACING TO WOOFER?";Q
650 INPUT"PANEL THICKNESS?";TH
700 Z=TAN(3.14159/N)
800 L=LDEG*.01745292
900 R=(S+2*P)/(2*Z)+(S/2+Q)*SIN(L)
1000 A1=((R-H*TAN(L))↑2)*Z*N
1100 A2=(R↑2)*Z*N
1200 V=(A1+A2+SQR(A1*A2))*H/3
1300 F1=2*Z*(R-H*TAN(L))
1400 F2=2*Z*R
1500 B=(COS(1.5708-3.14159/N))*COS(L)
1600 B=-ATN(B/SQR(-B*B+1))+1.5708
1700 B=90-B*57.29578
1800 C=ATN(COS(L)*(F2-F1)/(2*H))
1900 C=C*57.2957795
1910 H2=H/COS(L)+TH*TAN(L)
1920 OPEN 1,4
2000 PRINT#1,"HEIGHT=";H
2100 PRINT#1,"NUMBER OF SIDES=";N
2200 PRINT#1,"TILT ANGLE=";L/.01745292
2300 PRINT#1,"SPEAKER DIAMETER=";S
2400 PRINT#1,"SIDE SPACING TO WOOFER=";P
2500 PRINT#1,"BOTTOM SPACING TO WOOFER=";Q
2600 PRINT#1,"TOP INSIDE WIDTH=";F1
2700 PRINT#1,"BOTTOM INSIDE WIDTH=";F2
2740 PRINT#1,"CUT HEIGHT=";H2
2800 PRINT#1,"ANGLES TO CUT PANELS..."
2900 PRINT#1,"     PANEL SIDE BEVEL=";B
3000 PRINT#1,"     PANEL VERT ANGLE=";C
3100 PRINT#1,"VOLUME=";V
3200 PRINT#1,""
3250 CLOSE 1
3300 PRINT"F1 - TO RUN PROGRAM WITH CURRENT VALUES"
3400 PRINT"F2 - TO ENTER BOX HEIGHT"
3500 PRINT"F3 - TO ENTER NUMBER OF SIDES"
3600 PRINT"F4 - TO ENTER THE FACE TILT-BACK ANGLE"
3700 PRINT"F5 - TO ENTER THE WOOFER DIAMETER"
3800 PRINT"F6 - TO ENTER WOOFER SPACING TO THE SIDE    EDGE"
3900 PRINT"F7 - TO ENTER WOOFER SPACING TO THE         BOTTOM EDGE"
4000 GET K$:IF K$="" THEN 4000
4100 IF K$=CHR$(133) THEN GOTO 700
4200 IF K$=CHR$(134) THEN GOSUB 5200
4300 IF K$=CHR$(135) THEN GOSUB 5400
4400 IF K$=CHR$(136) THEN GOSUB 5600
4500 IF K$=CHR$(137) THEN GOSUB 5100
4600 IF K$=CHR$(138) THEN GOSUB 5300
4700 IF K$=CHR$(139) THEN GOSUB 5500
4800 GOTO 3300
5100 INPUT"HEIGHT?";H
5150 RETURN
5200 INPUT"NUMBER OF SIDES?";N
5250 RETURN
5300 INPUT"TILT ANGLE?";LDEG
5350 RETURN
5400 INPUT"WOOFER DIAMETER?";S
5450 RETURN
5500 INPUT"SIDE SPACING TO WOOFER?";P
5550 RETURN
5600 INPUT"BOTTOM SPACING TO WOOFER?";Q
5610 RETURN

READY.
```

FIGURE 4: Speaker enclosure program listing.

seated across from the sofa, my favorite spot.

BOX CONSTRUCTION. I strongly recommend, especially when building any nonrectangular cabinet, you use some scrap, and using the appropriate angles, cut out a scale model. Dry assemble this model (I rubber-banded mine together) and carefully examine the matchup of the sides to ensure the angle settings on your saw are exact. My 10″ Craftsman table saw is a wonderful tool, but for precision work I do not trust the sheet metal tab pointing at a decal angle indicator. Besides, when I describe my jig for cutting the upper box faces, you will realize the value in cutting the model.

The next challenge was this jig (*Fig. 5*), necessary because the 16″ tall faces would not fit between my Craftsman miter and the saw blade, given the length of the single miter guide runner.

As you can see, two runners provide a slip fit to the guide grooves in the table saw top. The runners must be cut long enough to allow for the height of the box being cut, the exposed blade length, the width of the new "miter" board, and the ⅛″ thick plate stiffener on the other side of the blade. I'm sure some shop teacher is cringing at the prospect of the plate stiffener being intact at the cutoff side of the blade, for kickback reasons, but my 10 fingers prove that if you are careful and go through the motions prior to turning on the saw, it can be done.

I was hoping the runners would be snug enough to the grooves so I could use simple trig to set the angle (POLYBOX vertical angle) at the miter board and not reinforce the setup. However, I found the angle would not be held accurately without support. In retrospect, I probably could have put the plate at the miter end, but this method would have wasted more material. Be sure to fix the runners to each other, via the plate before setting the miter angle. The miter height must be at least twice the thickness of the box wall to allow blade pass through (assuming the exposed blade height just clears the material).

Next, cut the desired height of the box face at the required tilt angle from a 4 by 8-foot sheet of ¾″ thick Timblend (high density particle board). I then cut rectangles a little oversized in the width to make the boards more manageable in the jig. Now set the saw blade tilt angle to the POLYBOX bevel angle (from *Fig. 3*).

After cutting one side, mark the inside width on the panel, flip end-over-end and spin it 180° to cut the other side without resetting any angles. The panel

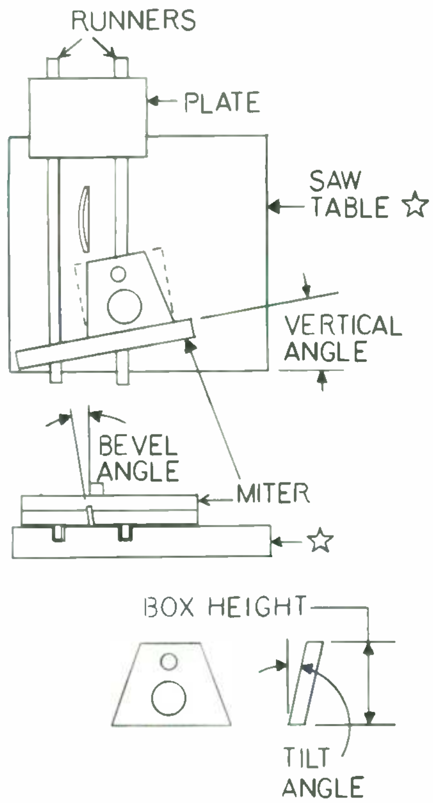

FIGURE 5: Cutting the angles on a table saw.

width can be transferred to the jig for future ease, and you can cut the other side. (Do not discard these cutoffs, as they will be used later as panel stiffeners.)

As I stated before, this is reasonably complicated, and the angles are sure to be off a little bit, so be sure to try your procedure on scrap first. When you dry assemble the model and look closely at the angle fits, you will probably need to do some tweaking. Since the jig is now rigid, the only adjustment possible is the saw blade tilt (side bevel angle). You may not end up with the exact tilt angle, but you will be very close.

After verifying the angles by dry assembly, cut the driver mounting holes. I cut the irregularly shaped tweeter mounting with a saber saw. The woofer hole was made nice and round by drilling a hole at the woofer center to allow a snug fit bolt to peek through. Using the edge contour guide and a small diameter straight bit on my router, I set the required radius of the woofer hole between the bit and a convenient place on the guide where the through bolt would ride. Setting the bit cutting depth to just under half the panel thickness, I cut by rotating the router around the bolt. Now flip the panel over, install the bolt from the other direction and cut the identical pattern on the second side. [*Most saber saws have an attachment for cutting round holes also.—Ed.*]

Since I set the cut depth to less than half, a sliver of wood holds the cutoff and the centering bolt intact as you complete the circle. A sharp rap at the center with a mallet will free the cutoff, a near perfect circle. Do not throw this away, as it will be used later as a panel doubler. Use a file to clean off the rough edge at the center of the panel opening.

Set the upper box jig aside and get out the rip fence to cut the pentagonal lower tower. Adjust the blade angle (36° for a pentagon) and construct another model to ensure the angle is accurate. Use the outside width of the upper box at its base to cut the tower sides. The tilt angle yields a smaller inside width on the upper box that should not be used to cut the tower.

Notice in the second sample POLYBOX output that the height is 26", less than the 28" height stated earlier, to allow for the top and bottom boards, with ¼-inch insets. The tilt angle is zero degrees; woofer spacings, zero; and the woofer size is measured from the scrap model, or calculated from the outside width of the upper box. These values are required to accurately calculate the lower box volume, which, according to isobarik theory should be half that of a single-woofer tuned, vented box.

I chose my volume larger than ideal, since every article I've read says to oversize for errors, bracing, and so on. It is easier to decrease than increase volume, as we all know.

ASSEMBLY. Putting the boxes together was another challenging aspect of construction. Not having any 90° angles with which I could use corner clamps, I was at the mercy of the accuracy of the angle cuts. This is why I am so emphatic about models. Before describing my methods, I will say the results were splendid, less than 1/32" off, easily within sanding tolerances.

First, prime all the abutting surfaces with yellow carpenter's glue and allow to dry. Also apply a layer to the edges of the driver openings to prevent any flakes from vibrating loose.

Start by assembling the lower box. Apply an even layer of glue to each board edge. Stand each board up, leaning one against the next, until all five are in position. Use a bungee cord around the middle to bring the sides together. Use three or four band clamps to apply pressure for curing. Remember to place wax paper between any clamps or bungee cords, and the sides, to prevent gluing them to the box. Use a rubber mallet to tap on the corners from the inside to align the edges as required. Stand the box on a flat surface and tap the ends to get them even as well, before the clamps are too tight.

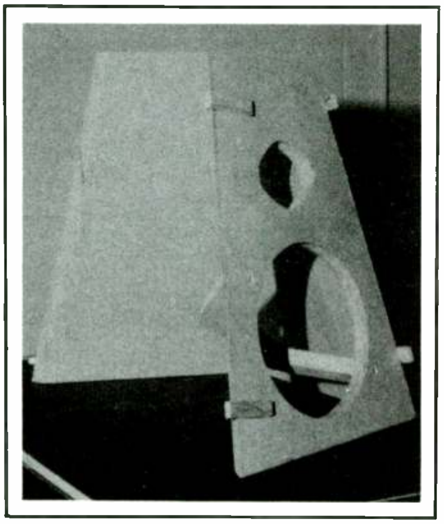

PHOTO 2: The front woofer support inside the box.

After you tighten the band clamps, insert small wood blocks between the panels and the bands at the center of each panel and work the blocks toward each corner to provide additional force on the joints. Continually monitor the matchup of the edges to ensure they are flush. The usual finger pressure along the inside corners to ensure a smooth continuous bond is required on all joints. Be prepared to work quickly as the glue begins drying in the time between the edge coating and the final alignment. Accuracy will save a lot of sanding later.

Use the same basic procedures for the upper box. However, the band clamps want to slide off the sloping sides; instead, use ropes, some good Boy Scout knots, and tourniquet techniques to apply the pressure (*Photos 2* and *3*). To keep the ropes from sliding, use hot melt glue to temporarily attach small blocks to the board edges. The ropes will be held in place and the blocks will help you align all the edges. Afterward, a tap will free the blocks. Putty any rough areas and sand smooth if you choose to paint. I veneered mine, so this step was not critical.

Using the cutoff you saved from the upper boards and plenty of glue, glue doublers to the four inner surfaces which have no drivers. Two small finishing nails help hold the doublers in position until the glue dries.

Since I will apply damping material to the entire driver mounting face, I do not recess the driver into the baffle surface. The irregular mounting flange of the Audax helped me to reach this conclusion.

Now, knowing the mounting surface, the panel thickness and the depth of the driver magnet from the mounting surface, cut out the driver support to go inside the box. Allow about 1/16" for the silicone to attach the driver later in final assembly. Screw and glue this support to the inside of the upper box, centered about the woofer opening (*Photos 2* and *3*).

The next step is the glue/sand "putty" application. Mix some clean sand with the yellow glue until you get a fairly thick putty. In general, when you think you have it thick enough, add more sand. You will be surprised how much sand the glue absorbs. Stuff the goop into each corner, including the doubler corners and at the base of the woofer support. Plan on doing one side per day to allow drying without runs.

Once the towers are dry, use them to trace a pattern for the four end panels. Allow for the panel thickness, as these end pieces fit inside the tower. On the two upper end panels, use the same router technique to cut out the rear woofer holes. I don't know whether one position is better than another, but I placed mine to the rear of the cabinet for two reasons. I wanted to keep the woofer magnets as far from each other as possible, and the panel would be stiffer. Add a stiffener across the center of this panel near the woofer hole. Use one of the circular cutouts to act as a doubler for the bottom panels. Again, use plenty of glue and a few small finishing nails to hold it in place while drying.

Now check to see how well the upper and lower boxes fit together. Try rotating one until you find the best match and mark the pieces for later identification. Both boxes needed light sanding to ensure flat surfaces.

My trick is to use a piece of particle board about 3" by 36" as a sanding block. I tried a 2 by 4, but it is not flat enough. Use masking tape to attach a section of coarse sandpaper just off-center of the long sanding block. The long end lies flat along the opposite and adjacent sides to ensure flat, even sanding. Verify your work on both boxes by periodically checking with a good straightedge and by stacking them up. Patience will result in a better fit between boxes and less filler required later.

Next install the rear woofer mounting panel in the tower box. Use three finishing nails to hold the panel in position with a recess of at least 1/4" from the top edge to allow gluing. Apply a bead around both sides of the panel, encouraging it into the crack. I helped it to stay

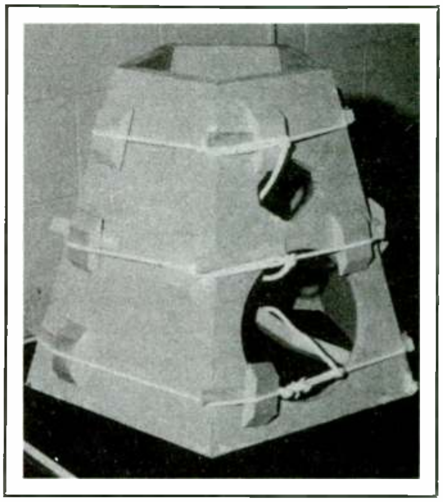

PHOTO 3: Rope and block technique to apply pressure for final assembly.

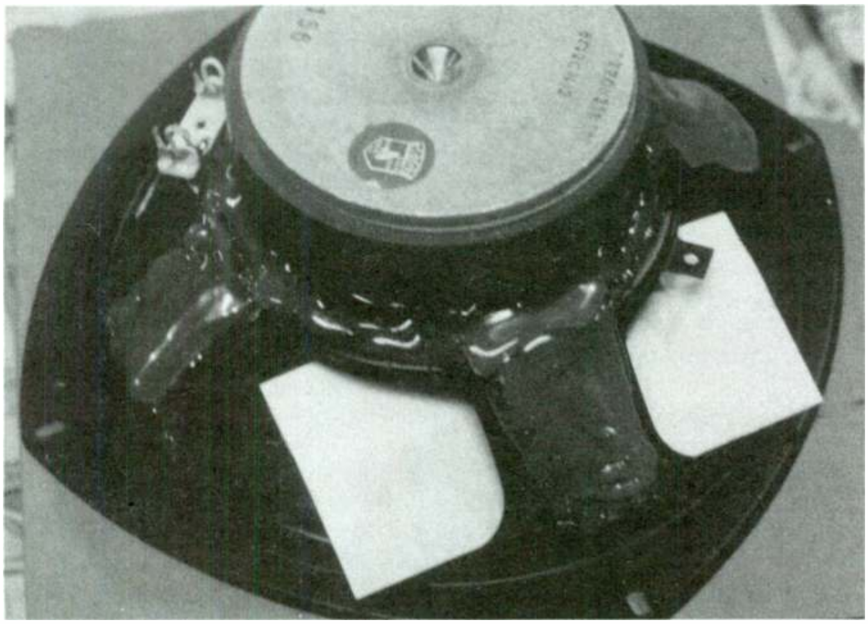

PHOTO 4: The author applied a thick layer of glue over the driver's rear surface to deaden spider ringing.

put by pressing sawdust in behind. It may take more than one session to be sure of a good, air-tight joint.

To retain the bottom panel, locate ribs 1″ up from the bottom, to allow for the bottom thickness and ¼″ gap to the floor. These ribs should be cut for a tight fit without too much glue or caulking afterward, and big enough to provide a good structural attachment for the bottom.

Again on the tower, use more scrap, or whatever suits you, to randomly glue stiffeners around the middle of each lower panel to break up these large areas (*Fig. 1*). Allowing room for mounting the crossovers, apply a series of thick urethane beads over the entire inside surface of the lower box. (I was fortunate to have a very cheap source that allowed me to use nine standard size tubes on each box.) I ended up with a heavy, stiff upper box atop a well-damped lower. If possible, allow the urethane to dry outside as it takes a while to cure and stinks in the meantime.

While the tower airs out, place the upper on its head to trace out its top piece. After you sand the upper box top edges flush, cut out and glue on the top, using more sand/glue putty to ensure a good seal. I did not try to cut the tilt angle into the top as it was easier to sand it flush later with my belt sander using coarse grit.

To assemble the two box units, first attach blocks, using band clamps, or hot melt glue, protruding past the edges of the tower to hold the upper in position while the glue sets. At this time, do not apply a lot of glue to this joint, it will only run down the sides. Go around a second (and third) time to fill in the gaps. Apply weight to the top to minimize these gaps.

It may be redundant, but at this point I whipped up another batch of sand/glue putty to go around the inside joint where the two boxes and the rear woofer baffle come together.

Now the major construction is complete. Go around all the seams and joints that you want to be airtight and finger in more glue and sawdust if necessary, until flush. Remove any guide blocks, then putty where required and sand smooth to suit your finishing needs.

Finally, I drilled holes for the T-nuts (just oversized enough to force them in for sealing) to use in mounting the drivers and bottom access board (two per side for a total of ten per box bottom). Use a Channel Lock type pliers to force the nuts in. After veneering, I had to go back to find these mounting holes and open them up without messing up the finish. In retrospect, I wish I had hammered in the nuts before assembling the panels, or forced each one in place with a C-clamp. It is no fun going back into a cabinet to reseat a nut that spins or pushes out ("Murphy" at his best). I should have used a little silicone in each nut hole for insurance.

FINISHING. If you elect to paint, you will hear no criticism from me. I can appreciate those who do not want the hassle of veneering.

Finding veneer in the size I needed was not easy. The local home improvement shops did carry rolls about four inches wide, but did not have enough in stock. Besides, it would have cost me another $100 or so to go that route. I ended up at a local roughcut lumber wholesale house that our wood shop at work uses for die models (good connections are something to appreciate). Only a little persuasion was required to get a cash sale. We went into the lumberyard, and opened a shed door to find a stack of twelve sheets of red oak veneer about 14 inches wide by 10 feet, all cut from the same tree. When he gave me a price per sheet, I quietly contained my enthusiasm and walked out with all twelve sheets for $55. Being a woodworker with a fondness for oak, I figured I could use it all.

With one large sheet, I found I could cover two sides, top to bottom, in a seamless flow of beautiful oak grain. Without lecturing on the methods of applying veneer, I was able to cover the 15° tilt without cutting. I found a cloth-wrapped, rubber mallet is better than beating on a block to flatten the oak to the particle board and conform it through the tilt line. Rounding off this line with the belt sander would have eased veneering this transition area, but the crisp definition is nice too.

I do one side per night to eliminate fatigue and anxiety related mistakes. After each side was fully dried, I went along each edge first with an X-acto knife, then a medium coarse wood file, and finally fine sandpaper to get each edge flush. Using this method, working from the back of the box to the front in each direction, I created an almost seamless looking block of oak. After completing the sides, do the top, carefully orienting the grain with the front face. After opening up the driver mounting holes, I applied a little stain and polyurethane to match the family room paneling.

The odd shaped front baffle presents another challenge in building the grilles. I carefully laid out the dimensions on a piece of scrap paneling and glued down ½″ quarter round molding to maintain the perimeter of the size and shape I wanted. I cut the pieces, using the POLYBOX angles, applied glue to the corners, and wedged these pieces into the fixture for overnight drying. Be sure to use straight sections of molding to ensure good results. Apply a flat board with weight on it to maintain flatness during curing. Do not be tempted to test the rigidity afterwards; it must support only a piece of cloth.

Paint the entire frame a flat black to conceal it behind the cloth. After it is dry, apply a continuous strip of ½″ wide, double-sided sticky foam tape to the frame, facing the box. I bought a few yards of very thin, black polyester from a local sewing fabric store. Cut a piece about 2″ larger than the frame. Stretch the cloth enough to eliminate wrinkles and stick it to the tape for retention. I had doubts whether this approach would hold, but after two months, it shows no sign of releasing. Fold the cloth into the corners as best you can, and trim the excess.

To minimize reflections from the frame, apply a layer of ½″ wide black, closed-cell foam around its inside surface.

FINAL ASSEMBLY. In assembling the crossovers, be sure to locate the inductors at opposite ends of your mounting board with their axes oriented 90° to each other. I used IAR Wondersolder in all the joints; it flows easily and makes a nice joint, especially when working with 10- or 12-gauge wire. By the way, my wife's leaded glass soldering iron was the only device I could use on these wire sizes with any success.

I used Solen polypropylene capacitors with IAR WonderCaps for small value bypasses, and air core inductors. Mount the crossovers securely inside the tower. Route the three speaker wires away from the side you elect to port and mount the binding posts. Be sure to seal the wires going through the rear woofer baffle and the binding post area.

TRIAL AND ERROR. I put everything together to fire them up, and found that the stuffing I used in the upper box ends up settling on the upward facing rear woofer. To prevent this, glue two dowel rods across the rear woofer opening (opposite side of the woofer of course). Next, lay out a sheet of cheesecloth about twice the size of the rear woofer baffle. On the center of the cloth, lightly fluff a 2″ layer of long hair wool to form a "pillow" the size of the baffle. Fold the loose corners of cheesecloth over the top, invert the pillow to get the multiple layers of cloth toward the woofer and place it on the dowel rods, over the baffle. In the event I wish to remove the driver later, this method keeps the wool from raining down into the tower. Also, I was not sure one layer of cheesecloth would be enough to keep out the vibrating fibers.

I chose wool over other materials, as Mr. Olsher and others have stated it is more suited to damping mid and high frequencies while passing low frequencies. I wanted those lows to get through to the rear woofer for the Isobarik effects.

Again, I may be going overboard, but at this stage I filled the upper box with more wool, fluffing it as much as possible, to barely fill the cavity.

The last trick before mounting the drivers is to dampen the woofer frames or "spiders." I noticed a swift knuckle to the spider resulted in a reasonably loud and undamped ring. The internal front woofer support is supposed to help this, but why not try an easy improvement?

Using my hot melt glue gun I applied a ⅛ to ¼″ thick layer over the entire rear surface except for the mounting area (*Photo 4*). Place a piece of paper between the spider and cone to catch any excess. Even the folks at Polydax noticed the effectiveness of the glue in dramatically deadening the spider ring; I left it on one of the drivers I had returned.

Next, solder the wires to the driver leads. Use black electrical tape or silicon cement to make sure the wire ends do not short out against the magnet.

To install the front woofer, lay a healthy bead of RTV on the inside support where the magnet will contact, to help stop spider resonance. Lay a small bead around the lip of the box opening. Align the screw holes, insert the driver, and apply a very slight rotation to smooth the RTV. Apply a small drop of Loc-Tite or similar product to the mounting screws and snug them up. Do the same for the tweeter, except for the inner support, of course.

For the rear woofer, place a ring of foam around the front face just inside the mounting screws, use more Loc-Tite on the screws, and snugly install it from the bottom. Connect the crossover to the drivers and binding posts.

I stuffed the tower with a single sheet of fiberglass. The sheet was big enough to cover three walls and long enough to fold over to cover the bottom too. I placed more foam around the bottom mounting flange, more Loc-Tite, then firmly tightened down the bottom board.

With a soft lead pencil, lightly trace the outline of the grille in position on the front baffle. Mark parallel lines inside these at a distance equal to the width of the grille assembly side (½″ molding plus the foam thickness). Now position black, closed-cell foam inside your scribe lines. This holds the grille assembly in place. Make adjustments before the foam has a chance to get a good grip, it should not move later. Gravity and the tilt angle hold it in place very nicely.

Once the outline is finalized, completely cover the front baffle with foam tape, including the driver mounting screws and exposed metal structure, right up to the dome and cone surround. In my prototype stages, I tried a sheet of automotive, noise dampening, felt type carpeting. It completely covered the front except for two circular cutouts and worked quite well except it shed fibers that stuck to the tweeter dome. My brother had better luck by spraying the felt with adhesive to hold down the fibers. I keep stressing closed-cell foam as I believe open cell will not block air leaks or absorb sound very well.

RESULTS. Connect the amp, crank it up, and enjoy. I admit I have not made the rounds lately to listen to state-of-the-art models, other than a quick listen to some KEF, ADS and Paradigm speakers in the $400-$500 range, but what matters is that I am more than satisfied with my results. The bass goes as low as my old Jensens ever did, but with the tightness and impact of the smaller driver. Bass resonances or boominess are neither heard, nor felt from the box sides. I am completely sold on the Isobarik principle. I am particularly impressed with the response on some good drum snaps; very crisp and authoritative. Imaging seems good; I know my room is poorly arranged with a picture window between the speakers and a brick and paneled wall across from it. I am pleased with the high end.

I confirmed a slight midrange deficiency at the crossover frequency with an analyzer borrowed from work. Testing capacitors indicated all are on the low side of nominal, so some tweaking is required. Perhaps the woofer isn't willing to go that high without a little more persuasion, or I could bring the tweeter down a little.

FUTURE OPPORTUNITIES. There are many sonic improvements I have not tried. Several that come to mind: the quantity of wool (if any) in the upper box, the amount of fiberglass in the tower, a thin layer of felt over the foam-taped front baffle, gold-plated binding posts at the box and amp connections, tweaking the crossover for more nearly flat response, bi-wiring, and some genuinely scientific fine tuning of the tower volume and port size.

To minimize breathing, I used a 4″ diameter port. Theory indicated, with the proper sized box, the port should be about 4″ long. During one tryout, when I discovered fibers settling on the rear woofer, my ears told me an 8″ length provided more reinforcement. After putting in the pillow, it sounds better with no length, just the hole in the box. Perhaps someone who understands acoustic principles better than I do can explain why the additional stuffing in the upper box affects the tower tuning so much.

I also think that putting the port on one of the rear surfaces (actually the side, when the speakers are angled in), reinforces the lowest, nondirectional frequencies, without any upper bass or midrange coming through the port to cause multiple sources. I believe KEF research on the woofer-inside-the-box concept indicates a gentle frequency rolloff above 150Hz emitted from the port. With the pentagon, these frequencies are shot out the side to be absorbed by the surroundings. I welcome any comments.

```
HEIGHT= 16
NUMBER OF SIDES= 5
TILT ANGLE= 15
SPEAKER DIAMETER= 8
SIDE SPACING TO WOOFER= 2
BOTTOM SPACING TO WOOFER= 2
TOP INSIDE WIDTH= 8.02687361
BOTTOM INSIDE WIDTH= 14.2565139
CUT HEIGHT= 16.7653807
ANGLES TO CUT PANELS...
     PANEL SIDE BEVEL= 34.5935196
     PANEL VERT ANGLE= 10.6496978
VOLUME= 3506.23707

HEIGHT= 26
NUMBER OF SIDES= 5
TILT ANGLE= 0
SPEAKER DIAMETER= 14.25
SIDE SPACING TO WOOFER= 0
BOTTOM SPACING TO WOOFER= 0
TOP INSIDE WIDTH= 14.25
BOTTOM INSIDE WIDTH= 14.25
CUT HEIGHT= 26
ANGLES TO CUT PANELS...
     PANEL SIDE BEVEL= 35.9995482
     PANEL VERT ANGLE= 0
VOLUME= 9083.48564
```

FIGURE 3a: Alternative Polybox configuration.

One change I plan to try is to switch to the MB Electronics titanium dome tweeter, as many have recommended it. The crossover Martin Colloms suggested in a follow-up article[3] a first-order woofer and third-order tweeter, sounds worthwhile also, budget permitting. Also, when I win the Lotto, I will go to IAR WonderCaps and Sidewinder coils throughout.

Perhaps bi-wiring sounds better than just doubling the wires, because broadband back electromotive force (emf) is reduced in each wire. Would it make any sense to put the crossovers at the amp side and keep the total signal from getting into each wire at the front end? Maybe someone with a better system than mine can try that and let us know.

Given the weight of these boxes, I do not anticipate much benefit from spiking to the floor, but I may at least try something in the future. A gentle push tends to rock the loudspeaker a little on the carpet and its pad.

3. Colloms, M., "A European Dahlia," *Stereophile*, Sept. 1987.

Part II

THE SWAN IV SPEAKER SYSTEM

BY JOSEPH D'APPOLITO and JAMES W. BOCK
Contributing Editor

In Part I the authors explained the background and development of the SWAN IV, including design criteria, crossover considerations and driver selection. In the second part, the authors roll up their sleeves and show us the attention that goes into cabinet construction and finish, to conclude their system.

Structural Considerations

SYMMETRICAL SATELLITE. We constructed the satellite enclosures from relatively dense particle board and ¼-inch plywood veneer epoxy bonded to the exterior. For support we used corner glue blocks, or cleats with epoxy resin adhesive. The resultant box is rigid and the laminated walls are self-damping to avoid resonance.

The front panel fits into a recess of extending edges of the veneer-faced plywood and hardwood pieces, epoxy bonded and later shaped to quarter round, to avoid corner mitering the plywood veneer. A one-inch deep depression in the rear exterior provides space for connectors and wires. *Figure 15* shows the construction details.

You can seal the driver flanges to the panel with silicone or, for greater ease of removal, with Mortite caulk. We trimmed the tweeter flange slightly to fit in the space between the mid-bass drivers. You can cut polyurethane acoustic foam, such as Sonex, or an open cell waffle pattern foam mattress pad, to fit the rear wall, over the crossover; and a foam septum fitted tightly to divide the cavity and provide high frequency isolation of the mid-bass drivers. High loft polyester fiber (Dacron) loosely fills the cavity. We mounted the Treble Coupler passive crossover securely within the enclosure on the rear wall, behind the acoustic foam.

SYMMETRICAL BASS. The bass enclosures also are fabricated from particle board and plywood veneer. We used scrap for interior bracing, and heavy wall plastic electrical conduit for the ducts. Mitering again is avoided by using hardwood pieces, later rounded. Fiberglass, or acoustic foam, lines at least three opposing interior surfaces. *Figure 16* shows the construction details for the bass enclosures.

BEFORE STARTING. The following discussion of construction contains cautions, many which we learned the hard way (losing part of a thumb and dermatitis). However, these cautions ought not to inspire trepidation.

Building accurate, square, air-tight boxes is surprisingly difficult. The key is the ability to cut pieces absolutely square and with precise dimensions.

Perhaps your most valuable invest-

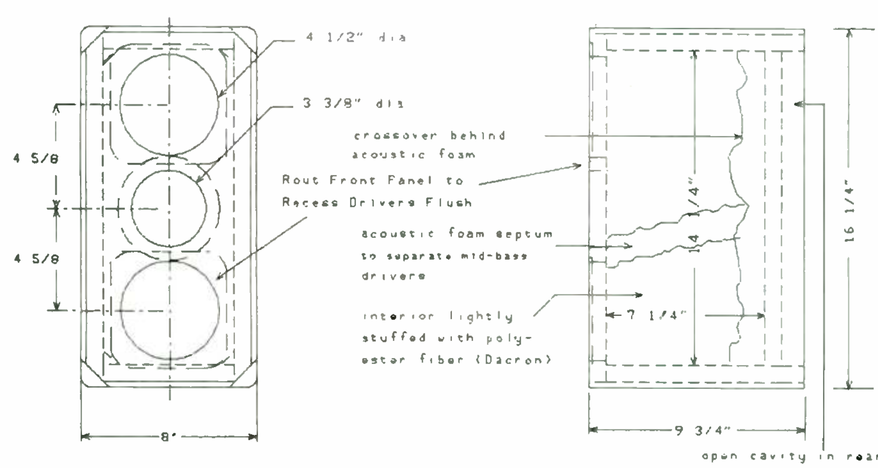

FIGURE 15: Symmetrical Satellite cabinet.

Erratum: In the Pedal Coupler schematic of *Part I, Figure 13*, and in the Parts List, resistor R15 at the "low output" should be 4.99kΩ.

ment of your time would be to tune up your saw. Start with a sharp carbide blade such as the Freud LU 85M and vibration damper discs. Make the arbor exactly square with the miter groove by precise measurement from the leading and trailing edges of the blade. Check the vertical squareness of the blade with the table and set the tilt stop. Fit your miter with a face board to extend the bearing surface. Square the fence and the miter to the blade. For safety, adjust the fence to diverge very, very slightly to relieve the tendency to kick back on ripping. A hold-down clamp on the miter and anti-kick back, hold-down wheels mounted on the fence are helpful. Proof your adjustments by trial cuts. Check for squareness and re-adjust. Be fussy! That done, go sharpen your planes, knives, and chisels sharper than ever before.

CUTTING THE PIECES. The veneer-faced plywoood pieces are more tolerant of small dimensional errors, so we begin with them. *Figure 18* is a layout for cutting the plywood veneer, using the minimum dimensions and with the grain direction proper. Note that the ¼-inch plywood sides, tops, and bottoms are larger than the corresponding particle board pieces, to recess the front panel.

Measure twice; cut once: to proof your saw further, cut the first few veneer pieces 1/16-inch oversize. With a table saw, the veneer side should be face down to reduce splintering. If all is straight, parallel, and square, re-cut those pieces to exact size. Make all your cuts of the same dimension at one setting. If you must set the saw a second time, use a previously cut piece rather than a ruler. *Do not* use the miter and the fence simultaneously. Doing so is hazardous. Instead, for multiple crosscutting of the same dimension, clamp a small block to the saw table front to set the proper dimension. Label the pieces in pencil on the back after each cut and stack identical pieces together to check for identity and squareness. Re-adjust the saw if needed.

Figures 17a and *17b* are the layouts for cutting the ¾-inch particle board. A medium density board called Novaply is available widely, but higher density particle boards (MD-44), which may be more satisfying to work with, are available in some areas. These pieces must be cut accurately; the integrity and strength of the boxes depend upon it. Wear a dust mask when cutting these urea formaldehyde bonded materials. If you make any small cutting errors, recut the pieces accurately, but slightly smaller. The boxes will perform as well with a slight reduction in volume.

If one side of the particle board is rougher, use it on the interior, and identify each piece on that side, including an "A" or "B" to distinguish pairs. Dry fit the pieces using clamps, to check for problems.

Cut a large number of glue blocks or cleats from softwood about ¾-inch square and perhaps 6″ long. Break (chamfer) the edges with sandpaper or a plane to allow a tight fit with the interior intersections.

Lay out and cut the driver and duct apertures in the front panels. A saber saw will do, but a fly cutter in a drill press is better for all but the bass driver apertures. Small notches in the apertures may be required to accommodate the driver leads and lugs of the satellites. Fit each satellite driver, trace around the rim, and rout the satellite front panels—about ¼-inch deep to recess the frame rims of the mid-bass drivers, and about ⅛-inch deep to receive the tweeter flange.

Note that the tweeter flange must be trimmed to fit between the mid-bass drivers. The final, flush front panel

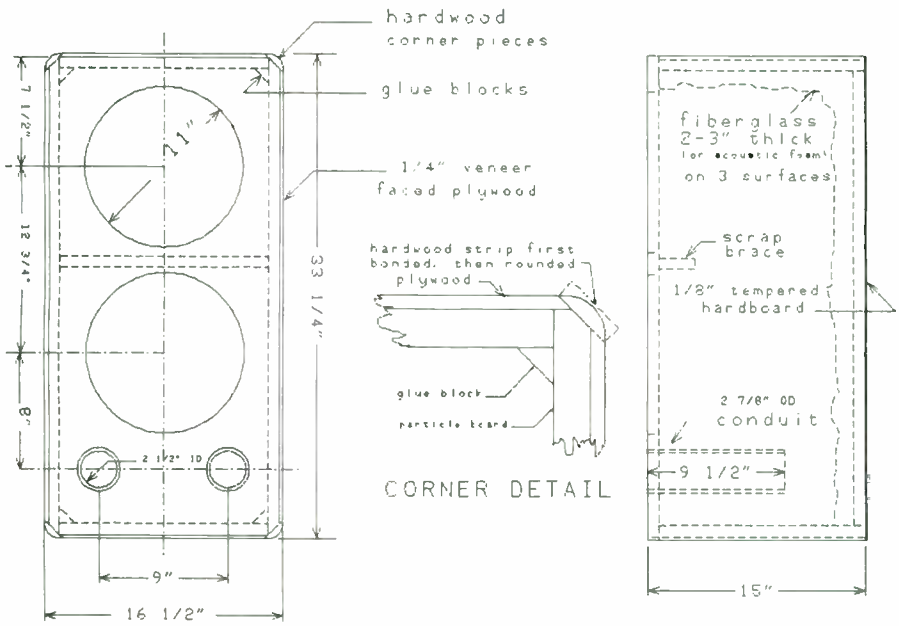

FIGURE 16: Symmetrical Bass cabinet.

minimizes reflection and diffraction of the higher frequencies. If you do not wish to rout, use a panel of plywood, hardboard, or dense carpet underlayment foam, ¼-inch thick, apertured to fit over the front panel snug to the driver rims. The tweeter will require a shim to become flush with the mid-bass rims; use a ring of hardboard.

Cut apertures for the terminal fittings in the back panels of both the satellite and bass boxes.

EPOXY CONSIDERATIONS. Before proceeding to the assembly, a few words about adhesives are in order. Most glues used in cabinet work are for wood rather than particle board, intended to be used with high clamping pressures, require swift assembly, are not capable of gap filling, and are chosen for reasons of economy. None of these apply here, for we are dealing with an awkward, slow assembly not amenable to high clamping pressures.

Particle board absorbs adhesive, and gap-filling qualities are required to insure against air leakage. Furthermore, a high strength adhesive is required because of the limited surface areas of the joints. Taken together, these factors virtually require epoxy adhesives.

Fortunately, all of the necessary materials, tools, and advice are available from Gougeon Brothers' West System. Other sources do not offer the completeness of Gougeon Brothers (see Sources, *Part I*).

We use West 105 resin with 205 fast hardener for adhesives, fillers, and coatings. The resin and hardener come with convenient dispenser pumps which ensure precise 5:1 proportions. We mix in plastic cups using tongue depressors and apply using polyethylene squeegees and throw-away acid brushes, all available from Gougeon Brothers. For epoxy coatings we add nothing; for epoxy adhesive, we add their #403 cotton fibers to the thoroughly mixed resin and hardener. We add a plastic coffee measure (a tablespoon) of cotton to each two shots of resin/hardener mix. The cotton fibers provide thickening for handling ease while allowing the epoxy to wet the joint surfaces thoroughly.

These epoxies appear to have cumulative effects upon skin. Carelessness can catch up with you years later in the form of permanent dermatitis problems. We never touch the stuff without wearing vinyl examination gloves. Those of fair complexion also should use a protective barrier cream. Epoxies essentially are odorless after mixing and modest usage with reasonable ventilation requires no respiratory precautions.

PREASSEMBLY. In our shop, we laminate the veneer to the particle board cabinet pieces before assembly, because we have the necessary clamping apparatus, and because it is difficult to provide the necessary evenly applied pressure. Even though the pressure is low in terms of pounds per square inch, the force required is great. We will take you through our technique first and then suggest an alternative which works.

Cut a 4-mil polyethylene roll into two-foot strips, the width of the roll. Locate an area of floor, preferably cement slab,

Continued on page 38

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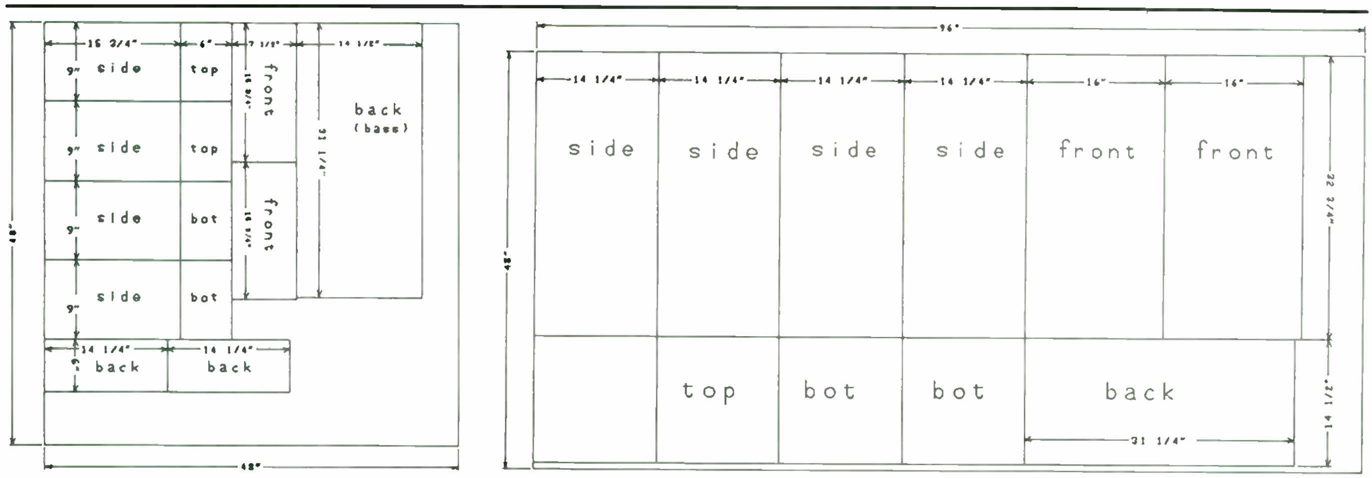

FIGURE 17: Layouts for cutting the particle board.

below a strong horizontal structure, such as a major beam (three or four overhead floor joists can be spanned with a temporary, short, stout beam tacked horizontally into place to the joists).

Find or make another stout beam about a yard long and set it aside. Find a pole, such as a pipe or 4 by 4 and cut it to equal the ceiling height minus the sum height of: the stack of pieces to be laminated, the set aside stout beam, a hydraulic jack, and the overhead beam. Alternatively, a screw jack floor post (available for under $20 at a builder's supply) may be used. Do a dry run assembly of everything without adhesive to assure that all will work. If the floor is uneven, first position a piece of excess particle board or the like.

Because they are smaller, do the satellites first. Lay down a polyethylene strip and a side piece, rough side (if any) down. Put on gloves and mix four shots (a shot is one resin and one hardener pump stroke) and then add two coffee measures of #403 cotton fibers. The desired consistency is a runny paste. Mix again, and use the squeegee to evenly spread about a quarter of the batch onto a particle board side piece, adding or subtracting epoxy as needed to get an even coat. Place the veneer faced plywood (veneer up) and align three edges, leaving one edge proud slightly more than ¾-inch. We use two temporary insulation staples placed near the top and bottom edges to insure against slippage.

Fold the polyethylene strip over the veneer face and stack the next side, taking care not to disturb the alignment of the first. Do the polyethylene folding, adhesive spreading, and veneer alignment for the remaining three sides. Now do the same for the tops, side by side, and stack the bottoms over the tops. Check to see that nothing has slipped in the slippery goo. Lay the stout beam over the whole stack, place the jack, get a friend to hold the pole between the jack and the overhead beam, and apply a little force. Check again for slippage and "tunk" with a rubber mallet until all is as it should be. Add some more force and check again.

At room temperature you have about 20-30 minutes of pot life in the mixing cup, but about an hour when spread out thinly on the panels. Add all the force you think you can without endangering your home. Even a ton of force distributed over the area of the bass cabinet sides equals only a few psi.

Clean up, and go away for 24 hours. Do not peek or check. The used mixing cup will show all you need to know about the status of the cure. Be patient. Wait until the thinnest film of the mixing cup is brittle before exploring the stack (thin sections cure last).

The next day, disassemble the jack arrangement and remove the temporary staples (Osborne sells a staple remover for the upholstery trade). Inspect. If after all precautions, something slipped, punt, for you will not disassemble the mistake.

Use a triangular scraper and chisel to remove any excess epoxy from only that edge beyond which the plywood protrudes. Any surface which will abut another to form a joint should be cleaned of any excess epoxy. Elsewhere, excess epoxy will not interfere. Follow the same steps for the bass cabinets.

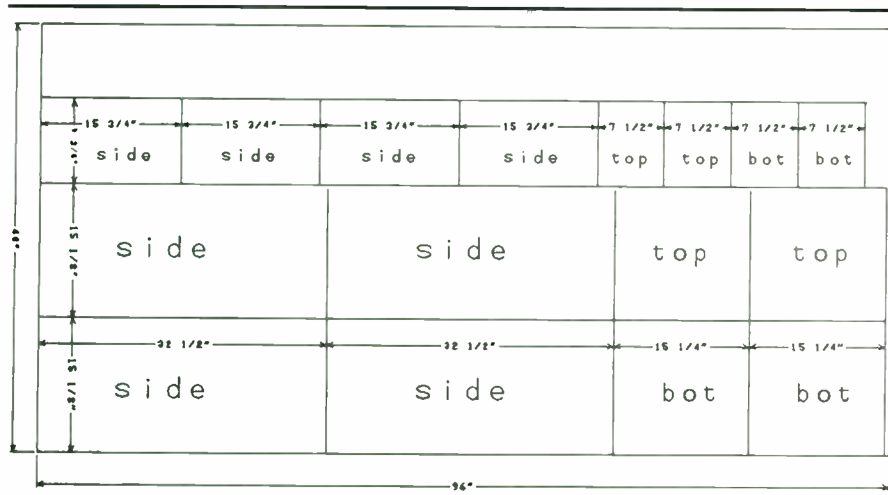

FIGURE 18: Layout for cutting the veneer.

ANOTHER METHOD. If you are unable to use the stack-and-jack technique, we have found a satisfactory alternative, using epoxy with ½ by 9⁄16″ insulation staples, spaced about six inches apart. The trick here is to staple over wooden popsicle sticks (or tongue depressors, split lengthwise, or some similar strip material, narrow enough to be embraced by the staple). These greatly assist later in removing the staples and prevent the staple crowns from denting the veneer. This method leaves many small holes to fill, but the relatively low pressure succeeds because epoxy adhesives are tolerant of low bonding pressures. We also use this method to laminate the plywood veneer to assembled particle board boxes.

ASSEMBLY. Because they are easier to handle, first assemble the satellites. Find four one-inch blocks to support the back panel above the work surface. Keeping "A" and "B" pieces separate, position the sides, and top and bottom about the elevated back, place the clamps and check for a tight fit. Adjust if necessary. Check that the front panel will fit well. Repeat for the other satellite and disassemble.

Glove up, mix four shots, and add cotton as before. Use a throw-away acid brush to apply the adhesive generously to both the abutting surfaces. Do not deal with the front panel at this time. Assemble and clamp. Use the rubber mallet to "tunk" things around until the fit is perfect. When all is well, use the brush to form a fillet on the interior of each joint intersection to ensure a tight seal. Coat two adjacent sides of the glue blocks (cleats) and push them into the intersections to add rigidity. Thin nails in pre-drilled holes may help hold the blocks in place until cured. Daub a bit more epoxy to fillet the blocks.

We then place a polyethylene sheet (to prevent adhesion) over the open front and fit the front panel to help to square things up. Trim that panel to fit snugly with the polyethylene, but without spreading the box joints. Proceed with the other satellite enclosure and set them aside to cure for 24 hours or more. After curing, sand off any lumps of epoxy on the veneer faces.

Follow the same steps for the bass cabinets. The scrap stiffener pieces need only to be epoxied in place with generous fillets and no fasteners. Again, fit the front panel with a polyethylene sheet.

If you cannot beg, borrow, or steal enough clamps, there is an alternative. You can drive square-drive 1¼-inch particle board screws if shank (but not pilot) holes are pre-bored before spreading epoxy. After curing, those screws which show or get in the way can be removed because, unlike slot heads or even Phillips heads, square-drive screws do not slip. Phillips head wallboard screws are less satisfactory, but will work.

CORNER TRIM. Now you are ready to bevel cut the corners. We have a large industrial table saw which allows us to slide the completed boxes against the fence, to cut a 45° chamfer or bevel, to receive hardwood corner pieces. Talcum powder helps the box slide. Those with radial arm saws may have an easier job of this than we do. Those without either will have to resort to the genuine pleasure of using hand tools. Cut a bevel which removes a small bit of the plywood, to provide a surface about 1¼-inch wide on the bass enclosures and about one inch on the satellites (see the detail in *Fig. 16*). Rip a ½-inch hardwood board, which matches the plywood veneer, slightly wider than these widths.

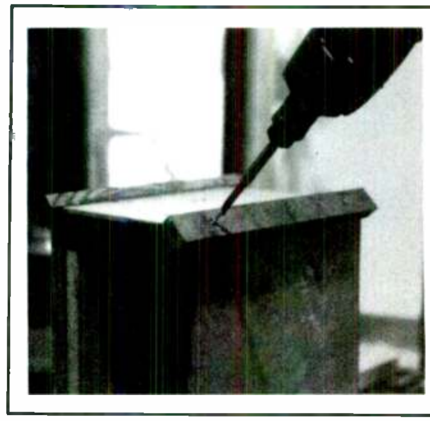

PHOTO 3: Square-drive screws can substitute for clamps until epoxy cures.

PHOTO 4: Round the corner pieces with a sharp plane.

Cut each piece about an inch longer than the front-to-back dimensions of the cabinets.

We use "C" clamps to hold these pieces in place. The recess in the back of the satellites provides an edge for the clamps similar to the open front. The rear of the bass cabinets presents a problem which band clamps can solve. We bore a shank hole about 2" from the rear end of the pieces and use a square-drive screw, which is removed later and the hole filled. Indeed, the whole job can be done with screws, as is shown in *Photo* 3, but that is a lot of holes to fill. If you are short of clamps, do only a few pieces at a time. Use epoxy/cotton and an acid brush.

Once cured and after any screws are backed out, you must shape the hardwood pieces. We use a Japanese Ryoba saw to cut away the overhanging edges to form a truncated section and then use a sharp plane to form the radius, as shown in *Photo 4*. An advantage of a Ryoba is the low set of the teeth which avoids scarring the veneer. We complete the shaping with an orbital sander and #80 open coat paper. Cut the rear protruding ends flush with the back, but in front leave them until later.

FRONT BAFFLE. Next, fit the apertured front panels; trim the corners to fit within the recess formed by the projecting plywood and corner pieces. Clean up any excess epoxy. When all fits well, cut the ducts, four 9½-inch pieces, to fit the duct apertures snugly. Use epoxy/cotton to bond them to the bass front baffles, flush with the outside face. Use generous fillets on the interior side and allow to cure. Then apply epoxy/cotton to the front edges of the box and to the inside of the projecting plywood. To apply clamping force, we use square-drive screws in the front baffle's corners and middle of the longer sides, which do not need to be removed later. When all is cured, trim the projecting plywood and corner pieces flush with the baffle.

PHOTO 5: Roller coating with epoxy.

FINISHING. Fill any problem areas on the fronts and backs with a mixture of epoxy and micro-balloons, an epoxy filler additive available from Gougeon, which bonds tenaciously and you can easily sand flush after curing. Use colored wood filler in any holes or dents in the veneer and corner pieces; allow to cure

PHOTO 6: Acoustic foam separates the mid-bass drivers.

and sand the veneer surfaces with an orbital sander and #80 paper. Epoxy smears only need to be level. Check and fill any imperfections. Dust everything and clean up the finishing room.

Glove up and mix four shots of epoxy/hardener without any additives, and pour it into a plastic roller tray. Using a 3½-inch roller, coat all surfaces with the epoxy, as shown in *Photo 5* (yes, the vinyl gloves are on). Soon you will see bubbles seemingly spoiling your shining coat of epoxy. Gently use the tip of a throw-away brush to break them, but it is not too important that all bubbles are caught. Knock off the wet roller before curing or it will be devilish to remove. You can preserve the roller for a day in a plastic bag in the freezer. Allow the residue in the plastic tray to harden; it will do no harm and can be broken out if desired.

Allow for a full 24-hour cure. Wash the epoxy surfaces with a wet cloth to remove "hardener bloom," a surface exudate which clogs sandpaper. Now sand with the orbital sander and #80 paper to dull and smooth the surfaces and remove the remaining bubbles. Use a dust mask; why inhale the stuff?

Next, we paint the back and front panels with flat black polyurethane enamel using another 3″ roller. Any which gets on the veneer can be wiped off or removed later with sandpaper. Finally, apply one or more coats of satin polyurethane varnish, including a coat over the black paint, paying attention to the advice on the can, and sit back to admire your handiwork.

FINAL ASSEMBLY. The Treble Coupler passive crossover is on two circuit boards which slip through the mid-bass driver apertures. Fit the boards with suitable length leads of very flexible, high purity copper, polarity coded #18 AWG wire, such as Apature. Secure the boards to the interior of the back panel with a viscous, tacky adhesive such as Goop, or screws, or both. The input leads are connected to the terminal fittings, which are then installed in the appropriate apertures. Place acoustic foam or fiberglass over the crossover on the back panel and secure it. Fit a piece of foam as wide as the interior and as long as the interior depth to the rear wall foam, to form a septum to divide the in-

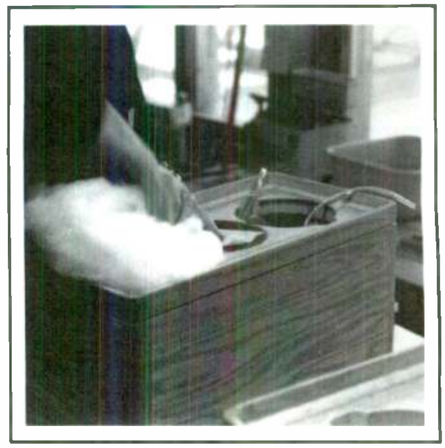

PHOTO 7: Stuff polyester fiber lightly into the cavity.

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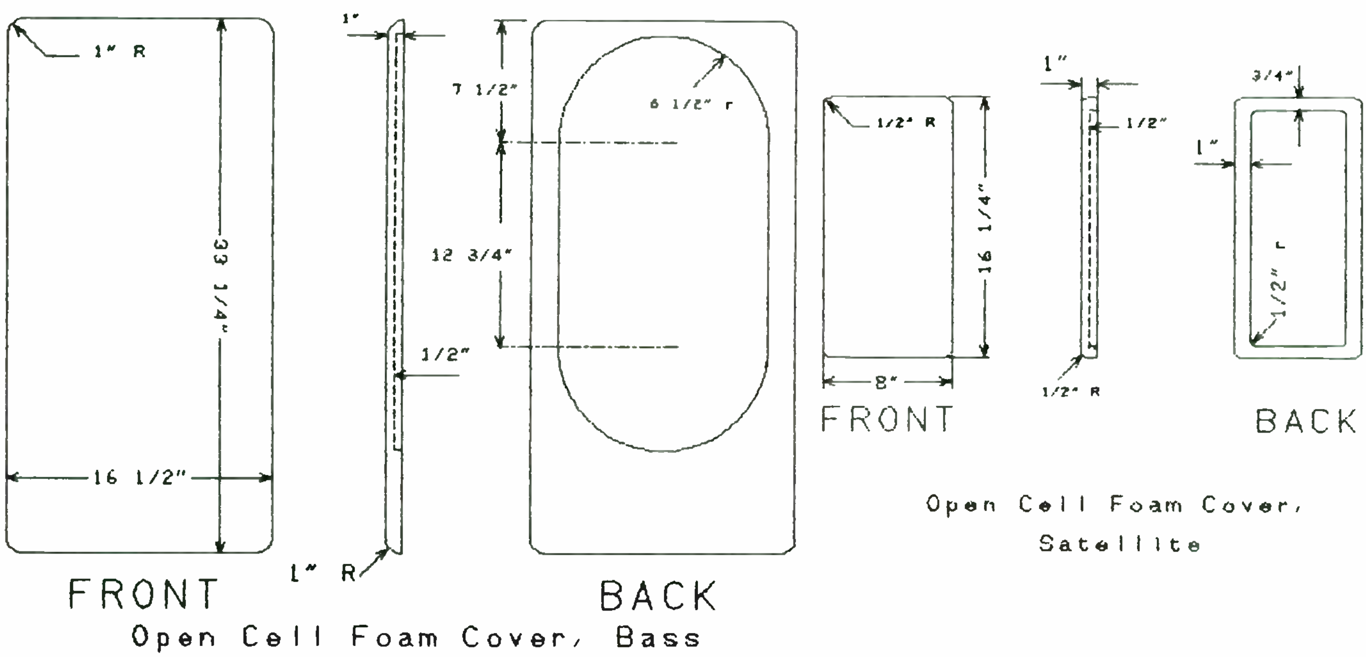

FIGURE 19: Foam grille covers.

terior, which separates the mid-bass drivers at high frequencies (shown in *Photo 6*). The remaining interior volume is stuffed lightly with high loft polyester fiber (Dacron) pillow stuffing (*Photo 7*).

We routed the front panel to thereby bring all driver rims flush with the panel. Connect the drivers to the appropriate leads and fit into the apertures. Drill lead holes for screws and apply a silicone (or Mortite) caulk bead. Place the drivers and tighten the screws gently, sequentially. Check air tightness by gently pushing in one cone and observe the other moving outwardly and remaining for a bit. You can work caulk into any gaps between the rim and the routed panel. Wait until the caulk cures before attempting to remove any excess.

Install terminal fittings in the bass enclosure back panels with #14 or #16 AWG polarity coded, flexible wire leads, long enough to reach through the front panel driver apertures. We use Monster Cable XP, #16.

Staple acoustic foam or fiberglass to at least three opposing interior surfaces, such as top, back, and left side. Connect the drivers in parallel to the leads and apply a bead of caulk. Place the drivers and rotate, if necessary, for appearance. Drive black wallboard screws gently (without lead holes) and then tighten sequentially. Again, wait until caulk cures before removing excess.

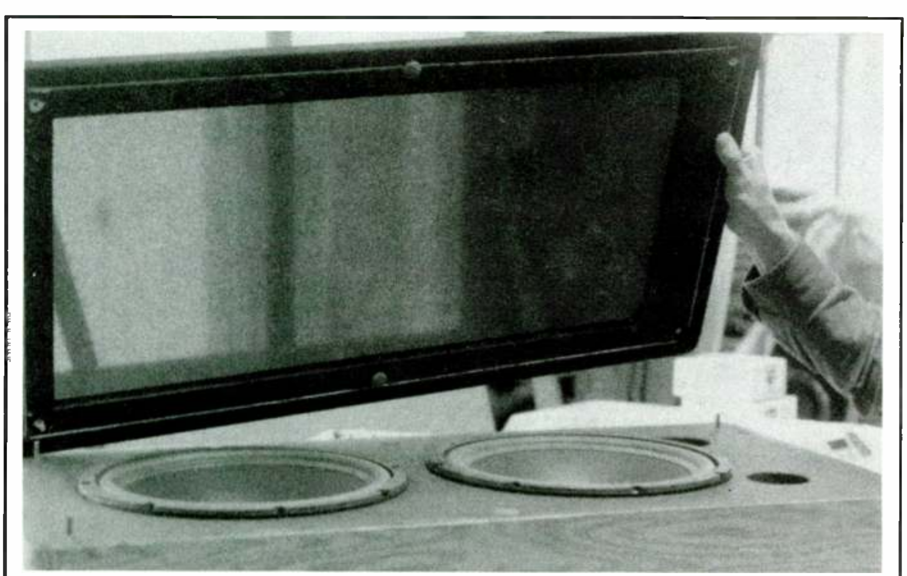

PHOTO 8: Cut a groove into the back faces of the bass grille frames to hold the fabric with screen bead.

GRILLE CRITERIA. For reproduction of frequencies above a few hundred hertz, we have found problems associated with grille cloths and frame structures. After an initial period of acceptance, we find that we tend to become dissatisfied and remove coverings, yet they are necessary for protection, decorative reasons, and seem to be beneficial psychologically.

Unfortunately, optical opacity and acoustical opacity seem to go hand in hand. According to our measurements, fabrics, screening meshes, and most foams affect performance by measurable ripples and sags in the response curves. More importantly, we find that we can hear differences. Fabrics such as open weave double knit speaker cloth cause discernable high frequency loss. Meshes often cause interior reflections. Both meshes and fabrics usually require a frame or other supporting structure, which causes significant audible and measurable diffraction problems, reflections, and cavity resonance problems.

Coarse, reticulated, open cell polyurethane foam offers the least acoustic interference. Proper foams are largely resistive, whereas fabrics are reactive. With certain foams, we measure about 0.5dB loss, evenly throughout the spectrum, whereas double knit grille cloth shows a loss which increases with frequency and which can exceed 3dB at

PHOTO 9: The bass grille frame is held with pins and velcro fasteners.

20kHz. This is not merely an academic exercise; we can hear a loss of imaging quality with cloth grille covers. Radio Shack and other sources sell a suitable foam which may have a pattern of sculptured cubes on one face. Humidifier belts of suitable open cell, reticulated foam are another source for small quantities and have a slightly more open structure than the Radio Shack foam.

We plan to make available custom made, charcoal brown, proper foam grilles for both the satellite and bass modules available, for individual builders. They will be essentially the grilles shown in *Fig. 19*.

The best appearance is had with double knit grille cloth. Neither the cloth, nor the frame to support the cloth, offer significant acoustic problems with the bass speakers. We construct frames of ¾ by 1½-inch softwood or particle board, shaped to match the cabinet periphery, with rounded outer edges. You may find appropriate sized canvas stretcher frame parts in an art supply store.

Photo 9 shows the frame, supported to the front panel with pins. Six 15/64-inch holes are drilled through the frame into the cabinet front panel, but not deep enough to penetrate the enclosure. Next enlarge the frame holes to ¼ inch. We dip ¼ by 1½-inch metal rods or nails in glue, drive into the cabinet holes through the frame holes, and grind any excess rod flush with the frame. A ⅛-inch groove (saw kerf), 5/32-inch deep, is cut about one inch in from the edge, on the back face of the frames (see *Photo 8*).

Then we stretch Radio Shack double knit cloth slightly, secure it with a few staples and push a window screen vinyl bead, 7/64-inch diameter, tensioned slightly, into the groove.[7] We cut away the excess cloth and remove the staples.

Assemble a few small pieces of Velcro fastener, with pressure sensitive adhesive backs, into mating pairs and position strategically on the frame. Push the completed frame onto the pins. The Velcro mates are transferred to the cabinet front, retaining the frame tightly against the cabinet.

The satellites are much more affected by the use of grille cloth, but we chose to consider the appearance more important. The frame normally needed can be avoided. Using a table saw, we cut a ⅛ by 5/32-inch groove, about one inch from the front edge, all the way around the cabinet (see *Photo 7*). Cut open cell polyurethane foam into ½ by ¾-inch strips and bond to the front panel perimeter. Trim the corners to match the round corners of the cabinet.

Humidifier belt foam or Radio Shack foam are sufficiently open, but weatherstrip foam tapes are not satisfactory. Stretch the fabric slightly and hold with tape, then press the vinyl screen bead, tensioned slightly, into the groove. Do not stretch the cloth so tightly to com-

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7. Gitto, Carmine, "Grille-Less Grille," *SB* 1/87, p. 40.

press the foam significantly, or it will become reflective; also, the cloth will be too close to the mid-bass cone surrounds. Cut away the excess cloth beyond the bead.

PLACING THE SYSTEM. The SWAN IV system should be placed at least three feet away from rear walls and preferably farther from side walls to obtain the response curves shown and to realize the full imaging potential of the system. Closer wall spacing will create bass overemphasis and mid-bass response anomalies, audible as thickened voices, and an overall degradation in imaging and depth reproduction.

If you must position the system closer to rear walls than we recommend, wall treatment with Sonex acoustic foam or waffle pattern mattress pads may be used to eliminate wall reflections above a few hundred hertz. Low bass emphasis can be controlled by reducing bass boost using the alternate values for R8 and R11 in the Pedal Coupler. Rooms without carpeting cause problems, as does hard-surface furniture in front of the speakers.

In the average size home listening room, the best listening area will be at about one to one and a half times the speaker separation distance. For starters, we recommend the Symmetrical Bass speakers be placed facing directly outward. The Symmetrical Satellite speakers should be placed on the forward edge of the bass speakers and toed in, toward opposite ends of the desired coverage area. Thus, the projected axes generally will cross just in front of your favorite listening chair. This alignment puts the listener well within 15° of the axis and aims sound away from side walls to minimize early reflections.

Use this as a starting point to determine the best orientation for your particular listening environment. Your ears are better than theory—experiment! Use the benefit of satellite enclosures which can be aimed independently of the bass enclosures.

The Symmetrical Bass speaker enclosures sonically and visually complement the satellites to provide a full-range system having tight, impressively deep bass. The modestly sized bass enclosures provide an appropriate height stand for the Symmetrical Satellite speakers. The paired bass drivers are placed vertically close together near the top of the enclosure cabinet. The effect of "floor dip" due to a virtual reflection from the floor is diminished because the two drivers are relatively far from the floor and because they are at different heights above the floor. Virtual reflection thereby is diffused. The absence of sharp dips and peaks in low bass is particularly noticeable with organ music and sustained deep pedal.

CRITICAL LISTENING. The SWAN IV speaker system comes as close as we can in a home listening environment to realize the dynamic range potential of digital recordings. It provides detailed image information, including definitive and stable lateral localization, depth, and hall ambience. The relatively broad soundfield allows us freedom from confinement to a unique predetermined listening location. It imposes no limited "sweet spot" upon the room, yet it controls, by polar response characteristics, undesired wall reflections.

We have installed the SWAN IV in rooms as small as 2,000 ft.^{3} and as large as over 20,000 ft.^{3}. In the small room, we thought some wall treatment was needed. Because the large room is wood and glass, and lacks carpeting and furnishings, placement far from the walls was required for good image detail. The results for large organs or large scale opera are delightful. Normal rooms (2,500 to 5,000 ft.^{3}) with carpeting and furnishings offer less problems, but benefit from placement experimentation. Unlike many fine systems, it sounds about as well to a listener standing or seated.

We believe that because of the acoustic coincidence of the tweeter and driver pair, the nearly ideal phasing, minimal time delay, and the attentions paid to seemingly small details, the sounds from the several drivers meld into a cohesive sound source. It "clicks in" to present a focused image. The listener's

APPENDIX

System Exercises

In the early days of interest in high fidelity, a recording entitled "Speed the Parting Guest" exercised our horn loaded monaural monsters while annoying the neighbors. Listed below are some CDs which we find useful in evaluating systems and listening environments.

For deep, deep bass we enjoy the following:

Dupre and Rheinberger—the organ at Royal Albert Hall (Telarc 80136)

Saint Saens No.3—the ultimate bass stress test (Telarc 80051)

"Time Warp"—Erich shakes the house (Telarc 80106)

Widor—on a truly great Cavaill-82—Coll organ (Philips 410 054)

"Encores..."—house shaking bass from Symphony Hall (Telarc 80104)

"1812"—Erich's cannon could damage things (Telarc 80041)

Ruffatti organ in San Francisco (Telarc 80097)

For individuality of instruments or voices:

"Appalachian Spring"—Best for focused image (Pro Arte 140)

Couperin—perfect harpsichord (Denon 7004)

Scarlatti—a fine harpsichord (Archiv 419 632)

Oedipus Rex—spoken, shouted, and sung (Orfeo 071 831)

Cantaloube—Kiri soars over instruments (London 410 004, & 411 730)

Cantaloube—Flicka flies also (CBS MK37299)

Chopin—finest piano ever (Telarc 80117)

Schubert—perfect voice and perfect piano (Denon 7240)

"Water Music"—a superb piano and pianist (Delos 3006)

For room ambience and orchestral image focus:

Elgar—ambience and string warmth plus deep bass (Nimbus 5008)

Mahler #1—Favorite for spacial image and depth (London 411 731)

Mahler #3—Best miking yet—you are there (Denon 7828-9)

Esther—Incredible hall presence (L'Oiseau 414 423)

Cosi fan tutte—same virtues as 414 423 (L'Oiseau 414 316)

Der Messias as it was and ever should be (Teldec 8.42349)

For large scale, powerful music:

Berlioz' Faust—Solti's thrilling version (London 414 680)

Verdi's Requiem (London 411 944)

Berlioz' Requiem (Telarc 80109)

Erich and Telarc at their best (Telarc 80115) attention is not drawn to the speaker enclosures. We find that we listen to the music, not to the speakers. The image or soundstage is not affected much as the listener moves within the room. We think the SWAN IV system passes the severe test of long term listening at fairly high levels without listener fatigue.

In critical listening to the SWAN IV speaker system in home listening rooms, we continue to be impressed with subtleties buried in recordings, new and old. The bass is unnoticeable until called upon, and then it avoids sounding like a loudspeaker in a box. Voices and strings are not colored by the bass, but the bass allows the pedal of large organs to be felt as well as heard. We can understand the words of grand opera and large scale choral music, as the voices are reproduced as separate entities. The sibilants are clear, but not emphasized. The ambience of the recording site is audible, which we think adds to the clarity. Feet clump, clothing rustles, artists breathe and hum, and music stands fall. Pianos and harpsichords are not merely reproduced, they seem to be recreated in the listening room along with the sounds of the human presence of the artists. The "Telarc" drum of the Cincinnati Orchestra yields a physical sensation not limited to ears alone.

RECOMMENDED TOOLS

Table saw, (preferred), or a radial arm saw
Low angle block plane (sharp!)
Hand saw (sharp, preferably a Japanese 210mm Ryoba)
Saber saw
Orbital sander
Clamps and more clamps (at least six Pony pipe or bar clamps with 24" and 36" pipes and pads, and at least six 4" "C" clamps. Four webbing or strap clamps are most useful)
Power screw driver (is very helpful)
Caulking gun with clear silicone
Stapler for 1/2" insulation staples
2-foot framing square
Carpenter's compass
White rubber mallet
Epoxy handling materials including vinyl examination gloves, tongue depressors, acid brushes, 4" paint roller and polyurethane roller covers, and 4 mil polyethylene sheet.
Dust masks
Staple remover
Miscellaneous screw drivers, wood rasp, hammers, chisel, knives, scissors, wire strippers, crimpers, soldering equipment, pliers, sanding blocks, triangular scraper, putty knife.

Addendum

We are pleased to talk with people about the system which we have perfected. However, we have received numerous calls asking about details different from our design. Because we have not tried those modifications, we are not qualified to discuss the possible results. We do know that the performance of our design is sensitive to seemingly trivial changes or substitutions.

The Precision TA305F bass drivers temporarily are unavailable. Precision assures us that they again will be produced. The parameters of the TA305F are unique and allowed us to use a cabinet as small as 100 liters. Several speaker manufacturers are modifying 12" drivers in an effort to meet our requirements. We will keep you advised of the availability of suitable drivers which will fit the cabinet with no modifications.

With the cooperation of several distributors, we have obtained, measured, installed and evaluated a number of 10" drivers. More are under evaluation. The use of 10" drivers lessens the maximum SPL at very low frequencies by 3-5dB. While this has little disadvantage with most music, some deep bass organ notes tend to cause over excursion. Accordingly, we have modified the Pedal Coupler crossover/equalizer sold by Swan's Speaker Systems. The modified version has less equalizer boost to accommodate the lesser volume throw of 10" drivers.

To use the following 10" drivers, the front panel hole diameters must be reduced to 9"

Continued on page 65

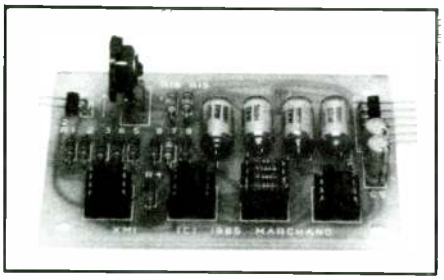
Kit Report

Marchand Crossover Model XM1

The Marchand Electronic Crossover kit provides one channel of an active fourth-order, two-way Linkwitz-Riley (all-pass) crossover and comes in two forms. The XM1-K version includes all parts with an assembly manual for $17.95 per channel, while the XM1-A version is fully assembled and costs $21.95 per channel. The kit requires a ±15V power supply, which is obviously not included at these prices.

For this review I received one assembled kit and one parts kit. Assembling the parts kit is straightforward, but I managed to foul it up by creating two solder bridges. My vision is no longer the best, so I didn't spot the shorts. In the time it took to read zero volts on the negative supply and react by pulling the plug, I had burned out the power supply. Because of the possibility of such events and the minimal price differential, I would opt for the assembled board.

The crossover is realized by a state-variable filter, so that the low-pass and the high-pass outputs are taken from different points in the same circuit. The crossover frequency is determined by the common value of four resistors which are mounted on a plug-in module. My boards came with 100Hz modules, but I wanted a 200Hz crossover point. Since suitable plugs are sold by Radio Shack and I had the correct resistors on hand, I simply made my own 200Hz modules. The assembly manual gives the formula for calculating the resistor value.



To test the completed Marchand crossover, I used it in place of the low-frequency section of my own crossover system and listened for any differences. I heard absolutely none. As far as I am concerned, the Marchand network is equal to my sixth-order all-pass networks, built with Jung rumble filter boards.

Following my article, "Passive Crossover Networks, *Part III*" (*SB* 3/85), you could build a stereo fourth-order, all-pass crossover using Jung boards, but four are required, whereas you need only two Marchand boards. The complete crossover with a power supply would cost about $150 using Jung boards, and less than $100 with the Marchand boards.

On the other hand, the Jung boards have a built-in gain adjustment, while the Marchand boards do not. If you have amplifiers with gain controls this is no problem, but if you need to manage sensitivity matching at the crossover, then the Jung boards may be preferable. Also, the Marchand boards are not suitable for lower orders or different crossover types, while the Jung boards are.

If you want a high quality fourth-order, all-pass active crossover network and have no need for a sensitivity matching capability, the Marchand crossover boards are an excellent value. As far as I know, they have no competition at their price.

Marchand Electronic Crossover, available from: Marchand Electronics Inc., 1334 Robin Hood Lane, Webster, NY 14580, (716) 872-5578.

Robert M. Bullock, III
Contributing Editor

Phil Marchand comments:

I thank Mr. Bullock for a favorable review of our XM1 electronic crossover network. Level control can be easily accomplished by connecting a potentiometer at the output of the crossover. Lower order crossover slopes (first-, second- and third-order) can be achieved by changing some components. Details are provided on an application sheet we now include with the boards. But I think that all things considered the fourth-order slope is best. We now also offer the XM6 crossover network, which has level controls and front panel control of the crossover frequency.

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David Davenport reporting in SPEAKER BUILDER MAGAZINE issue 2/88.

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Fast Reply #JC563

Pox Humana

Loose Lips Sink Ships

By Dick Pierce

LET'S BEGIN OUR SERIES on hi-fi foibles by taking a look at some of the problems dealers and customers face when dealing with manufacturers. Back in the mid-seventies, there was rumored to be a most amazing loudspeaker designer working out of the Midwest (Indianapolis), whom we will call Roy Zork. That he used some novel and questionable design techniques may be something of an understatement. For example, his solution to the problem of cone breakup, which is, essentially, standing-wave patterns on the cone, was to slit the cone with a razor blade. Interesting, if somewhat misguided.

Well, one day Roy shows up in the Boston area, and starts haunting our store; listening, asking questions, probing, and so forth. All in all, not a bad fellow, although somewhat of a nuisance. It seems he's decided to design yet another loudspeaker. "Gee, Roy," we say, "that's nice."

Several months later we get a call from Roy. He asks if we would like to hear his loudspeaker. Sure, why not? It's winter and nobody is coming into the store. Roy says he'll be right down. Fifteen minutes later, in ambles Roy, carrying his speakers. The first thing I do is rip off the grille cloth and comment, "But Roy, the cone is in one piece." Roy was not amused.

We sit down and listen to the speakers. Not bad—not great, but quite inoffensive. The one obvious drawback is they have no bass. Trying to be as diplomatic as possible, we say, "But Roy, your speakers have absolutely no bass."

Surprisingly, Roy agrees, "Yeah, I know, I can't quite figure out why. But they are reasonably efficient."

That they were, and that was the obvious clue to the problem. It seems Roy had selected a woofer that was far too damped, electro-magnetically, for the enclosure he had designed. Thus, while the large magnet on the woofer contributed to a high efficiency, it also meant the bass was far too tightly controlled.

To me the answer was obvious. If you want bass, and you have decided on a certain enclosure size and a certain woofer, you get a choice: either low efficiency and bass, or efficiency and little bass. Opting for the former, my choice would have been to save money and order the woofer with a smaller magnet. Roy would hear none of that, however. He wanted a big magnet to properly control the woofer, which was exactly his problem.

The discussion went on and on; Roy dismissed anything about efficiency/bandwidth/power-handling tradeoffs. Finally, in frustration, I said, "Well, Roy, why don't you just stick an eight ohm, 50 watt resistor in series with the whole damn speaker. That'll give you some bass!" Feeling even more bold, I said, "In fact, why don't you put a big switch in there and convince customers they have a "Q" switch?"

Well, everybody laughed. Even Roy laughed a little bit. We closed up and went home.

A year or so later, when I was working at another store, in marches the sales rep for Zork Loudspeakers. The speakers looked the same and of course, had no bass. When that objection was raised, the rep said, "No problem, Zork has developed this revolutionary new method for increasing bass response. We have added a Q-control switch." He promptly threw the toggle switch at the back of the enclosure and, obligingly, the efficiency dropped by half and the bass came back.

I turned to the rep and the store manager and said, "I bet you $1,000 that switch is connected to an eight ohm, 50 watt resistor." They looked at me incredulously. I just said to check it out.

About half an hour later, the manager and the rep came running into my lab yelling "Dick, look. You were right."

Sure enough, Roy had taken any advantage that his big magnet had provided and thrown it out the window with a big resistor. I figured at the time the retail price of the speaker could easily have been reduced by $100 a pair by not using the resistor and having the right sized magnet to begin with. And the speaker, in the "High-Q" position, was not a bad loudspeaker—not a great one, but quite inoffensive. Although it did have reasonable bass.

New Software

IBM Programs

Driver Evaluation and Passive Crossovers

by G.R. Koonce
Contributing Editor

This article describes programs on two 5¼-inch, 360K format computer disks, distributed by Old Colony, which I designed for the home constructor. These programs, which basically cover driver evaluations and passive crossover design, are designed to be used with any IBM compatible computer.

They contain no graphics, use a monochrome display, but will use the coprocessor if your computer has one. I've made every effort to allow speaker builders, not just computer experts, to use these programs.

The basic computer requirements are an IBM compatible machine, running DOS 2.0 or later. I believe 256K of memory would be sufficient, but have not verified this; an 80-column display is required. You need a printer if you want hard-copy output.

The data entry has been optimized for those who work with measured driver parameters, not files of catalog data. In my experience, catalog data is not representative of the individual drivers which I purchased. I therefore recommend that you use actual test data, but catalog data certainly can be entered into the programs if that is all you have to work with.

The Basics

All the major programs on these disks are compiled in QuickBASIC 3.0 from Microsoft. If your computer does not have a coprocessor, you must tell the software this. The following line, typed at the DOS prompt, will accomplish this:

```
SET NO87=Use of coprocessor suppressed
```

If you plan to run these programs regularly then I recommend you enter this line in your AUTOEXEC.BAT file.

If you are a novice and confused by all this, then at the DOS prompt simply type: NOCOPRSR <Enter>

The batch file NOCOPRSR.BAT will install the needed line in the computer's environment. You must do this each time you start your computer *before* you try to run any program.

You will note that the title page for each program tells you the source of the equations used in the program. This permits you to understand just how the computations are being made. Also, each program allows you to enter a System Name. This is optional and can be bypassed by pressing <Enter>, but I recommend that you type in a simple name. Then, all charts and hard copy you produce will include this name and the date, for a reference aid. With a computer you can generate more output data than you will possibly be able to associate with the proper system by memory alone! For this same reason I recommend that if your computer does not have a real-time clock you enter the proper date when prompted by DOS.

Some programs will operate on two drivers simultaneouly; these allow an optional entry to identify each, for example, "Unit A," and so on. Again, I recommend you enter such identification.

All the programs are menu driven rather than command driven. Command driven programs can be faster if you use them regularly, but for occasional use you spend most of your time looking up the correct commands. The menu offerings have longer descriptive names, in terms understood by those familiar with speaker system design.

The programs give complete descriptions of required data entries including the units expected. Sometimes you are offered an option of input units, so observe the "boxed" information displayed for each data entry. The program checks data entries for reasonableness; if outside of reasonable bounds (where the program might "lock-up" or produce severe error) you will receive an error message telling you the acceptable bounds. Simply re-enter the data inside of the indicated bounds. If the correct data for your driver is outside these bounds then the program cannot be used for that driver. You may have to enter false "in bound" data to get the program to proceed to a point where you are permitted to exit.

Data Output

All data outputs contain as full a description, including units, as is permitted by the available screen space. Also all the data you enter is included on the output screens so you do not have to independently record what you enter. Each output screen represents a complete record of the computations involved. At times you may find a chart displays only a few entries, or possibly, no entries. This is because "unreasonable" output data is suppressed. For example on port duct design, only ducts between 1–8 inches in diameter and 1–20 inches long are displayed.

Most menus do not offer any option to print out data. Instead the output has been arranged in a series of single page displays. You may print any page by holding down the shift key and pushing the PRT SC key (some machines do not require the shift). Most menus do offer the option of "advancing the paper on the printer." This allows you to advance the printer to the top of a new page when desired.

Note that all of the major programs require the run-time module BRUN3087. EXE (copyright by Microsoft Corporation) to be on the same disk (and directory). I compiled the programs this way to reduce the size of the overall execute code. If you copy any program to another disk you must also copy BRUN3087.EXE or tell the computer where it can find the program via suitable PATH command.

MS-DOS is a registered trademark of the Microsoft Corp.

IBM is a registered trademark of the International Business Machines Corp.

I can never remember the name of all the programs on a disk and hate to have to call a directory and try to figure out which program is which. If you are a novice computer user or otherwise get lost on either disk, simply type at the DOS prompt: MENU <Enter>

The program MENU.COM will display a list of the available programs, a summary of what each one does and how to call each program.

Each disk contains a README.DOC file containing information about the programs on that disk and how to run the disk. Some further details not included in this article will be found in these files. To produce a printed version of these files use one of the following methods:

X:print Y:readme.doc <Enter>; X is the letter representing the drive holding your system disk and Y the program disk.

type Y:readme.doc > prn <Enter>; Y is the letter representing the drive with the program disk.

Disk One

This evaluates the suitability of drivers for application in closed, vented and passive-radiator enclosures. It then goes on to allow you to design vented boxes, in detail, complete with port duct calculations. These programs, in general, will not handle closed-box, passive-radiator, large-signal or equalized-enclosure design. This disk does contain an IBM version of the *original* version of BOXRESPONSE by Bob Bullock and Bob White (*SB* 1/84). None of the revisions in more recent issues of *SB* have been incorporated into this version. BOXRESPONSE will allow you to evaluate closed and vented boxes, with or without equalization, but is not really a design program. Disk #1 contains the following programs:

1. README.DOC repeats the operating instructions if you do not have this article handy. More detailed information is provided.
2. NOCOPRSR.BAT is a batch file which installs the line in the computer environment to tell the computer that no coprocessor is available.
3. MENU.COM displays the list of programs, and a summary of their function with instructions on how to run them.
4. BRUN3087.EXE is Microsoft's runtime module.
5. LXCOMP.EXE is a simple program associated with inductance measurement.[1] This program does not have source information on the title page and does not check most data entries.
6. QKLOOK.EXE allows you to enter Thiele/Small (T/S) parameters of one or two drivers and then displays a chart of how the driver(s) will perform in closed boxes

1. Koonce, G.R. "A Technique to Measure Inductance," scheduled for publication in *SB*.

(Q_{tc} = 0.71, 0.86 and 1.0), in vented boxes (QB_3, BB_4 and C_4 alignments) and in a passive-radiator box with alpha = delta. It is meant to give a quick look at how a pair of drivers might be used. Frequency response charts and "text" plots for any selected system are provided, as is vent design.

The program basically operates in English units, but will permit V_{as} entry in liters. It also allows you to add resistance (wiring, choke resistance, etc.) in series with the driver(s). Note that this program designs vented boxes by use of Bullock's alignment charts and is intentionally set up to "round Q_{ts} up," since the real world always works to raise Q_{ts}. The raise is minor, never over 0.01, but may cause some to wonder why they do not get the identical results from hand calculations.

7. NONOPT87.EXE uses the equations developed by Small and Margolis to approximate vented-box design in a specified box volume (*SB* 3/87, p. 24). It allows fitting the drivers you have, into the boxes you have, when the alignment is not optimum. It will handle two drivers at one time and will do vent design.

8. OPTIMUM.EXE is identical to NONOPT87 above, but starts by first doing an approximate optimum design. Can actually do any calculations that NONOPT87 can do.

9. BOXRESQ.EXE is the original version of BOXRESPONSE coded for the IBM as discussed above.

10. VBTUNE87.EXE provides tuning data on a vented-box design. It will, however, allow you to do optimum designs, approximate non-optimum designs or enter just the needed data from an existing design. It has the added benefit, along with vent design and f_B, of computing the locations of f_L and f_H, the two impedance peaks in the input impedance. These are very helpful in tuning an enclosure.

11. PORT87.EXE provides design of the old-fashioned "hole type" ports in vented boxes, which has been omitted from most modern programs. If you are working on older systems it can be very useful. It does no enclosure design; you enter V_B and f_B, which yields required port area for a variety of board thicknesses. Diameters are given for single and multiple round holes along with dimensions for square and rectangular holes fitting the required area.

Disk Two

This disk is intended mainly for the design of passive crossovers, but since space on the disk permitted, I included a few of the programs from Disk #1.

Disk #2 thus allows the evaluation of the suitability of drivers for application in closed, vented and passive-radiator enclosures. The design of approximate optimum and non-optimum vented boxes is also included. Disk #2 contains files 1 through 7, which are the same as those described for Disk #1. In addition, Disk #2 contains the following:

8. TWWYCO87.EXE allows the design of first-, second- and third-order, two-way crossovers. The drivers are assumed to be equalized to fixed resistive loads. Several crossover frequencies are allowed for each output chart. Once you enter resistance and frequency data, you can move from crossover to crossover, allowing a variety of crossover types to be evaluated for your system. You are permitted to independently clear either frequency or resistance data to try different approaches. Easily printed "text" schematics are provided for each crossover type. The program contains a set of definitions of the terms used, provides both series and parallel configurations, and has Butterworth and the older M-derived shapes along with all-pass when the Butterworth does not meet this requirement. This program provides a total of twelve crossover types and arrangements.

9. TRWYCO87.EXE—This program allows the design of first-, second- and third-order, three-way crossovers, and otherwise has the same features as file 8. The design of Butterworth, constant-power and all-pass crossovers is provided; many with both the cascaded and transposed topologies. This program provides a total of eighteen crossover types and topologies. The output chart provides information when midrange peaking occurs or gain adjustment is required.

Conclusions

No knowledge of how to load BASIC or any other computer language is required to run these programs. Files are provided to handle the problem of whether you have a coprocessor or not and to quickly display what is on each disk and how to run each program right from the DOS prompt. No color or graphics are included so the programs are usable on lower-priced computers. The capability to produce hard-copy output of any design work is built right into your computer.

Available on IBM compatible 5¼-inch 360K DSDD computer disks. Distributed through Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, (603) 924-6371; Disk #1, SBK-F2A; Disk #2, SBK-F2B; $12.50 each.

A Two-Way Active Crossover Program

Version 1.2, ©1988

by Gary A. Galo
Contributing Editor

This program would not have been possible without the landmark series on crossover networks by Robert M. Bullock III published in *Speaker Builder* (1, 2 and 3/85 and 1/86). It will perform the necessary calculations for the eight two-way active crossover designs described by Bullock in *SB* 3/85. The program uses Bullock's formulas exactly as given in the article, and therefore uses the "convenient capacitor" method for calculating the resistor values.

I offer no comments on which type of crossover I think is best. I do, of course, have my own preferences, but there is simply too much controversy in this area to allow my own biases to influence users of this program. For a better understanding of the available choices, I strongly suggest reading the articles cited above. I believe this program can only be used to full advantage with a thorough understanding of the material discussed in Bullock's articles.

Using this program is largely self-explanatory, but I'll offer a few helpful comments. Type "Active2" at the DOS prompt to start the program. You may enter lower or upper case letters. The opening menu offers eight crossover choices, a program for calculating Op-Amp V_{th}, as well as an option to exit the program. You must hit RETURN after entering your choice. If you make an incorrect choice, an "Invalid Choice" message will be given.

Entering your menu choice takes you to the appropriate sub-program and you are prompted for a system name. This entry is optional.

Next, you are asked to enter the crossover frequency, followed by a convenient value for the capacitor. I have included capacitor recommendations for various frequency ranges. I recommend 0.1μF and divisions by 10 of this value rather than the divisions of 0.15μF recommended by Bullock. Exact values of 0.1, .01 and .001 are readily available, whereas Bullock's choices often necessitate choosing the nearest available value, typically 0.147 and divisions thereof. I believe my values result in greater accuracy. Nearly exact resistor values can be easily obtained by series and parallel combinations of available values, which are numerous.

I agree with Bullock regarding choosing a value for C that allows you to use resistors no smaller than 1kΩ. After entering a value for C, the calculations will appear followed by an OPTIONS menu.

You may now print the component values, display a schematic diagram, and print that diagram. The schematics are drawn in the text mode using the IBM line drawing characters. No graphics adapters are needed to display them, but your printer must be set up to emulate the IBM ProPrinter in the alternate character set mode. I use a double-lined character to represent ground. Some printers, notably certain Panasonics, will not print the double-lined characters, instead defaulting to their single-lined equivalents. If you select one of the printing options, your printer must be connected and on line. If not, a Device Fault Error message will appear and you will be returned to the DOS prompt.

When the component values are printed, your system name, if entered, along with the current date and time will appear at the top of the page. I suggest printing the component values, followed by the schematics, for the neatest printout. If you wish to include Op-Amp V_{th} calculations on your printout, simply return to the Main Menu and select the Op Amp V_{th} Calculator. I've included an option to advance the printer paper on each menu. If you begin your system printout with the printer head lined up with the perforation in the paper, you can advance the paper to the next perforation when the system printout is complete.

I have written the Even Order Compromise programs so they are user-adjustable. The Equal Compromise Network, with a Voltage Magnitude Peak of 1.50515dB (Bob Bullock gave me the exact value) is the default, but you may input other values within the specified range. There is no sense in calculating a compromise crossover with a Voltage Magnitude Peak of 0dB or 3dB, since these values produce All Pass and Constant Power crossovers, respectively. Far greater accuracy will be achieved by using the appropriate programs for these networks.

I cannot overemphasize the importance of component selection. Use the best available metal film resistors and polypropylene capacitors. The parts available from Old Colony are good choices. The Old Colony crossover circuit boards (DG-13R, board #C-4) make excellent generic crossover boards. Using only two op amps per board, two first- or second-order filters can be constructed with ample space left for adding power supply bypass capacitors and other support components, including R_B. You may need to cut a trace or two, and add a few jumpers. The Old Colony board is quite flexible if you use some ingenuity. R_B, the bias return resistor, is often neglected when designing low-pass filters, since it is normally assumed that the source impedance will provide this function. I recommend including the 1MΩ resistor in each low-pass filter.

The Jung 30Hz Filter (WJ-3, board #F-6) has gained wide acceptance as a crossover circuit board. It is a particularly attractive choice for third-order filters. I do not use quad op amps; obtaining devices in which all four op amps have sufficiently low DC offset to allow direct coupling is difficult. It is also difficult to find quad devices with sufficiently high V_{th}. Using the Old Colony board, you can hand select individual devices for offset. There should be no more that 5mV DC at the output of the crossover if direct coupling is to be used. Third-order filters can be built on the Old Colony board, but one filter per board is the practical limit.

As an aid to selecting op amps suitable for use in audio circuits, I have included a program to calculate V_{th} (Input Dynamic Range). This program is available thanks to Walt Jung, particularly his article in *Audio Amateur* 3/86. The demands of CD players have made V_{th} an important consideration in designing analog circuitry. For a better understanding of this, I recommend you read Walt's article, along with the references he cites. Finally, I thank Contributing Editors Bob Bullock and G.R. Koonce for their encouragement and suggestions during the development of this program.

Available on 5¼" 360K DSDD or 3½" 720K disks. Price includes one year of support for registered users; $20, postpaid in USA. Distributed by: Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458 (603) 924-6371; order #SBK-F2D (5¼"); SBK-F2E (3½").

Tools, Tips & Techniques

Stirred Not Shaken

I hate headphones. Unfortunately, they are necessary in certain instances. Recently, I have been forced by those special circumstances to bring my headphones out of retirement.

Most music is mixed for listening via speakers, not in the closed world of the headphone. Channel separation intended for a normal acoustic environment becomes objectionable using headphones. Fifteen years ago while in college, I had a device which blended the two channels together to reduce the extreme left ear, right ear separation problem. I decided to re-create the device using an empirical approach. The goal was to reduce the low-frequency separation while preserving the left/right perspective at the high frequencies.

The circuit shown in *Fig. 1* was arrived at by a series of listening trials. I made no measurements or calculations until after I had decided that it did the job of blending the two channels satisfactorily.

Blending the left and right channels is easy; just connect a resistor between them. This approach is overkill since you lose virtually all the separation. My circuit yields a modified blend. The addition of C1 creates a low-pass filter in the blend; as frequency increases, the separation increases. *Figures 2* and *3* show the frequency response and separation. If the 470Ω resistors are changed to 220Ω, the midband separation drops to about 3dB, but the audible effect is almost unnoticeable. The capacitor controls the total high-end separation, but I was surprised to note that while the difference approaches 10dB, the audible effect is again minimal.

Most amplifiers and receivers have resistors in series with the headphone outputs. For example, my amplifier has 330Ω in series with each headphone channel. R1 and R2 exist primarily to isolate the high-pass filter from the possible effects of these resistors.

After extended listening, I find the subjective sound to be much better on all types of music, compared to normal headphones. The blend is especially useful when listening to sixties era ping-pong recordings.

Vern Mastel
Mandan, ND 58554

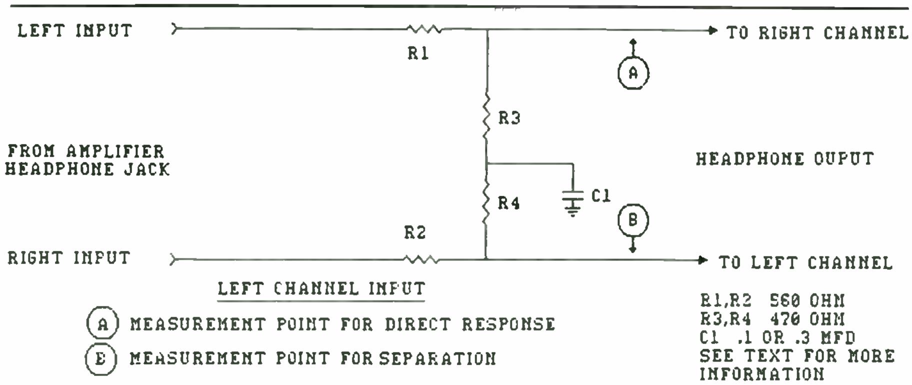

FIGURE 1a: Headphone blend.

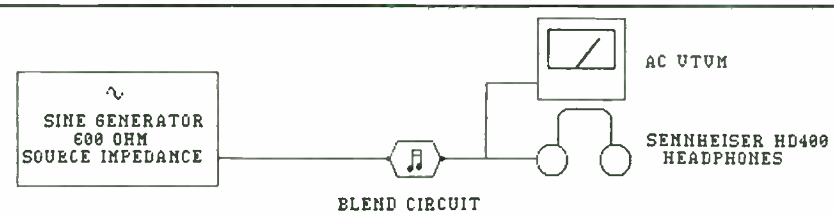

FIGURE 1b: Test setup.

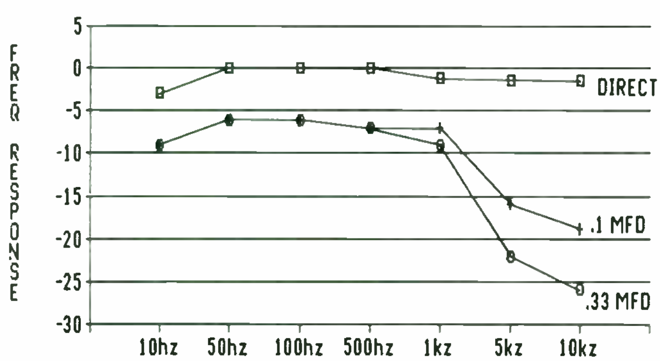

FIGURE 2: Headphone blend circuit output values—frequency response.

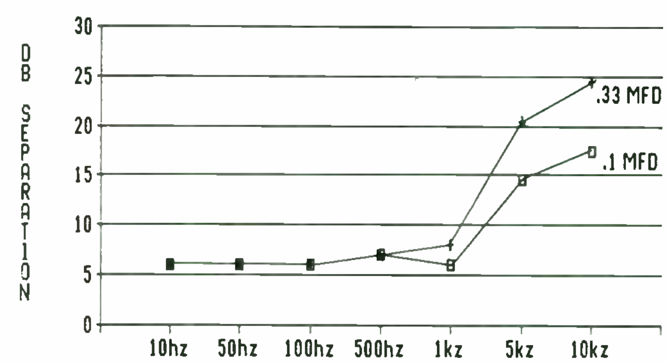

FIGURE 3: Headphone blend circuit output values—separation plot.

SB *Mailbox*

UNLINE ERRATA

A couple of errors crept into the text of my Unline article (*SB* 4/88). On page 29, the sentence, "Even with Q_{ts} .5, Q_{ts} will be down 6dB.", should read, "Even with Q_{ts} .5, 40Hz will be down by 6dB." The following sentence should correctly read, "A speaker with F_s = 40Hz and Q_{ts} .3 will be 3dB down at 121Hz."

In my review of Acousta-Stuf in the same issue, to avoid ambiguities, the sentence on page 49 should read, "I prefer a more reactive material, such as fiberglass for use in vented systems however, as it would adequately damp the midrange without impeding the action of the vent at lower frequencies." I should point out that this is a personal bias and concerns only the Thiele/Small/Keele type enclosures, where alignment accuracy is important. Many fine vented systems have been designed using resistive damping and I'm sure this trend will continue.

Also, in the same issue, my reply to Thomas Cox (*Mailbox*, p. 58) referred to a peak or valley of 1dB; it should be a peak or valley of 1Ω.

John Cockroft
Mountain View, CA 94041

DELAY LINE PHASE SHIFTS

I would like to suggest a change to the procedure in "Electronic Time Delay Line for Speakers" (*SB* 3/88), for determining the acoustic center of the drivers in a multi-way system.

As Mr. Rumreich mentioned, spurious data may result from phase shifts in the drivers. In fact, I don't think he should neglect these, which will often exceed ±90° near each driver's high and low cut-off frequencies. These phase shifts are *not* linear, so it is unlikely that *any* assignment of N values will indicate a constant time delay over more than a very small part of the driver's range.

Continued on page 57

KITS
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FILTERS & SPEAKER SAVER

KH-2: SPEAKER SAVER AND OUTPUT FAULT DETECTOR [3:77]. This basic two-channel kit includes board and all board-mounted components for control circuitry and power supply. It features turn-on and off protection and fast opto-coupler circuitry that prevents transients from damaging your system. The output fault detector has additional board-mounted components for speaker protection in case of amplifier failure. **Each $62**

KF-6: 30Hz RUMBLE FILTER. [4:75] This kit implements Walt Jung's 1975 design for a low frequency garbage filter. The filter knee is set to 30Hz. Roll-off below that knee is the 18dB/octave characteristic of its three pole design. Gain for the filter is unity (0dB) but can be simply adjusted for up to 12dB of gain. The reprint of Jung's article explores the use of the filter with other components in crossovers (see kits SBK-C1A, C1B, C1C). He shows how to obtain slopes of 6, 12 or 18dB in high and low pass filters. The kit contains all parts for building a two channel HPF including a board (3" x 3"), quad op amp IC, precision resistors and capacitors. Requires a bipolar supply of ± 15V, the KE-5 is suitable. **Each $28**

AIDS & TEST EQUIPMENT

KK-3: THE WARBLER OSCILLATOR [1:79]. This unit will produce a swept signal covering any ⅓-octave between 16Hz and 20kHz. The total harmonic distortion at the output is less than 1.5%. The output voltage is adjustable from 0 to 1V. When used with a microphone it is as effective as a pink noise source in evaluating speaker system performance. It also reveals the listening environment's effect on sound through reflection and absorption. The sweep rate is set at about 5Hz. The kit includes 3¼" x 3⅞" circuit board, transformer, all parts and article reprint. **Each $65**

KH-7: GLOECKLER PRECISION 101dB ATTENUATOR. [4:77] All switches, 1% metal film and 5% carbon film resistors to build prototype. Chassis, input/output jacks are not included. **Each $62**

KC-5: GLOECKLER 23-POSITION LEVEL CONTROL. [2:72] All metal film resistors, shorting rotary switch and two boards for a two-channel, 2dB per step attenuator. Choose 10k or 250kΩ. **Each $42**

KL-6: MASTEL TIMERLESS TONE BURST GENERATOR. [2:80] All parts with circuit board. No power supply. **Each $24**

KP-2: TWO TONE INTERMODULATION TEST FILTER. [1:82]. This filter is designed to isolate the two high frequency tones at an amplifier's input from low frequency intermodulation products present at the output. The high pass filter corners at 2kHz and rolls off at 24dB/octave. A 5kHz signal at the low pass input will be down at the output by 80dB. An article reprint detailing design and use is included with the kit. All parts are supplied including quad op amp IC, circuit board and precision resistors and capacitors. **Each $26**

SBK-D2 WITTENBREDER AUDIO PULSE GENERATOR. [SB 2:83] All parts, board, pots, power cord, switches and power supply included. **Each $80**

SBK-E4: MULLER PINK NOISE GENERATOR. [SB 4:84] All parts, board, 1% MF resistors, capacitors, ICs, and toggle switches included. No battery or enclosure. **Each $32**

CROSSOVERS

KC-4A: ELECTRONIC CROSSOVER, KIT A. [2:72] Single channel, two-way. All parts including C-4 board and LF351 ICs. Choose frequency of 60, 120, 240, 480, 960, 1920, 5k or 10k. KE-5 or KF-3 supplies are suitable. **Each $12**

KC-4B: ELECTRONIC CROSSOVER, KIT B. [2:72] Single channel, three-way. All parts including C-4 board & LF351 ICs. Choose two frequencies of 60, 120, 240, 480, 960, 1920, 5k or 10k. **Each $15**

KK-6L: WALDRON TUBE CROSSOVER LOW PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Choose 1: 19-210; 43-465; 88-960; 190-2100; 430-4650; 880-9600; 1900-21,000 hertz. **Each $58**

KK-6H: WALDRON TUBE CROSSOVER HIGH PASS: Single channel, 18dB/octave, Butterworth, [3:79] includes three-gang pot. Please specify 1 of the frequencies in KK- 6L. No other can be supplied. **Each $60**

KK-7: WALDRON TUBE CROSSOVER POWER SUPPLY. [3:79] Includes board, transformer, fuse, semiconductors, line cord, capacitors to power four tube crossover boards (8 tubes), 1 stereo bi-amped circuit. **Each $100**

SBK-A1: LINKWITZ CROSSOVER/FILTER. [SB 4:80] Three-way crossover/filter/delay. 24dB/octave at 100Hz and 1.5kHz and 12dB/octave below 30Hz, with delayed woofer turn-on. Use the Sulzer supply KL-4A with KL-4B or KL-4C.
Per channel **$72** Two channels **$132** SBK Board only **$14**

SBK-C1A: JUNG ELECTRONIC TWO-WAY CROSSOVER. [SB 3:82] 30Hz filter with WJ-3 board & 4136 IC adapted as one channel crossover. Can be 6, 12 or 18dB/octave. Choose frequency of 60, 120, 250, 500, 1k, 2k, 5k or 10k. The KL-4A/KL-4B or KW-3 are suitable supplies. **Each $30**

SBK-C1B: THREE WAY, SINGLE CHANNEL CROSSOVER. [SB 3:82] Contains 2 each SBK-C1A. Choose high & low frequency. **Each $58**

SBK-C1C: TWO CHANNEL, COMMON BASS CROSSOVER. [SB 3:82] Contains two each SBK-C1A. Choose 1 frequency. **Each $62**

SBK-C2: BALLARD ACTIVE CROSSOVER. [SB 3,4:82] three-way crossover with variable phase correction for precise alignment. Kit includes PC board ($5^{3}/_{8}$ x $9^{1}/_{2}$"), precision resistors, polystyrene & polypropylene caps. Requires ± 15V DC power supply—not included. Can use KL-4A/KL-4B or KW-3. Two channel **$145**

SYSTEM ACCESSORIES

KW-3 BORBELY IMPROVED POWER SUPPLY [1:87] This single channel, low impedance supply was designed for the exacting requirements of Erno Borbely's moving-coil preamp [2:86, 1:87]. The design utilizes polypropylene caps and 1% metal film resistors. LM317/337s are used in the preregulator and Signetics NE5534 in the op amp regulator. The kit includes a low profile 24V toroidal transformer, 4¼" x 5½" circuit board and all board mounted components. Chassis and heatsink are not included. Each **$130** Two or more **$122**

KE-5: OLD COLONY POWER SUPPLY. Unregulated, ± 18V @ 55mA. **Each $20**

KF-3: GATELY REGULATED SUPPLY. ± 18V or ± 15V @ 100mA. **Each $48**

KL-4A: SULZER POWER SUPPLY REGULATOR. **Each $40**

KL-4B: SULZER DC RAW SUPPLY. ± 20V @ 300mA. **Each $42**

KH-8: MORREY SUPER BUFFER. [4:77] All parts, 1% metal film resistors, NE531 ICs, and PC board for two-channel output buffer. **Each $20**

SBK-E2: NEWCOMB NEW PEAK POWER INDICATOR. [SB 2:84] All parts & board, new multicolor bar graph display; red, green & yellow LEDs for one channel. No power supply needed. Each **$14** Two for **$22**

KL-2: WHITE DYNAMIC RANGE & CLIPPING INDICATOR. [1:80] One channel, including board, with 12 indicators for preamp or crossover output indicators. Requires ± 15V power supply @ 63 mils. Single channel. Each **$58**
Two channels. **$110** Four channels. **$198**

KW-1: MAGNAVOX CD PLAYER MODIFICATION. Improves frequency response. Includes two Signetics NE5535s, two Panasonic HF series 330µF capacitors and four 3.92k, 1% metal film resistors. **Each $12**

KW-2: MODIFICATION. As above, but with two AD-712 op amps in addition to the NE5535s. **Each $16**

KX-1: DISC STABILIZER. Set of 3 Sorbothane feet, 3 Tiptoes and Mod Squad's Disc Damper with 15 centering rings. **Each Set $70**

KX-2: POOGE-4 CD PLAYER MOD. [1:88, 2:88] Jung/Childress extensive rework of the Magnavox CDB 650. **Each $163**

HDHFT: HI-FI TIPS. Imported for Old Colony. Solid brass, ⅞"H conical feet for components and loudspeakers. Includes self-adhesive pad. **Each $3.00**
10 or more **Each $2.50**

What's Included? Kits include all the parts needed to make a functioning circuit, such as circuit boards, semiconductors, resistors and capacitors. Power supplies are not included in most cases. Unlike kits by Heath, Dyna and others, the enclosure, faceplate, knobs, hookup wire, line cord, patch cords and similar parts are not included. Step-by-step instructions usually are not included, but the articles in ***TAA*** and ***SB*** are helpful guides. Article reprints are included with the kits. Our aim is to get you started with the basic parts—some of which are often difficult to find—and let you have the satisfaction and pride of finishing your unit in your own way.

Continued from page 55

Instead, I would choose to assign N values which result in a phase shift of about +90° near the driver's low-frequency cutoff and −90° or −180° near the high-frequency cutoff (depending on whether the high-frequency roll-off slopes more closely approximate 12dB or 24dB/octave, respectively), *relative to the phase shift at the center of the driver's frequency range.* Use the delay calculated at the center of each driver's frequency range to determine the relative delay compensation.

Notice that the delays which result from these calculations will often agree with the time delays corresponding to the *physical distance* between the drivers' diaphragms and the microphone element. Since each driver acts as a band-pass filter, introducing nonlinear phase shifts near the extremes of its frequency range does not introduce a *time delay* per se. Thus, we may simply measure the distance between each driver's diaphragm and the listening position, and calculate the relative delays corresponding to these distances: delay (sec) = distance (inches) / 13,500.

Most drivers have natural 12 or 24dB/octave slopes below and above their frequency range. If your crossover frequency is within an octave or two of the driver's response limits (which is usually the case), you must consider the magnitude and phase effects of these slopes when designing your crossover. Using active crossovers and eliminating interdriver depth displacements via electronic time delay simplifies crossover design and generally results in less crossover ripple, but does *not* avoid the effects of magnitude and phase responses of their individual drivers.

I recommend using a modelling program like LMP (*SB* 1-3/87) to predict the effect of combining crossover filters with real-world drivers in multiway systems. If you use LMP with the electronic time delay, be sure to specify "0" for the interdriver DEPTH DISPLACEMENT prompt. Also, you can model active crossovers with the existing LMP subroutines by calculating the component values for the equivalent passive crossover.

Because of driver response limitations, you will find it nearly impossible to obtain a true first-order, minimum-phase crossover. Unfortunately, no LMP subroutines can allow modelling of Lipshitz and Vanderkooy's "delay derived" minimum-phase crossover networks. Perhaps Mr. Rumreich or a technically-inclined reader could produce such subroutines?

Ralph Gonzalez
Philadelphia, PA 19143

Mark Rumreich replies:

You have a valid concern about the phase shifts which accompany a driver's natural rolloff. I agree these phase shifts are not linear and therefore will not produce a constant delay calculation for the correct (or any) assignment of N values. I believed these phase shifts were small enough in the midband to allow unambiguous assignment of N by my method.

In my example for a Dynaudio D-52 AF, good consistency was obtained for the midband points. The out-of-band points were discarded when making the final delay assignment. I found similar consistency in the midband regions of the woofer and tweeter.

Trying to derive useful N-assignment information from the out-of-band points is interesting. I would like more details of your idea for assigning N values which result in a 90° phase shift near cutoff with respect to midband. Have you found drivers whose midband phase shifts cause ambiguity in N assignment by my method? Does your proposal resolve them?

Your alternative measurement procedure avoids my more complicated method. I chose to measure the acoustic centers of my drivers because I was not sure which point on the diaphragm (if any) to use. Assuming the acoustic center of the mike is at its tip, I obtained the following differences in the driver's acoustic center (inches behind the cone center) between the two approaches: D-21 AF, 0.26; D-52 AF, 1.23; 21W54, 1.93.

The large spread between drivers is surprising. Unfortunately, I lack the manufacturer's specs. However, a final system check using a pulse generator and HP3562A Dynamic Signal Analyzer showed proper delay alignment using measured delays. I am interested in understanding what determines the location of the driver's acoustic center.

The biggest problem in building the delay-derived networks of Lipshitz and Vanderkooy is the lack of an ideal delay line. Long delays are required, and small amplitude and delay ripples cause large frequency-dependent subtraction errors. I am in the process of developing a three-way system which deals with this problem, but I think it's safe to say the delay-derived networks are not yet ready for LMP.

SUBMARINE SONICS

I applaud Peter Muxlow on his article, "Loudspeaker Enclosures," (*SB* 2/88). This subject is important to home builders, who don't have the cost and schedule constraints of manufacturers. We don't mind putting in the extra effort to build enclosures with odd angles, extra bracing, wall damping, and so on.

Mr. Muxlow covered cavity resonance and its effect on cone motion and cabinet vibration very well.

The sonic improvements attainable with competent cabinet construction can be remarkable. A number of years ago, I was privileged to hear a demonstration tape made by KEF on the development of the Model 105; recordings of a solo cellist using a 105 of basic construction, and another that received treatment based on experimentation with accelerometers. The modified unit was much clearer and better focused.

I have performed a lot of sound and vibration testing on naval nuclear submarines. Just as for a loudspeaker enclosure, it is not desirable for a submarine hull to be a sound source. Submarine hulls do receive treatment against acoustic coupling of sound to the hull, but most of the treatment is for mechanical coupling. In Mr. Muxlow's research, which type of coupling is dominant in exciting cabinet vibrations?

Older articles I have read would indicate that acoustic coupling is the dominant force. My experience and recent statements by Linkwitz and D'Appolito indicate that mechanical coupling is the main problem with cabinet wall excitation.

Mr. Linkwitz suggests mounting the driver by its magnet structure and using a compliant seal between the basket flange and enclosure front (*Mailbox, SB* 2/86, p. 54). This sounds like the best method I've heard of so far.

I no longer have access to accelerometers, velocity pickups, sound analyzers, spectrum analyzers, and so on. Testing the different driver mounting schemes could make an interesting article for *SB*. I hope Mr. Muxlow can shed some light on this.

William Harnois
Vallejo, CA 94591

Peter Muxlow replies:

I don't know what creates the most delayed resonance distortion, acoustic pressure inside the cabinet, or driver vibration. It would depend on the type and size of cabinet, driver specs, and so on.

I first became aware of the resonance problem Linkwitz mentioned, from Harwood's paper.[1] He mentions two resonances, the first between the mass of the driver unit and the stiffness of the front panel. The solution to this is to make the front panel only slightly wider than the driver unit. This maximizes panel stiffness.

The second resonance can occur between the basket stiffness and the magnet. This can be minimized by bracing the back of the magnet. Harwood suggests a layer of dense felt as a mechanical resistance between the magnet and the brace. He does not mention decoupling the driver from the front panel. KEF, B&W and Jamo mount their low-frequency drivers on rubber bushes or grommets to decouple the driver from the cabinet. In my experience this technique seems to produce a poor definition. Perhaps I'm biased because I don't think drivers bouncing around in their mountings is a very elegant engineering solution.

Another method which cancels the reaction force is using a second loudspeaker wired out of phase. This is coupled mechanically to the first loudspeaker via a rod. As one cone moves forward the other cone moves in the opposite direction, cancelling the reactive force. KEF and Mordaunt-Short use this for their top-line speakers; a good but expensive solution.

An interesting concept has just been pioneered by British maufacturer Mordaunt-Short. They attach the drivers to a stand assembly and the cabinet is then hung around it, suspended on rubber sealing gaskets. This decouples the whole cabinet from the frame and drivers.

NOISY APC NETWORK

I recently constructed a 24dB/octave APC active network per Bob Bullock's article (*SB* 3/85) and have encountered a noise problem with the circuit. I've taken every precaution possible to ensure quiet operation but I'm getting a noticeable hiss at the outputs. I selected all components for lowest noise, though I replaced the 4136s with TLO-75s which, admittedly, are slightly noisier. The power supply is a Sulzer regulated supply which has been built for lowest noise and impedance.

The trouble starts when the boards are cascaded for the 4-pole slope. By themselves, they appear to exhibit quite acceptable noise levels. The gain is set at unity for each board and I checked the active components for proper operation. How do I deal with this noise?

Second, when measuring a closed-box system for final system Q (Q_{tc}), it seems

1. Harwood, H. and R. Mathews, "Factors in the Design of Loudspeaker Cabinets," *BBC Report,* 1977.

the resistances of R_X and R_G should be figured into the overall closed box Q, since they would raise this figure when connected into the complete system. Is this a correct assumption?

Third, I've been designing a dual voice coil, closed-box system. Since both coils must be measured to obtain the parameters and are driven in a common box by two amplifier channels, shouldn't R_X and R_G be doubled to account for the two separate sources feeding the single box/driver combination?

Finally, how could I incorporate the active filter used in Mr. Bullock's design of a sixth-order vented alignment, to augment the response of a closed-box alignment of various system Qs?

David Del Zotto
Seattle, WA 98155

Contributing Editor Bullock replies:

I have used Jung's rumble filter in several different configurations and I have never been aware of excessive hiss. My current crossover is sixth-order, so the band-pass and high-pass paths involve four cascaded sections. Here are some possible explanations for your problem:

You have some defective components, although I doubt this.

Your grounding scheme may be inadequate. I had a problem in the first active crossover I built because of poor grounding technique.

The hiss you hear may be masked in my system but not in yours, or perhaps I tolerate more hiss than you. My noise criterion is not at all sophisticated: my active network systems are audibly less noisy than the same loudspeaker setup with a passive crossover. I wish I could be more helpful.

The effect of R_G is probably negligible with a transistor amplifier. If a passive crossover is used, it is probably a good idea to simulate R_X in the system to get the best value of Q_T.

I have never thought about how to model a dual voice coil driver, but my impulse would be to treat it as a single driver with an R_E of one coil for impedances purposes. I would not double R_X and R_G.

Your last question sounds like you are asking about fourth-order closed-box alignments. Offhand, I don't know any tabulations of such alignments. The easiest suggestion I can make is to try trial-and-error alignments using BOXRESPONSE with the second-order equalizer. That is, keep the box fixed, but vary the equalizer corner frequency and damping coefficient until you get a response that suits you. Then use the formulas (*SB* 1/82, p. 24) to design the filter. When using the formulas, take A to be the damping coefficient and f_a to be f_E.

MAGNET SOURCE

I'm glad I was able to find a magazine like *Speaker Builder*. I loved the issue this year on the kitchen bench ribbons. I am in the process of putting my own pair together.

If anyone is interested in a good magnet supplier, I ran across one:

Magnet Sales and Manufacturing
11250 Playa Court
Culver City, CA 90230

They offer a good variety of ceramics and even rare earths with samarium and neodymium for stronger fields. They will custom manufacture whatever you want. Hope this will help out some builders.

Greg Hansuld
Ames, IA 50010

BUILT-IN CURRENT PROBLEM

Mr. Wagaman's letter ("Full Range ESL," *SB* 4/88, p. 56) depicts his proposed planar speaker with a built-in problem. His *Figure* shows magnet polarity, when read from left to right, is north to south across every ribbon conductor. Since the aluminum conductor follows a continuous serpentine path, current flowing up in one vertical leg at a given instant will be flowing down in the adjacent legs.

If the conductors are numbered from left to right, this will result in the odd numbered legs moving one way, for in-

Continued on page 61

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CD-457 *Warren Bernhardt* Warren Bernhardt (piano), Peter Erskine (drums), Marc Johnson (acoustic bass), Anthony Jackson (electric bass), John Tropea (electric guitar), Kenny Ascher, Robbie Kondon, Rick Tuttobene (synthesizers). *"...a stunning piano recording, flawless playing, exceptionally well recorded."* —*Hi Fi News*

CD-458 *Neon* Flim & the BBs. Billy Barber (keyboards), Bill Berg (drums), Flim Johnson (basses), Dick Oatts (sax, flute). *"Another exciting triumph from Flim & the BBs! There's a marvelous blending of energy and gentle shimmer to this CD music. Highly recommended.* —*High Performance Review."*

CD-459 *Braziliana* Manfredo Fest. Manfredo Fest (keyboards), Portinho (drums), Cyro Baptista (percussion), Paul Socolow (bass), Roberta Davis (vocals). *"...immaculately clean, dynamically exciting sound...."* —*Audio*

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Continued from page 59
stance, away from the frame, while the even legs move toward the frame. The result will be cancellation of all output for frequencies with wavelengths large compared with conductor spacing. This is not a desirable outcome.

Robert Kuntz
Medford, OR 97501

ODD-ORDER BESSEL FILTERS

In Mr. D'Appolito's AES preprint #2000, he concludes that with his driver arrangement, odd-order Butterworth filters with their 90° phase shift offer the best frequency response. Would it also be possible to use an odd-order Bessel filter, since, if I understand correctly, it also has a 90° phase shift?

Can the trade-off of improved step response (no ringing and overshoot) versus shallower initial rolloff, be minimized by using a steeper, third- or fifth-order Bessel filter, or are the advantages of a steeper slope nullified by having to cascade several op-amp filters to obtain the steeper rolloff?

Lastly, will a second-order, active equalization network for a sixth-order Thiele/Small alignment, change the relative phase between the high-pass and the low-pass filter sections of an active two-way D'Appolito speaker system?

Thilo Stompler
San Diego, CA 92122

Contributing Editor D'Appolito replies:

In the Preprint I concluded that the odd-order Butterworth filters, which have an interdriver phase shift of 90°, produced the most uniform vertical polar response with my driver arrangement. This should not be confused with frequency response. Such polar response may or may not be desirable. My recent article on the Swan IV system (*SB* 4, 5/88) argues for a rather narrow vertical polar response. The odd-order Bessel filters also have an interdriver phase shift of 90° and would therefore produce polar response patterns very similar to those of the same order Butterworth filters I discussed in my paper.

Bessel low-pass filters have maximally flat time delay or equivalent maximally linear phase response. Thus, they provide minimal time smear at the expense of amplitude accuracy. They are not completely free of overshoot or ringing (a common misconception), they simply have less than other filters. The reduction in overshoot is obtained at the expense of rise time, so whether or not this implies superior transient response depends on what you are trying to accomplish.

Unfortunately, the properties of the Bessel low-pass filter are not inherited by the Bessel crossover and the presumed advantages do not exist. The coefficients of the Bessel characteristic polynomial are not symmetric and thus the high-pass filter derived from the low-pass to high-pass transformation does not have a constant 90° phase shift relative to its low-pass prototype. The high-pass filter of the so-called Bessel crossover uses the low-pass polynomial to retain the 90° phase shift, but otherwise has no particular property of its own. The high- and low-pass outputs do not sum to flat amplitude response, nor is the sum minimum phase. In fact, in the crossover region the summed response dips about 3dB. Thus, I see no particular advantage to this crossover as the attendant amplitude response dip is quite noticeable.

The low-pass Bessel filter has been combined with tweeter time delay and high-pass filters of other alignments to produce fairly successful crossovers with minimum phase response over a reasonable frequency range. Ralph Gonzalez' article, "Minimum Phase Crossovers," (*SB* 3/88) reviews this work. The interdriver phase angle for these configurations, however, is not constant, inducing polar-axis wander in conventional geometries. My 3/2 geometry will reduce the effect of the phase angle variation by stabilizing the vertical polar response. Tweeter delay relative to the mid-bass driver, however, is best obtained by electronic means. The slow rolloff of the Bessel low-pass filter means that you must be careful to include the effect of the driver response in the overall Bessel characteristic. At the very least, you'll need a good simulation program like LMP to perform your design. Because of the complexity of the design process and the strong dependence on driver responses, any such design should be measured to determine actual performance. Higher order Bessel filters don't really help since they all have pretty slow roll-off characteristics.

Regarding your last question, as long as the corner frequency of the sixth-order alignment is a decade or more below the mid-bass/tweeter crossover frequency, the effect on interdriver phase should be minimal.

NONPARALLEL LINE

I read Mr. Cushing's article, "A More Compact Transmission Line Subwoofer" (*SB* 1/87) with great interest. Using a reducing section line with a 4:1 taper reduces the volume of the enclosure substantially.

I have a specific design in mind which will need only two folds in the length of the line. My plan is similar to *format B* from *Loudspeaker Design Cookbook*, (p. 39). I had intended to use a gradual taper in my design by angling the center partition between the two folded portions of the line. Can I use a gradual taper instead of the stepped tapering used in Mr. Cushing's enclosure, without adverse resonances? It seems that this would actually reduce unwanted resonances since two of the walls of the line would be nonparallel.

Are the exact measurements of Mr. Cushing's enclosure available? He mentions the line length is over seven feet. What was the approximate cutoff frequency (f_3)? Can I use a 9 or 10-foot line length to lower f_3 to 20Hz?

Lastly, can two identical drivers be used in parallel in the same transmission line, providing the cross-sectional areas are doubled? If I obtain good results I will send a layout of the plans to *SB*.

Thanks for a great magazine and any help you can offer.

Ricky Judge
Andrews AFB, MD 20335-5000

Craig Cushing replies:

Your proposed design should function well and you're probably right that the gradual taper is less likely to induce resonances than my stepped design. Because virtually any design is a series of compromises, my main one was to use the stepped mode to minimize size, maximize line length and simplify construction, while hoping the multitude of line direction changes would cut down on any nasty resonances—and it works.

If you'll reread my article, note the total line length is 11.3 feet—the last four of which are not tapered. This gives a cut-off frequency of 25Hz, which is the approximate resonant frequency of the PLI 12-inch woofer I used. It is entirely possible to make the cabinet taller, increasing the line length to any desired f_3, if the chosen driver has a comparable resonant frequency. Anything longer than about 14 feet becomes pretty academic, as you're into infrasonic territory.

My most recent subwoofer uses two PLI TD205R, 8-inch woofers in parallel. The line itself is virtually identical to my first published design (*SB* 1/85). The line cross section is sufficient to allow using these two drivers. Power handling is exemplary, bass extension remains excellent, sensitivity is improved and cabinet size is essentially the same. If you decide to use the double woofer format, make sure the resonant frequencies of the drivers are within 2 or 3Hz, so they'll work equally hard. I hope to see your design using two 12-inch woofers per cabinet in a future issue of *SB*.

INSULATED vs. NONINSULATED

I've read Mr. Sanders' articles, Mr. Wagner's book and Peter Walker's work, as well as others. In addition, I have used, repaired, tested and built a range of ES drivers and systems.

Clearly the dielectric strength, or breakdown voltage of air (which varies according to dust and humidity) is the determining factor in deciding what the combination of stator-to-diaphragm spacing vs. voltage can be. Given a particular spacing, then the maximum driving voltage *plus* the polarizing voltage cannot exceed the breakdown voltage of the air gap. Don't forget the gap gets smaller as the diaphragm moves towards the stator.

Adding insulation to the stators (or plates) then increases the total breakdown voltage across the gap. In addition, it changes the dielectric constant of the capacitor (which is formed by the stator and the diaphragm) in the same way that capacitance and breakdown will change in standard capacitors using different dielectrics, e.g., Mylar, paper, polypropylene, polystyrene, oil-filled, vacuum, and so on. This also tends to vary the total capacitance of the system, usually towards more capacitance. I am not sure how this affects the charge transferred to the diaphragm, but my empirical tests seem to show that the diaphragm doesn't care since it operates just about the same way, indicating that the charge that it sees is essentially the same.

As long as you do not exceed the breakdown voltage of the gap, insulation is not required. Want to make a stronger electrostatic field? Want to drive your system with more voltage? Use a higher polarizing voltage, or use a bigger amplifier. Given the same physical setup, you need insulation. What then is the issue?

Let's talk about some designs that have been successful. The industry standard Janzen (4 by 4″) and RTR mid-tweeters, 2 by 4″ (RTRs were used in Infinity Servostatic 1s) are both the same design, employing graphite-coated (medium Megohm resistance/area) diaphragms, and insulated wire stators. They will handle up to <3kV polarizing voltage with no problem. The gap visually is small, to be sure, although I have not measured the RTR gap precisely. However, based on prototypes tested without insulation, they would surely arc at voltages well below 3kV without insulation. Keep in mind that they are designed from the factory to run at about 1kV. The ES mids in the same system used 7kV and a large gap, and were frightfully inefficient.

It did take more than 20 years, but these tweeter panels do fail, mostly due to migration or loss of the graphite from the diaphragm contact. Disbelievers can come here and examine some cells where this has occurred, so rubbed-on graphite is not perfect material either.

Harbeck, a 1950s manufacturer, made panels without obvious insulation; I have some running at present. These panels were about 3 by 10″. They used perforated metal finished in flat black, the surface of which is conductive (on both sides), not anodized (an insulator). They used two metallized Mylar diaphragms! Cleverly, two diaphragms per cell, each had the metallized side facing in, toward each other, (with a gap between each about equal to the gap to the stators) leaving the insulated (un-metallized, nonconductive) side out, toward the stators. They were surprisingly efficient, roughly equal to the Janzen's (RTR's) efficiency, yet covered a wider frequency range, going lower into the midrange, and used a lower polarizing voltage (<750V) with a much wider gap <.0625″.

The much wider frequency response of the Harbecks can be attributed to the large (relatively), free diaphragm of 30″², while the Janzen is in effect made up of approximately ¾ by 4″ strips. The result of the mechanical construction limits the excursion of the Janzen's diaphragm so that it does not hit the closely-spaced stator. This sort of trade-off is typical of the real world of speaker design, as many of us have learned through our own trial and error.

Speaking of free diaphragms, the Martin-Logan system does not have a completely free suspended curved diaphragm. Clearly, the diaphragm has horizontal supports spaced about every 5-8″ in the vertical axis. It appears that these supports are spaced such that they taper, starting with small spacing at the top and ending with a larger spacing. I presume this permits the mechanical resonance which must be associated with the vertical dimension (probably in the low midrange) to be spread across a range of frequencies, rather than occurring at one single frequency. These supports appear to be made of grey urethane foam and sandwich the diaphragm front and back.

Little is gained (the curve is nice, though) in this method, beyond the closeness with which adjacent panels (cells) can be placed compared with individually built cells. However, since in most practical systems the high-frequency output tends to be at a maximum from the *center* of a cell (or a vibrating and suspended surface), the design advantage is nil, since the centers of the cells are still about 7" apart. The curve of the diaphragm does afford smooth horizontal dispersion.

I am not sure why this effect occurs. Perhaps, at the center of a uniformly suspended diaphragm, the center point has maximal excursion; at the same time, it has less stiffness and is less controlled. For a uniform diaphragm, this point would seem to exhibit the lowest effective mass. Therefore you can most easily drive the center point, and barring reflections from the edges, this point will have the greatest output (excursion) for a given input.

Of course, you may argue that the diaphragm is driven equally over its entire surface, with equal force, and therefore behaves like a piston of minimal mass. Why then do many real electrostatic (and other) systems "beam" the high frequencies from their midpoint? If ES diaphragms really behaved like flat pistons driven equally across their surface, acting like a single stiff unit, then why would large unsupported diaphragms tend to touch the stators at the midpoint?

The Martin-Logans use a synthetic coating on their perf metal stators. They are well-coated and the diaphragm is just about transparent. Probably this indicates the use of ivory soap or graphite! Could these speakers still work with just a raw metal stator and no insulation?

Let me pose a question. If the diaphragm can be made with a sufficiently and uniformly high impedance of many Megohms, does this mean a current cannot be drawn which is sufficient to create a breakdown of the air gap, regardless of the typical ES operating voltage? Is this what Mr. Sanders is saying?

Specifically he states his diaphragms will not arc, since they will not permit the full capacity of the cell to discharge across the gap at *one point*! This is a key statement. Many designs seem to rely upon a series resistance (of many Megohms) placed between the polarizing supply and the diaphragm (for current limiting?). Perhaps this is the source of the problem. Peter Walker speaks of various modes of operation for the ES cell, and identifies very high impedance as a requirement for constant charge operation.

Regarding the constant charge requirement, if the impedance of the diaphragm is "too" high, will that affect the time it takes to "recharge" the capacitor? Peter Walker states that the time constant of the system, or time it takes to recharge or maintain charge, must be large enough to allow for depletion from the lowest frequency input (where the charge will be depleted longest). Would a resistor in series with the polarizing supply have the same effect as a high impedance diaphragm? Does this high Z diaphragm affect "charge migration?"

Of course low impedance diaphragms seem to operate just the same. Or do they? What are the mechanics of the breakdown of the air between the conductive diaphragm surface and the stator? Will a sufficiently high series resistance suffice, or does a distributed resistance across the diaphragm surface solve the breakdown problem? Let us not forget that an insulator may be thought of as merely a very high impedance material.

Are these the essential issues when discussing insulated vs. noninsulated stators?

Randall Bradley
Hannacroix, NY 12087

Mr. Sanders replies:

Mr. Bradley's letter is refreshing and stimulating, bringing up more questions for thought than answers. I find myself in agreement with most of his observations and encourage interested readers to consider his points carefully and perhaps do some experiments to determine the answers

Continued on page 66

Amp/Loudspeaker Interface

Continued from page 21

types of protection circuits, including those controlling DC offset at the output terminals as well as those controlling potentially damaging combinations of output voltage and current. The first part of this addition, dealing with the latter subject as considered in this paper, is to be circulated shortly by IEC Technical Committee 84 to IEC national committees under the accelerated procedure.

EDITOR'S NOTE: *In the next issue, we will publish a practical implementation of Peter Baxandall's test device devised by Reg Williamson, with collaboration by Alan Watling. If you are interested in a kit of parts and circuit board for the project, please use Fast Reply No. JC101.*

QB_3 Vented Box

Continued from page 24

8Ω woofer) and correct the driver Q for this resistance. I believe you will be delighted with the system that results.

CONCLUSION. I perceive better bass quality from the QB_3 VB alignment than from other VB alignments, or from the typical CB system built with $Q_{tc} > 1$. I believe CB systems with low Q_{tc} values should also offer excellent bass quality.

I have shown that one feature of both low Q_{tc} CB systems and QB_3 VB systems is that major resonances are below the pass band and I believe this is related to the sonic advantage I hear.

Any sonic advantage may be related to other characteristics of these two classes: for example, their slow, smooth rolloff as frequency decreases and the resulting transient response performance this offers. Whether the sonic advantage results from suppressing the resonances as I have proposed, or some other effect, I strongly recommend you consider using the QB_3 VB alignment for your building projects.

Swan IV

Continued from page 45

with the same center spacing. (Later, you could enlarge the holes for 12" drivers.)

The ECLIPSE W1038 is a "drop in" replacement for the Precisions. No other changes are needed.

The VIFA M25WO-31-06 has too low a Q_{ts}, but a 1.5Ω, 20W (or more) resistor in series with the paralleled driver pair allows them to be used.

The FOCAL 10K515T and 10N515 drivers can be used if the length of the ducts is reduced to 6½" (F_b=27Hz rather than 22Hz).

The JUST SPEAKERS DR 1056 can be used with 6½" ducts.

His comments regarding the voltage breakdown of air vs. voltage are correct, in my opinion. A compromise you must make when designing ESLs is determining the diaphragm-to-stator spacing. I chose a rather small distance (about 60–80 mils) largely because of the air gap breakdown problem. The closer the gap, the easier it arcs, but the voltages are reduced by the square of the distance for a given SPL. The result is an arc resistant speaker.

I currently drive my speakers with a Hafler 500 amplifier. This amp is capable of swinging an output voltage of nearly 180V. Using transformers with a 45:1 turns-ratio gives me a peak drive voltage of around 8kV. Add a 3kV polarizing voltage and you have 11kV! By all rights, the speaker should arc...but it does not. For that matter, the transformer, rated at 1.5kV RMS, should arc, but it does not. I have measured this voltage, but I do not understand why it does not cause arcing. I would be most interested in authoritative answers.

Another good question is how a 15W transformer can take the full output of a 250W amp without distorting. Clearly, the voltage and current are out of phase when driving a pure capacitor (the current leads the voltage by 90°), but a transformer driven capacitor is not a pure capacitive load, so a considerable percentage of actual power must run through the transformer. The amp easily pops 3A fuses in the output circuit (I run 5A fuses without difficulty), so a lot of current and voltage are going through the transformer...and quickly too! How?

Mr. Bradley states that as long as you do not exceed the breakdown voltage of the gap, insulation is not required...so what is the issue? Actually, the most important one has perhaps been overlooked: ESLs usually arc for reasons other than air gap breakdown! The most common reason is a foreign object, for many, a gnat or mosquito, that gets between the stator and diaphragm; although in my case it is often metal chips and carbon fibre.

Another common cause is the diaphragm being driven into a stator, usually caused by testing the polarizing supply and exceeding the stable voltage of the speaker, or by a badly warped stator, or by excessive excursion caused by pressure waves in the room, generated by the woofer. It is surprising that the ESL is so closely coupled to the air. You can commonly see your reflection in the diaphragm moving radically at low frequencies. In a hybrid system, this is caused merely by the woofer! If you doubt this, simply turn off the ESL drive and play the same passage of music...the ESL will continue to move, driven totally by the sound pressures generated by the woofer. In rooms with severe standing wave resonances, this is a serious problem, but it can usually be solved by moving the ESLs to the long wall, or by changing to linear woofer systems if you are not using TLs. You may also have the woofer level excessively high compared to the ESL. In any event, the ESL must sustain these arcing conditions with no or minimal damage...hence the insulation issue still remains. My ESLs prove that speakers will survive without insulation, and spray-on varnish is not very good insulation.

Finally, my designs are carefully engineered to be easy and simple for amateur builders, as long as this does not compromise performance. Omitting insulation is one more step in this direction.

Mr. Bradley discusses several points with regard to diaphragm movement, an area where hard data is extremely scarce. Much more work seems to be needed, and I would be most interested in any information that readers might be able to supply. My experiences are quite different than his, and they do not support his statements. For example, Mr. Bradley believes the diaphragm moves in an arc, with maximum excursion at the center. This may be true in a vacuum (anyone care to check it out?), but there is evidence that it is not true in air. I submit the following supporting data:

1. SPL measurements at near field reveal the same output over the entire surface except within about ¼" of the restraining structure;
2. You would hear a "venetian blind effect" as you passed in front of an ESL with insulating strips, if the movement was in an arc, but you do not;
3. Distortion measurements would be high, and they are not;
4. Watching one's reflection in the diaphragm using a triggered strobe reveals that the reflection is not significantly distorted;
5. Uncoated areas of the diaphragm as small as 2" in diameter result in zero output there, with normal output all around it. This suggests that the diaphragm is indeed being driven precise distances by the electric field rather than being "carried along" by the surrounding diaphragm. Such tight control by the electrostatic field precludes aberrant motion such as arcing;
6. Arcing occurs randomly on the diaphragm, not primarily at the center. This is readily observable, or inspect an old ESL—likely it will have numerous holes burned in the diaphragm and these will be randomly placed, not just at the center of the diaphragm;
7. Planar ESLs are highly directional, which could only be true if the diaphragm moved in a piston-like manner.

Mr. Bradley suggests that high frequency "beaming" is due to the diaphragm moving in an arc rather than in a piston like manner. Actually, wide dispersion is well recognized to be caused by curved wave fronts, such as delay lines (like the new Quad), or by curved surfaces, or by surfaces that are smaller than the wavelength being generated. Planar speakers "beam" because they produce flat wavefronts at frequencies whose wavelengths are shorter than the speaker. Wavelengths that are long can be thought of as "spilling over" the edges of the speaker and producing curved wavefronts. A speaker diaphragm that moved in an arc would tend to produce curved wavefronts and therefore wider dispersion.

A very important psychoacoustic phenomenon exists regarding directionality of planar speakers, that I have never seen mentioned in the literature. This strongly affects the perception of directionality in such a way that the speaker seems even more directional than it is. Try this simple experiment—assume that you have a pair of 2-foot wide, planar ESLs which are driven over their entire surface and are set up in a typical stereo soundstage manner. Now turn off one channel. Listen to only one speaker and you will note the sound spreads over a 2-foot wide area (including the highs). It doesn't matter how far away you are, there will always be a path at least 2-feet wide that contains the full spectrum of music. Now turn on both speakers. Note that the sound is only "correct" at a small point directly between the two speakers and equidistant from them. This point is not 2 feet wide, but rather infinitely small. Crossing the beams from the two speakers causes the effective listening area to decrease to much smaller dimensions. I suspect this has something to do with phase relationships perceived by the brain, and is not actually a part of speaker physics. If we study this, perhaps we could develop planar speakers to correct this, while still retaining all the attributes of the planar speaker. Studies in this area are nonexistent to my knowledge. I would be most interested in hearing from any reader who can shed some light on the subject.

Mr. Bradley speaks about fundamental resonance and the possibility of differently spaced insulators being a way of breaking up the resonance. I have found fundamental resonance in ESLs to be quite different from dynamic speakers. In a dynamic speaker, the cone is quite massive compared to the air it is driving and the resonance of the system is determined by the mass of the cone working against the spring rate of the suspension system. In a large ESL, the mass of the driver is really the mass of the air in the room (the mass of ¼ mil Mylar is equal to about 7mm of air). This effective mass is variable, depending on the room size, and is further modified by large openings in the room, as well as standing-wave resonances. This air mass is quite complicated and works against the spring rate of the diaphragm. I have found it impossible to predict the fundamental resonance of an ESL because of this.

I haven't found that making slightly different spring rates makes much difference, (as Martin-Logan may be doing by using different distances between supports) in the massive fundamental resonance that plagues ESLs operated into the bass. This is just one more reason to use a conventional dynamic woofer system. Cross over the ESL well above fundamental resonance and then the only problem with resonance will be caused by woofer coupling, previously described, that is not nearly as troublesome.

There appears to be some confusion regarding "free" diaphragms as they relate to curved, or any other, ESLs. The construction of the Martin-Logan is considered a free diaphragm because there is no restraining structure used to hold the diaphragm in its curved shape. It is free to move, just like a planar ESL. That a rare insulator is required (to maintain the 100:1 ratio between diaphragm unsupported area and diaphragm to stator spacing for stability) does not change the fact that the diaphragm is essentially "free." Compare this to the usual method of curving a diaphragm, where a solid sheet of soft foam behind the diaphragm keeps it from being pulled straight by diaphragm tension.

I would like to make one last point of clarification. Mr. Bradley states that it is his understanding that my speakers will not arc since they will not permit the full capacity of the cell to discharge across the gap at one point. While it is true that my speakers do not arc when driven with music, they will arc with foreign objects in them or with excessive polarizing voltages. However, they do not burst into flames when this happens, as a low impedance diaphragm does. I believe this is because the high diaphragm impedance prevents the total capacity of the cell from discharging at that point. Since the electron flow is limited, the heat generated by the spark is limited, and the temperature stays low enough so the cell will not burst into flame, or even melt a hole in the diaphragm.

Furthermore, I have found exactly the same behavior even with the polarizing supply disconnected (after the diaphragm has been brought up to voltage). I think the high series resistance in the polarizing supply, most importantly, prevents rapid electron flow onto the diaphragm, and therefore prevents the quick destruction of the electrical contact between the diaphragm and the diaphragm contact point. Perhaps this was the cause of the contact failure in the Janzen ESLs to which he referred. Secondly, the series resistance further limits how much more current can be delivered to an arc, in addition to the basic diaphragm capacity.

Like Mr. Bradley's letter, I hope this reply raises even more questions and generates more study into the actual operation of ESLs. Interested readers are encouraged to share their findings. I may be reached by writing to R1, Box 125, Halfway, OR 97834, or by phone, (503) 742-2122.

Mr. Wagner replies:

Mr. Bradley asks about the mechanics of the dielectric breakdown, and if a high series resistor can prevent this condition from occurring.

Briefly the steps which lead to the breakdown condition are as follows. As the voltage between the diaphragm and the plate approaches the dielectric strength of air (approximately 75V/mil) there is a sharp increase in the leakage current between the plate and the diaphragm. As the number of charged carriers increases, so does their speed. During the transition state, these high velocity carriers can strike some of the atoms that make up the dielectric material.

When this happens, additional electrons are released, and this produces an increase in the current flow. At some point the voltage, between the plate and the diaphragm, collapses and there is a discharge of the electrostatic energy. Then, the breakdown enters its last stage, and is characterized by a low-voltage, high-current condition. If these last two conditions are prevented, the speaker will not experience any dielectric breakdown.

Mr. Bradley's second question is related to the series resistance that is located between the high voltage power supply and the diaphragm. Before explaining why this resistor has no effect on the dielectric breakdown, it should be understood that its purpose is to produce a constant charge on the diaphragm, and eliminate any distortion.

The following example will show why the resistor does not affect the current that flows during the breakdown period. Suppose that a 500pF capacitor is connected through a large series resistor to a voltage of 5kV. After some time has elapsed, the capacitor's accumulated charge will be:

$$Q = C \times V$$

where

Q = the charge on the capacitor in coulombs
C = the capacity in farads
V = the voltage across the capacitor

and this is equal to:

$$Q = 5.0 \times 10^{-10} \times 5000 = 2.5 \times 10^{-6} \text{ coulombs}$$

Now suppose both the 5kV source and the series resistor are removed. A 10Ω resistor is now connected across the capacitor. For a short time, the capacitor will establish an equivalent voltage across the resistor and a current will flow through it. As time passes, the charge on the capacitor will decrease, and when the discharge time is equal to five time constants, the voltage and the current will be zero. A time constant is defined as the product of the resistance, in ohms, and the capacity in farads. That is:

$$t = R \times C$$

where

t = the time in seconds
R = the resistance in ohms
C = the capacity in farads

For the above values, this is equal to:

$$t = 10 \times (5 \times 10^{-10}) = 5 \times 10^{-9} \text{ seconds}$$

During this short interval the voltage across the capacitor will decrease to 37% of its initial value, or 1,850V.

The current that flows through the resistor, in this time interval, can be calculated by the following equation:

$$i = [(v_o - v_1) \times C] / t$$

where

i = the current flowing through the resistor
v_o = the capacitors initial voltage
v_1 = the voltage on the capacitor after one time constant
t = the time constant in seconds

Using the indicated values, the current flow is equal to:

$$i = (5000 - 1850) \times (5 \times 10^{-10}) / (5 \times 10^{-9}) = 315\text{A}$$

As indicated, this current has nothing to do with the initial power supply or any resistor that might be connected to it. The calculated value is solely dependent on the capacitor charge and the resistance during the discharge time.

This same condition exists in an electrostatic speaker when a low resistance diaphragm comes in contact with one of the plates. In this case it is assumed that the resistance "R," during discharge time, is equal to the resistance of both the diaphragm and the plate. Although this current will flow for only a very short time, it is enough to damage the speaker.

When the diaphragm has a high resistance the conditions that will cause a breakdown are changed. The 500pF capacity is now made up of many smaller capacitors. A value of 10 will be chosen for this example. In addition to this, each capacitor will have an assumed series resistance of 10MΩ. After the capacitors are initially charged, the power supply is disconnected and the two connections representing the plates are again connected to a 10Ω resistor. This will place all of the series resistor/capacitor combinations in parallel with each other.

Because of the high resistance, the 10Ω resistor has little effect on the time constant. The value of t, for each of the capacitors, is now equal to:

$$t = 10^7 \times (50 \times 10^{-12}) = 500 \times 10^{-6}$$

The current flow from each capacitor is limited by its 10MΩ series resistance, and makes each of the branch currents equal to:

$$i = [(5000 - 1850) \times (50 \times 10^{-12})] / (5 \times 10^{-4})$$

$$= 0.000315A$$

The current through the 10Ω resistor is equal to the sum of the branch currents and this is 0.00315A.

These two examples illustrate the advantage of the high resistance coating, on a diaphragm, and show how it limits the short circuit current.

Although the above technique will usually solve the major problems associated with dielectric breakdown, the speaker can still produce some mini-arcs. This condition is created by sharp points or edges on the surface of the plates, and is usually the greatest around the holes. The sharp edge caused by the perforating process can produce an increase in the strength of the electric field. Because the plate resistance is low, this concentration can cause a series of very tiny arcs to occur between the sharp edge of the plate and the diaphragm. If the speaker is placed in a dark room, the location will be displayed by the bluish glow.

The application of an insulated coating to a speaker with a high impedance diaphragm is a precautionary step, but one that can prevent a lot of aggravation after the speaker has been assembled. In my book, I not only advocate this procedure, but also suggest that you conduct a breakdown test. This is relatively easy to do. Hold the two plates together, without a diaphragm, by a set of "C" clamps. Apply a voltage between them that is twice the normal DC value. In a dark room, any tendency to arc will be visible, and corrective steps can be taken.

I hope this explanation helps Mr. Bradley understand the mechanics of the dielectric breakdown, and the effect of the series resistor.

OUT-OF-PHASE RIBBONS. . .

I have no experience in ribbon design, but I have an idea why Ole Thofte's pulsating cylinder (*Mailbox, SB* 4/88, p. 57) sounds timid to him. If the out-of-phase sound from the *rear* of the ribbons on the *opposite* side of the cylinder travels between the ribbons, on the side facing the listener, it reaches the the listener's ears. This should not occur in a true pulsating cylinder.

At low frequencies you will have complete cancellation, while at higher frequencies you will have a comb filter effect, due to the time delay corresponding to the cylinder's diameter. Maybe a half-cylinder would eliminate this problem.

Also, since there is no baffle between ribbons, the rear radiation from the ribbons facing the listener also begins to cancel the front radiation, possible from 5kHz on down. This is why ribbons are usually placed in a 6-inch or wider baffle. One solution, as Mr. Thofte suggested, is to make a planar membrane (possibly by gluing ribbons to a plastic sheet) in a cylinder shape; or, insert narrow baffles

Continued on page 73

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between the ribbons or place the ribbons closer together.

If you make a planar membrane cylinder, perhaps you could seal the top and bottom. If not, you will obtain cancellation at low frequencies due to the out-of-phase radiation from the inside surface of the cylinder. Of course, all planar drivers suffer from this problem, and designers usually compensate by using deliberate high-Q low-frequency resonances. If you do seal the cylinder, then you will limit bass response due to the low compliance of the enclosed volume of air; that is, you will have an acoustic-suspension speaker! On the other hand, the suspension caused by the stretching membrane may be more significant.

A half-cylinder with baffling on the rear should help extend bass response by increasing the time delay of the rear radiation with respect to the listener.

Ralph Gonzalez
Philadelphia, PA 19143

. . .AND CYLINDRICAL FLAWS

Two reasons are immediately apparent why Ole Thofte's cylindrical ribbons (*Mailbox, SB* 4/88) don't work well. First, a theoretical flaw exists in the system that can be solved, and the second flaw is inherent in the design of a cylindrical radiator.

I assume the system is used full range, in which case I am amazed his system has any bass response at all. The reason is simple—nothing is *between* the strips of ribbon to prevent front-to-back cancellation. Perhaps Mr. Thofte thought the relatively large surface area would produce "rather powerful" speakers. If he refers to "La Folia" (*SB* 1/88), that design worked well because it used relatively large *continuous* membranes for the woofer, as opposed to thin strips of ribbon. For the same reason, a woofer must be mounted in a box to work—the air in front of the woofer must be physically isolated from the air in back. The pressurized air his ribbons are so mightily trying to push out, in the form of sound waves, are literally curling around behind the ribbons and equalizing the pressure, resulting in front-to-back cancellation. The solution is hinted at by Mr. Thofte; use a planar membrane instead of ribbons.

I can't think of any way to salvage the design as it stands. Theoretically, you could calculate the exact frequency at which the cancellation occurs, based on the length of the ribbon, and cross over to a subwoofer. However, many practical reasons that I won't elaborate on would render this approach useless. I believe, though, this front-to-back cancellation would occur diffusely across the audible range, resulting in Mr. Thofte's impression of inefficiency.

The second reason for inefficiency is unavoidable. The cylindrical design radiates the majority of its output *away* from the listener. From the photo, it appears only three or four of the 30 ribbons used would make any contribution to the direct sound heard by the listener.

I hope these points will help answer Mr. Thofte's inquiries. It takes courage to submit original designs to the scrutiny of *SB* readers, and I look forward to his future efforts in this magazine.

R.B. Boenig
Marietta, GA 30067

DRIVER DREAMS

I am a new subscriber and so far *SB* is just what I hoped it would be. However, could you publish some information on what type, size, and alignment of speakers are best for various types of music.

Carl Richard
Enfield, CT 06082

[The impossible dream!—Ed.]

INDUCTANCE VS. CAPACITANCE

This is a belated response to Mr. Coyle's article, "Custom Wound Inductors," (*SB* 3/82). I am also interested in first-order series networks, even though they are in bad odor for some. I.M. Fried has suggested series networks produce a difference in sound that can be heard. There should be more research on this.

Some authors, Baekgaard and Ashley, for example, do not agree on how to design a quasi second-order network. Baekgaard's equations[1] are:

$$C = a/R\omega_o \text{ and } L = R/a\omega_o$$

Baekgaard specifies: $a = \sqrt{2/3}$, for flat response; and a = 1 for constant input impedance, the normal design value. If $a = \sqrt{2/3}$, then C must be smaller in value and L must be larger in value, than in the case of a normal first-order network.

Ashley apparently had the reverse situation, a larger C and smaller L. A Speakerlab publication[2] mentions that Ashley repudiated his earlier recommendation of

1. Baekgaard, Erik, "A Novel Approach to Linear Phase Loudspeakers Using Passive Crossover Networks," *JAES*, May 1977.

2. *Loudspeaker Design Guide*, Speakerlab, 1983, p. 20.

the quasi second-order network at the 1976 International Loudspeaker Symposium. He noted then that the larger value for C and smaller L drops the input impedance of the network by a factor of 2, doubling the power drain. This was for:

$$C = 2C_N \text{ and } L = L_N/2$$

where C_N and L_N are the normal values. According to the Speakerlab manual, Ashley also said that he had decided systems using the normal values sounded better.

If C is doubled and L is halved, in the Ashley quasi second-order network, then it appears Ashley's equations (using Baekgaard's notation) would be:

$$C = 1/aR\omega_o \text{ and } L = aR/\omega_o$$

And Ashley's original recommendation was: a = 0.5.

Has anyone researched the effects of increasing capacitance and dropping inductance versus increasing inductance and dropping capacitance, in series crossover networks?

David Weems
Newtonia, MO 64853

Daniel Patrick Coyle replies:

The odor is sweet to me. I have preferred series to parallel passive crossovers for some time because of what I subjectively identify as the better transient clarity of the series configuration. I came to this impression listening to domes.

I don't have the Speakerlab publication you refer to, but there is no disagreement between what I have read in several of Ashley's papers and the Baekgaard paper. Perhaps Mr. Ashley will set us all straight, but here is my view.

In "Active and Passive Filters as Loudspeaker Crossover Networks," (JAES, June 1971), Robert Ashley and Allan Kaminsky explain that Kaminsky developed the quasi second-order filter by relaxing the constant input impedance criterion. In both papers, the transfer characteristic for the quasi second-order network is the same, although there is a notational difference. (Ashley and Kaminsky use zeta for Baekgaard's lower case "a"; and omega for the wave number, which Baekgaard normalizes to one (see equation 1).

The value of "a" (or zeta) may be thought of as the inverse to a damping factor, and Ashley and Baekgaard select different values. If the damping factor is changed, then the behavior near resonance is affected. You may choose a value to optimize input impedance or time delay or amplitude linearity. If zeta is fixed at 1.0, then the filter behaves as a first-order filter and the inductive reactance balances the capacitive reactance, producing a resistive load which is easy for the amp to drive; if set at 0.5, then the filter behaves as a quasi second-order filter, which means slightly steeper roll-off rates at crossover. Baekgaard chose the root of ⅔ to optimize for maximum flat response for each driver. As zeta changes, the values for L and C also change for a given crossover frequency.

Here is the effect: when L is increased and C is decreased, the value for "a" or zeta increases and dampens the circuit. Conversely, when L is decreased and C is increased, the circuit produces a slight increase in amplitude for each driver just above crossover, followed by a steeper dip at crossover.

Regardless of the value of zeta, in theory, the ideal amp will not have to deliver more power because the net impedance should remain constant, although its reactive component will be more capacitive or inductive as theta is varied. Real amps have decided "preferences" as to which domain they will best drive at a certain frequency. A reactive load may be thought to spit back electricity at the amp, making it harder for the amp to develop a given voltage across a speaker's terminals. For this reason, it may be better to not get too reactive with too low a zeta.

UPDATE: RCA BATHTUB HORNS

In response to the *Mailbox* section, *SB* 3/88, and Todd Wilson's request for information on the RCA Bathtub low-frequency horns: the cabinets he refers to were used quite extensively by A-1 Audio, a Los Angeles sound company, for stage sound for many years. I suggest Mr. Wilson try and contact them, at the following address:

A-1 Audio
6322 DeLongpre
Los Angeles, CA

They used these cabinets and made modifications. I have rented systems using these cabinets and was impressed with their work. At this time, I do not know whether they still use them, but I'm sure they they could supply some information on the cabinet design and modifications.

Kevin Ruud
Las Vegas, NV 89108

DRIVER SPECS WITH CROSSOVERS

I recently was able to replace the ADS speakers in my main stereo system and would like to build new crossovers to use them in in my family room audio-video system. These speakers have a 10-inch woofer, a 2-inch dome mid, and a 1-inch dome tweeter, crossing over at 750Hz and 3kHz, in a second-order crossover.

The speakers have a harshness in the upper midrange which I have long suspected is crossover related. After reading Bob Bullock's articles and Vance Dickason's (*Loudspeaker Design Cookbook*) crossover chapter, I think the problem may be that the tweeter is crossed over too low with its resonance in the pass band; interference patterns from closely-spaced crossover points; and impedance problems as shown by Bullock's article (*SB* 4/87).

I have decided to use, if possible, a third-order crossover at 750Hz and 6kHz, or 625Hz and 5kHz. I can make all the measurements Mr. Dickason describes, but I am uncertain about two things. For one, it would be a great help to know the sensitivity of each driver and the upper frequency response of the midrange. I have written twice to ADS, but have no response so far. How can I obtain the specifications I need, in a way which would be practical for a home hobbyist?

Edwin Gianelli
Ephrata, PA 17522

QUALITY IMPROVEMENTS

I want to thank you for putting out such a fine magazine. I have recently upgraded my home stereo system from what I thought was a great sounding Japanese system to a superb sounding American system, purchasing a new Hafler DH-100 preamp and DH-120 amp at a local sale and I'm very proud to own them.

Using designs from *SB* back issues and from my own experience, I am building a fine sound system. I am using the Morel MW160 6-inch woofers with the Audax DTW100SP25BACAUFF, crossed over at 3kHz (recommended by Madisound). I am using Octaline enclosures (*SB* 3/87) as my rear fill ambience speakers, using Infinity's A42, 4-inch full-range car speakers.

I added Monster Cable interlink to the CD and preamp, and also on the front speakers, for a big improvement. I can't say enough about what high quality cable does for a system; I recommend it to all my friends who own hi-end equipment.

In a future issue, I plan to describe my latest projects. I wish to add a subwoofer system and a Hafler XL280, and upgrade to a DH-110 preamp.

Frank DiCristina
St. Louis Park, MN 55416

EQUALIZER IMPROVEMENTS

Loudspeakers have been my hobby since the twenties. My first was a Tower horn, made in Boston; the second, a Brandes horn. My last commercial speakers were big Bozaks, which I gave away when my hobby became active again after retirement.

Since 1978, I have done a lot of experimentation, most of which was fairly futile until I got down to business and devised a measurements procedure to enable me to see on a sound level meter exactly what I was doing when I moved my equalizer levels. Because each band affects, not just one band, but a considerable part of the response curve, the actual response curve is often a far cry, in the most critical places, from what you expected. Automated equalizers assume wrong suppositions, and this shows in the results, and also that equalizers have not picked up the market share they should have.

I'm sure many experimenters have shared my exasperating experiences, when extremely careful adjustment of equalization levels produced unexpected results. Using careful measurements, I found a ⅓-octave equalizer can always produce an improvement. I wanted to find a low-cost way to equalize good quality, high production drivers that would give one point of origin for the sound they produced, and with dependable characteristics over their normal life. I was disgusted with equalizers until I learned to use them in a constructive and precise procedure.

Another problem was how to get realistic sound over the whole listening room. After much experimenting, I have arrived at a few oddball basics. Speakers should be located, not at the end of a room, but near the center (length), about one foot out from the side walls, with the speaker heads about four feet from the floor. My enclosure is designed for rigidity and lined with material that blocks low frequencies from energizing resonances in the wooden box, and from passing more low than high back-wave energy out to the listening room.

I think Bell Labs, many decades ago, established that galvanized iron was almost as good as lead, and much lighter and safer, to block lows. I use it, buying it in roofing rolls.

The boxes should be tipped back slightly toward the side walls, so their sides are not parallel to room boundaries. I use three Jensen triaxials in each enclosure, one each to serve the the two halves of the room, and one facing up. This approximates a point source, the intersection of the three driver axes, inside the enclosure, more closely than any other arrangement. The two horizontal speakers in each box are angled in, such that all points in the room get their most nearly on-axis radiation from the more distant speaker.

Results are very good, and I have made no changes for many months. I use a powered M&K subwoofer for the bottom octave, positioned near the center of the end wall. Not only is the sound realistic in the listening room, but the illusion of reality extends through the whole house.

Have you ever thought of publishing subscribers' names so we can share our mutual interest with others nearby?

Edgar Jones
Englewood, FL 34223

The Editor replies:

Readers interested in group activity are welcome to use the classified columns to contact like-minded area audiophiles.

Classified Advertising

TRADE

FALCON ACOUSTICS is one of the largest independent manufacturers of crossover networks and speaker system accessories in the UK, supplying manufacturers, retailers and export. We have outlets in both the USA and Canada. Your guarantee is the trust our manufacturing customers put in the quality and reliability of our networks. **Inductors:** Ferrite (up to 200W), air-cored and transformer; **Capacitors:** Reversible electrolytic 50V/100V, polyester, polycarbonate and polypropylene; **Networks:** choice of over 70 for different unit combinations; **Components & Accessories:** most except for the wood! If you have supply problems, please send for details, $2 air, $1 surface, to our mail order retail outlet. ***FALCON ELECTRONICS***, Tabor House, Norwich Rd., Mulbarton, Norfolk, England NR14 8JT. TTF

SPEAKER REPAIR. 4″ to 18″ speakers reconed, Grilles recovered, custom crossovers. ***ORBAN AUDIO***, 119-7th St. N.W., North Canton, OH 44720. (216) 497-9932 6:00-9 p.m. T5/88

5¼ INCH CAST FRAME, Neoflex cone, vented pole piece, 40mm Kapton former, aluminum, edgewound flat wire. Voice coil length, 18mm, 15Ω driver, Fs = 43; Qts = 0.274; Vas ″ 11.4 liter; 87dB/W/M. These special drivers are brand new and the original contractor could not take delivery. These are a special one-time offer at a cost far below market value. They are ideal for line array systems. For prices and quantities, call ***FOCAL AMERICA***, (818) 707-1629. T5/88

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ACOUSTA-STUF—New long fiber synthetic stuffing material, multi-directionally crimped for sonic purity comparable to wool. $7.00 per pound. ***MAHOGANY SOUND***, 2430 Schillingers Rd., #488, Mobile, AL 36695. T3/89

FOCAL AND ETON KEVLAR drivers with parameters optimized for transmission line use. Cast baskets, vented coils, high compliance, low resonance and Q. Call or write for specs and prices. Enclosure plans and consultation available with driver purchase. ***EPIK AUDIO***, 1720 Lilac Dr., Walnut Creek, CA 94595, (415) 930-9104. T5/88

LONG HAIR WOOL carded, cleaned for stuffing speakers. $13.50/lb. including shipping. ***J. EBBERT***, 431 Old Eagle School Rd., Strafford, PA 19087. (215) 687-3609. TTF

MAGNAVOX CD PLAYERS: CDB480, $145; CDB482, $169; CDB486, changer, $259; CDB473 remote/volume, $239; CDB582, $199; CDB586, changer, $279; CDB650, reconditioned, $259; other models reconditioned, call. Trades. Extended warranties. TVs, VCRs, Monster Cable, Esoteric Audio, Ceramic Borosilicate coating, MTX raw drivers. Free catalog. Call ***DIGITAL SOUND***, (312) 674-8024. T4/89

FOR SALE

Ten used Dayton-Wright electrostatic cells, $6 each; ten tested only, $10 each. 18¼" x 8" 2". David Landrie, 120 S.E. Everett Mall Way, #1113, Everett, WA 98208, (206) 353-4036.

Several DEC PdP11/03, 11/23 systems including lots of memory, floppies, removable hard disks, many peripherals, including high speed A/D and D/A, terminals, rack enclosures, operating systems, WP software, languages and much more. Call Dick Pierce, (508) 433-9283.

Heathkit capacitor testers, models IT-11 and IT-22. Contain "cyclops" tubes (green-eyed monster). Both need work. Sell as pair, $25. Solid wood cabinet for H.H. Scott receiver, mint, $15. Both items plus shipping. B. Hefner, Box 6601, Newport News, VA 23608.

Hafler DH101 preamp kit, factory sealed, $100; Sony SQD-1000 SQ decoder with manual, $20; AR XA turntable with DC and headshell, $35; Audio Control D-11 octave equalizer/analyzer, $125. Bob Kuhn, (404) 475-0694 before 9:00 p.m. EDT.

Dahlia-Debra speakers designed by Dick Olsher (*Stereophile* Vol. 10, No. 4), has Audax TX2025TSM with MB Electronics' MCD25M titanium dome, Built with updated angled front baffle, all SiderealKaps. Can send a review by J. Gordon Holt. Gary Kimes, (408) 394-8880.

Electro-Voice Intervoice-C, equalized, mint, $650; Mobile Fidelity Beattle Box, mint, $550; Thorens TD-H6 MK II turntable. Dan Dennison, RFD 10, Box 386A, Concord, NH 03301, (603) 798-5802.

Klipschorn, homebrewed from Klipsch plans. $100 plus shipping, $50 if you pick up; Lowboy infinite baffle, large and heavy, for 32 speaker array plus 40 unmounted 5" speakers, type used in AR1, $50 plus shipping, $25 if you pick up. Norman Stanley, PO Box 723, Rockland, ME 04841, (207) 594-7053.

Tektronix spectrum analyzer 5L4N in storage scope 5103 mainframe, $1850 includes manuals and external local oscillator to create TDS system. Lin or log sweep to 100kHz 80dB range. Marshall Buck, 3221 Provon Lane, Los Angeles, CA 90034, (213) 559-3947.

Dynaudio DS4AF, new, $140/pair; D28, $60/pair; Morel PP8 with grilles, 10 hours, $100/pair; MW220, $20; AR SRC remote system, $65; Numark DM-1550 mixer, $50. I will pay shipping. Dan Patten, 720 E. 3230 N. Provo, UT 84604, (801) 375-1086.

DeCoursey model 120-R, two-way electronic crossover, 100Hz, rack mount, 18dB/octave, 4 level controls, $225; Technics SH-8055 12-band graphic equalizer with real time spectrum analyzer and RP-3800E microphone, $200. Dave Schneider, (319) 373-2275.

Fried Studio IVs with new high power tweeter in oiled walnut, $800; Duntech PCL-3 wall speaker, great for surround sound or where space is at a premium, $400. Tony, (508) 478-0426.

Janis woofer, no cabinet, $100; SEAS CA25RE4X/DC pair, unused, $65; Sony 5" square flat mid-bass, unused, $35 pair; 6.5"square flat bass, unused, $65; AR receiver, $175; EPI LS841 auto system, $185. R. Omegna, (201) 543-2971.

Threshold 400A Class A amplifier, 100W/channel, mint, $550; SAE 2700B graphic equalizer, pro model, 20 bands per channel, $450. Sy Saks, (213) 939-3482.

Real time spectrum time delay, reverberation time, equalization and other test system with extensive microprocessor analysis and hard copy. Complete system design algorithms included. Manual available. Ivie Electronics IE-30A, IE-17A, IE-12P and IE-20B. Bruel & Kjaer microphone and SPL calibrator. Hewlett-Packard 7035 X/Y recorder. $4,350. Paul Allyn, (206) 454-9562 or 746-6087.

Pair Morel MDM 75, 3" dome, never used, $50; pair Peerless TO115F 4Ω, used, $16. David Long, 3248 Chatsworth Rd., Dalton, GA 30721.

Dynaco ST-70 with Van Alstine mods and new tubes, $125; Dynaco ST-35 "stock," $65; Threshold CAS-1 75W/channel dual mono, $375; Van Alstine MOSFET 120B 40W/channel, $200; NYAL Moscode Super-It, phono only preamp, $185; B&W DM 100 speakers with film caps and cabinet damping, $185; Stax SRX MK III electrostatic headphones with 16 ft OFC extension cable (needs Stax preamp or transformer), $150; RussSound speaker switcher, $30; Grundig two-way mini-speaker, $50; Old Colony Dynaco MK III power supply mod kit for two amps, $30; Old Colony moving-coil headamp kit, partly assembled, all parts and tubes, $60; Pioneer 9500 MK II tuner (analog), recently aligned, $175. C.J. Poulos, 40 Hodge Rd., Marlborough, CT 06447, (203) 295-8714 evenings.

Focal 5N401 5", $26; SEAS H398 1" aluminum, $14; 21FWBX-DD 8", $20; P14RCY 5" poly, $24; Audax MHD 17B37R, 6½" cast-frame, $29; HD20B25H2C12 8", $24; TX2025 RSN 8" TPX, $24; MHD 24B45 10" cast, $38; TW51A $6; DW6.11M, $7; Dynaudio D28AF, $25. Trades for Scanspeak 13M/8621 midranges. Steve, (503) 644-4965.

WANTED

B&O Beogram 3000 cartridge and or stylus, Beomaster 3000-2 for parts only. Would also like to trade my cabinet-building expertise for your crossover or related expertise. Carmel-Monterey-Salinas areas only. Gary Kimes, (408) 394-8880.

Quad 303 power amp. Can be in any condition. Jerry McNutt (205) 745-3131 days or 821-8426 nights.

Information on designing horn loaded drivers, formulas, etc. Also interested in articles on building and modifying ribbons and horn loaded Heils. Gary Clausen, B. Co. 11th AD SIG BN, APO NY 09801.

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Copy of shop manual, schematic and/or owner's manual for a C/M Labs 35D stereo power amplifier. Charles Ford, 1080 N. Carpenter Rd., Titusville, FL 32796. (407) 269-2508 after 4:00.

Dyna PAS 3X, GSI 4tp or similar PAS type tube preamp in good working condition. Steve Frankowiak, 323 N. Stewart, Lombard, IL 60148.

Need access/photocopies of plans for JBL Hartsfield speaker system, old JBL literature. D.R. Schaller, 6704 Schroeder Rd., Suite 6, Madison, WI 53711.

Crossover schematic and parts list for ADS L810 or L710 loudspeaker. Peter Coutts, 2928 Windsor St., Halifax, NS, Canada B3K 5E8; FAX 902-420-0366.

VSP 150 or 200 amp and preamp. Joe Kalnas, 204 Wood Ridge Dr., Mars, PA 16046.

To correspond with owners of Tangent speakers, specifically models RS6, RS8, PS6 and PS8. Carl Van Camp, 4235 W. St. Joseph St., Lansing, MI 48917.

CLUBS

Space in this section is available to audio clubs and societies everywhere free of charge to aid the work of the organization. Copy must be provided by a designated officer of the club or society who will be responsible for keeping it current. Send notices to Audio Clubs in care of the magazine.

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.

AUDIOPHILES IN CENTRAL PENNSYLVANIA (also eastern Pennsylvania and Delaware): Interested in forming a serious audio organization? Contact Steve Gray, 625F Willow St., Highspire, PA 17034 or phone (717) 939-4815.

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, recording studio visits, annual picnic and audio fun. The club journal is *LC, The SMWTMS Network*. Corresponding member's subscription available. Call (313) 477-6502 (days) or write David Carlstrom, SMWTMS, PO Box 721464, Berkley, MI 48072-0464.

PACIFIC NORTHWEST AUDIO SOCIETY (PAS) consists of 50 audio enthusiasts meeting monthly, second Wednesdays, 7:30 to 9:30 p.m. at 4545 Island Crest Way, Mercer Island, Washington. Be our guest, write Box 435, Mercer Island, WA 98040 or call Bob McDonald, (206) 232-8130.

SIXTIES MUSIC FANS. The first club for fans of Sixties Era music wants to meet you. Strong audio and quality recording emphasis. How-to information shared. Worldwide membership. Informative, entertaining, and provocative newsletter published bimonthly. Help the world remember when rock had real artistic merit. Free brochure: send SASE or IRC to Classic Rockers Music Club, PO Box 1043C, Stevens Point, WI 54481.

AUDIOPHILES AND EQUIPMENT BUILDERS in the Long Island area—well established club looking to expand and share out experience. Monthly meetings, record critiques, technical help. Island Audio Club, 589-4260 (Suffolk), 271-4408 (Suffolk and Nassau), 825-2102 (Nassau).

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

THE AUDIO SOCIETY OF HONOLULU cordially invites you to attend one of our monthly meetings and meet others like yourself who are interested in the hows and whys of audio. Each meeting consists of a lively discussion topic and equipment demonstrations. For information on meeting dates and location, contact Craig Tyau, 2293A Liliha St., Honolulu, HI 96817.

HI-FI CLUB OF CAPE TOWN, South Africa issues monthly newsletter for members and subscribers. Get a different approach to understanding audio, send two IRCs for next newsletter to PO Box 18262, Wynberg 7824 South Africa.

WASHINGTON AREA AUDIO SOCIETY (N. VA, MD and DC) is looking for sincere audiophiles who are eager to devote their time and get involved with the direction of the society and the publication of a monthly newsletter. Please contact: Horace J. Vignale, 13514 Bentley Circle, Lake Ridge, VA 22192-4316.

SAN DIEGO AUDIO SOCIETY forming for hi-fi tinkerers and do-it-yourselfers. If you enjoy collecting, building, rebuilding and repairing classic audio equipment, especially tube-type, call Mike Zuccaro (619) 271-8294 (evenings & weekends). Old timers and engineers welcome.

About Your Personal Classifieds . . .

An increasing number of readers are telephoning for information about their personal classified ads. Ads are filed as received, set in type late in the magazine production cycle, and put on page approximately one month before the magazine is mailed. Delivery of the mailed magazine can take from four days to three weeks.

Personal classified ad copy that is included in one issue is discarded when it goes on the page, and a new file for the next issue is begun. Ads arriving after the issue closes will be run in the next issue.

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1. All "For Sale" ads must be personal sales and not for profit. Ads for resale at a profit must be submitted as "Trade" ads at 55¢ per word, prepaid.
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MINNESOTA AUDIO SOCIETY. Monthly programs, newsletter, special events include tours and annual equipment sales. Write Audio Society of Minnesota, PO Box 32293, Fridley, MN 55432.

NEW JERSEY AUDIO SOCIETY meets monthly. Emphasis is on construction and modification of electronics and speakers. Dues includes monthly newsletter with high-end news, construction articles, analysis of commercial circuits, etc. Meetings are devoted to listening to records and CDs, comparing and A-Bing equipment. New members welcome. Contact Bill Donnally, (201) 334-9412 or Bob Young, 116 Cleveland Ave., Colonia, NJ 07067, (201) 381-6269.

THE ATLANTA AUDIO SOCIETY started in October 1983 and has regular meetings on the third Sunday of each month as well as special programs with leaders in the industry, such as Mr. William Conrad of Conrad-Johnson and Mr. William Johnson of Audio Research. We are currently looking for additional members in the Southeast. all members receive the minutes of each meeting and program, as well as other relevant announcements and correspondence. For full information and membership packet, write Atlanta Audio Society, PO Box 92130, Atlanta, GA 30314, or call Howard Royal in Newnan, GA, (404) 253-6419.

THE BOSTON AUDIO SOCIETY INVITES YOU to join and receive the bimonthly *B.A.S. SPEAKER* with reviews, debates, scientific analyses. and summaries of lectures by major engineers. Read about Apogee, Nytal. Conrad-Johnson, dbx digital, Snell, music criticism and other topics. Rates on request. PO Box 211, Boston, MA 02126.

SAN FRANCISCO BAY AREA AUDIOPHILES. Audio constructors society for the active, serious music lover. We are dedicated, inventive and competent. Join us in sharing energy, interest, expertise and resources. Send self-addressed, stamped envelope to S. Marovich, 300 E. O'Keefe St., East Palo Alto, CA 94303 for newsletter.

THE INLAND AUDIO SOCIETY IN THE SAN BERNADINO-RIVERSIDE AREAS, now in its third year of existance, is inviting audiophiles and music lovers in the San Bernardino, Riverside. Orange and Los Angeles counties to join us at our bi-monthly meetings and through our quarterly publication, in the pursuit for that elusive sonic truth. We provide a forum for auditioning equipment, sampling live music for educational purposes, guest presentations, discussing recordings, and the sharing of ideas, tips, theories, opinions, experience, and new product news relating to audio systems. Additionally we cater to the hobbyist who designs. builds and/or modifies electronic components and tranducing gear. Write for information concerning membership, dues and subscription. IEAS, PO Box 77, Bryn Mawr, CA 92318, (714) 793-9209.

ORGAN MUSIC ENTHUSIASTS: If live recordings of fine theatre pipe organs or electronic organs are your thing, I have over two thousand of them. I lend you the music on reels or cassettes. All operation is via the mail. A refundable dollar will get you more information. E.A. Rawlings, 5411 Bocage St., Montreal, Canada H4J 1A2.

THE COLORADO AUDIO SOCIETY is a group of audio enthusiasts dedicated to the pursuit of music and audiophile arts in the Rocky Mountain region. We offer a comprehensive annual journal, five bimonthly newsletters, plus participation in meetings and lectures. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212, or call Art Tedeschi, (303) 477-5223.

TUBE AUDIO ENTHUSIASTS. Northern California club meets every other month. For next meeting announcement send a self-addressed, stamped no. 10 envelope to Tim Eding, 2113 Charger Dr., San Jose, CA 95131.

THE VANCOUVER AUDIO SOCIETY publishes a bimonthly newsletter with technical information, humor and items of interest to those who share our disease. We have 40 members and meet monthly. Six newsletters per year. Call (604) 251-7044 or write Dan Fraser, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be on your mailing list.

MEMPHIS AREA AUDIO SOCIETY being formed. Serious audiophiles contact J.J. McBride, 8182 Wind Valley Cove, Memphis, TN 38115. (901) 756-6831.

ESL DIY'ERS: A new electrostatic loudspeaker do-it-yourselfers group is now forming. Our purpose is to share valuable theory, how-to, and parts source information for building our own state-of-the-art electrostatic loudspeakers. For further information, please write (SASE please) to: Neil Shattles, 829 Glasgow Dr., Lilburn, GA 30247.

THE WESTERN NEW YORK Audio Society (WNY Audio Society) is an active and growing audio club located in the Buffalo area. We issue a quarterly newsletter and hold meetings the first Tuesday of every month. Our meetings have attracted many local and distant manufacturers of audio related equipment. We are involved ein all facets of audio—from building to purchasing at discount prices. For a copy of our current newsletter and information regarding our society, please write to M.A. Monaco, WNY Audio Society, PO Box 312, N. Tonawanda, NY 14120.

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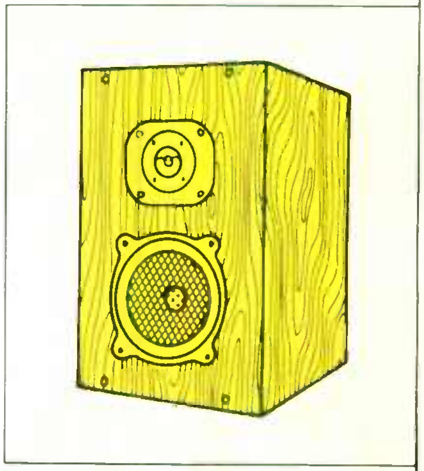

Illustration depicts FOCAL 280DB Kit. T-120 Tweeter and 7N402DB Dual Coil Driver.

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Fast Reply #JC61

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Fast Reply #JC29

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high fidelity range

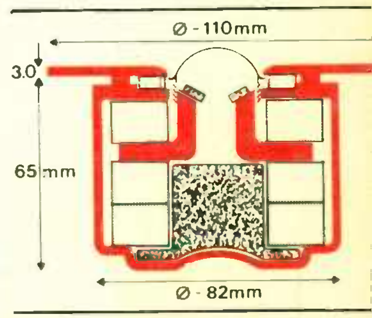
The MDT 33 is an extremely fast Tweeter, using a 28mm (1⅛") diameter voice coil and a chemically treated soft dome, and is ideally suited for two way systems with the possibility of a lower than normal crossover frequency, as well as for three and multiple way systems.

Incorporating the Morel Hexatech voice coil technique, aluminium wire wound on an aluminium former and using flexible wire termination ensures excellent high frequency performance with exceedingly high power availability. The power handling is further enhanced by using Ferrofluid in the magnetic circuit.

The magnetic system itself is an ingenious Morel double magnet design and is completely enclosed. By venting into the enlarged area of the double magnet system, a low resonant frequency of 500Hz is obtained with a remarkably smooth roll off from 1000Hz through this damped resonance area. The subsequent wide range response of 1400-20000 ± 0.6dB is obtained with a harmonic distortion of below 0.8% over the entire range. The distortion figures quoted are with an input power giving an output level of 96dB at 1 metre. The MDT 33 sensitivity is 92.5dB for 1 watt 1 metre, and a power handling capability of from 100 to 500 watt subject to crossover frequency.

With such a dome tweeter design, the acoustic qualities at lower than normal crossover frequencies are excellent with an absence of honking, and even at the more normal crossover frequencies this excellent acoustical behaviour is evident to the ear. With the lower crossover frequency available and high capability, it is ideal for consideration in two way systems using a 10" or 12" woofer.

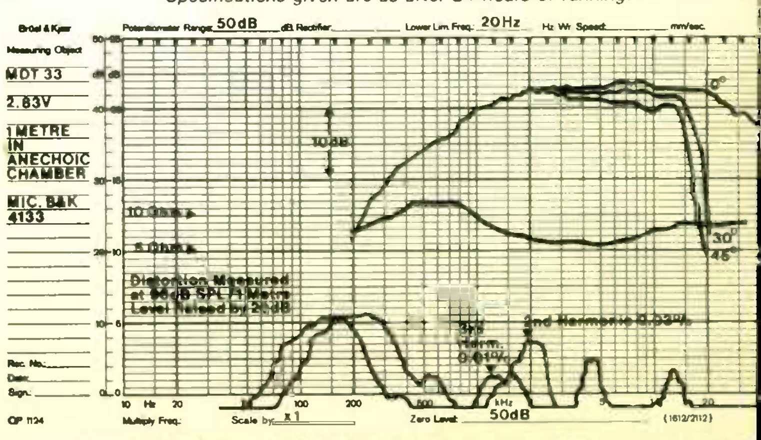
To utilise the dome at the lower than normal crossover frequency available makes it necessary to have a sharp roll off below 1400Hz of minimum 12dB per octave to protect the tweeter from mechanical damage. This makes it ideal for use with active systems.



Specification

Overall Dimensions	Ø - 110mm x 68mm	Vas	0.016
Face Plate Thickness	3mm	Moving Mass including Air Load	0.44 gr
Voice Coil Diameter	28mm (1⅛")	Effective Dome Area	8.5
	Hexatech Aluminium	Dome Material	Treated Fa
Voice Coil Former	Aluminium	Frequency Response	1400-20000 ± 0.
Number of Layers	2		(1000-40000 – 5
DC Resistance	5.2 ohms	Resonant Frequency	500
Nominal Impedance	8 ohms	Power Handling Din:	
Voice Coil Inductance @ 1 Khz	0.09mh	X-Over 1400 Hz	10
Air Gap Width	0.75mm	X-Over 5000 Hz	50
Air Gap Height	2.5mm	Transient Power 10ms	150
Voice Coil Height	2.7mm	Sensitivity	92.5dB (1W/
Flux Density	1.95T	Rise Time	1
Force Factor (BXL)	4.76 WB/M	Intermodulation Distortion	
Rmec	2.09ns/m	for 96dB SPL	<0
Qms	0.66	Harmonic Distortion	
Qes	0.38	for 96dB SPL	<0
Q/T	0.24	Nett Weight	1.

Specifications given are as after 24 hours of running.



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

morel acoustics USA
414 Harvard Street
Brookline MA 02146
Tel. (617) 277-6663
Telex 6502499475

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

Fast Reply #J